

Richard H. Dorf is President of The Schober Organ Corporation, which makes electronic organs in kit form and sells them throughout the world. He was a cofounder of the company in 1954 and has been its principal designer. Prior to the company's inception, he had gained a reputation as the leading independent authority on electronic musical instruments by acting as a consultant and through his many technical magazine articles, of which he has had several hundred published, as well as three other books, on various subjects.

A love of music stemming from his first pianistic efforts at the age of five combined with a strong interest in electronics to put him in the vanguard of high-fidelity enthusiasts long before the public at large ever heard of fidelity, high or low. Much of his writing and consulting were in this field during hi-fi's emergent era. His study of electronic organs began in early 1948 and his first series of magazine articles shortly thereafter was the most comprehensive information ever published on the subject.

Mr. Dorf continues to wear two hats—one as head of Schober, and the other as the independent writer on organ subjects who continues to hold the confidence of organ manufacturers who freely supplied him information for this book. A writer of some wit and polish, Mr. Dorf presents a text which is anything but dry, freely larded with light touches, and replete with examples of his unique talent for making complicated explanations simple, a quality for which his now infrequent magazine articles are still prized by editors.

ECTRONIC MUSICAL INSTRUMENTS

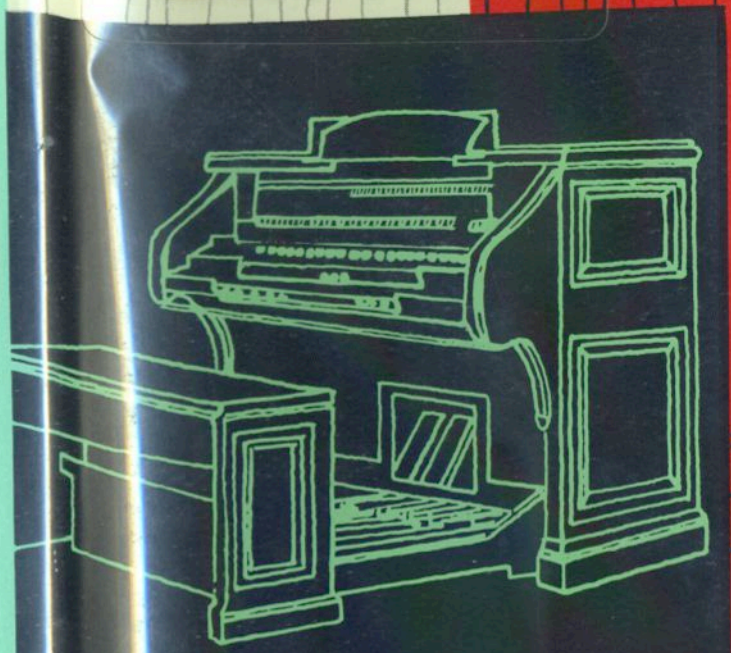
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Richard H. Dorf

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This latest edition of what has been since 1954 the definitive technical book on electronic organs brings the literature up to date on the most recent thinking and technology in this important field of electronics and music. It is a new "edition" in name only. Actually, it is a completely new book, nothing having been retained from older editions.

Electronic organs today occupy a place of honor in hundreds of thousands of American homes. Their manufacture and sale is Big Business. Yet the technical literature appearing in periodical and book form is too sparse and incomplete to be useful. Technicians, engineers, and hobbyists, as a result, have not had the opportunity to learn the fascinating technical background of musical electronics, and the number of those who know the subject is still so small that most servicing, for instance, is *still* done by local organ retailers.

This book will give the reader who has a basic understanding of electron-

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Electronic Musical Instruments

Third Edition

Richard H. Dorf

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Preface

Since the first (1955) and second (1958) editions of this book, electronic musical instruments—meaning almost entirely electronic organs—have become an important and, I am certain, a permanent and highly desirable fixture in American life. In the home, where most electronic organs are to be found, the instrument very often replaces the piano; and in terms of sales dollars the organ is the principal keyboard instrument of the day.

There are very good and easily demonstrated reasons for this. First, when you hold down an organ key, sound continues to pour out until you decide to let the key go, rather than beginning at once to fade away as on the piano. This means that the less adept player (who is in the enormous majority) can hold a chord and a melody note, complete with vibrato and at least two different tone colors, as long as necessary until he decides what comes next. Second, organs of today are capable of a great many tone colors and special effects, all of them available at the touch of a control and usually without any added skill. Much less technique is necessary to produce acceptable results on the organ than on the piano.

As a result, a relatively small investment in learning time pays tremendous dividends in musical satisfaction. Of course, the music played by most amateur organists, at least at the beginning, is simple and superficial, artistically speaking, but any organ owner will tell you that playing a simple tune yourself is ten times the fun of listening to a record of the best professional musician. As a player of a great many years standing, I am in total agreement.

We thus have the wholly new phenomenon of adults learning for the first time to play a musical instrument—and amazingly well. People in all walks of life are enriching their lives by making music. Any instrument that can accomplish this desirable end is worth attention, whether it is powered by electronics or ground-up unicorn horns.

The purpose of this new Third Edition is the same as that of the first two—to show you from the electronic technical viewpoint what organs are made of and how they work. Your purpose may be to design organs or service them—or just to satisfy your intellectual curiousity about a fascinating modern device. Your background may include an engineering degree or the practical electronic knowledge gained by experimenting in the basement. But if you have not dabbled previously in electronic organs, you will find a great deal that is new and interesting. I am proud to be able to say that the organ world contains more than one engineer and technician who entered the field because of an interest awakened in organs by this book.

Electronically speaking, organs have become much more sophisticated since the last edition. Tone generation, keying, and tone coloring are being done by more efficient and resultful means. A large number of musical "effects" is com-

mon now—percussions, sustain, rhythms, glides, and so on. And of course solid-state devices have almost entirely replaced tubes in recent organ models and are influencing organ design.

So I have written a completely new book from cover to cover, rather than simply amending and adding a bit as is customary with "new editions."

Two cautions. First, you will not get a great deal from this book unless you have a reasonable working knowledge of electronics. You don't need an engineering degree; if you read and understand most of such magazines as *Radio-Electronics* and *Electronics World* you are qualified.

Second, this book does not constitute a service manual for any organ. While the organs described as examples are described in considerable detail, there are so many models and designs change so fast that we would need thousands of pages and a new edition every four months to give you adequate service data.

Our purpose is very simple—to give you the knowledge and background to understand whatever organs you may run across, to recognize what the organ does, and, with the help of that organ's own service manual, to understand how it is done. If your object is organ design you must know what has already been done in the field. You will find that here.

In other words, this is a teaching text—but one which I believe you will find practical and interesting, lacking in didacticism, and containing no mathematical obstructions. If you find it in some places a bit "flippant" (as did one reviewer of the original edition) I will be delighted. I would like you to read the whole book and you are not likely to do that if you find it dull.

My publisher has kindly allowed me two spheres of personal lattitude which are unusual in a book about a serious technical subject. If you are accustomed to reading the weightier tomes in the engineering realm you may be shocked to encounter (a) frequent use of the personal pronoun "I" in place of the circumlocutory "the author;" and (b) some expressions of personal opinion and value judgements.

There are too many books of highly impersonal theory which succeed only in reciting that which is well known and may be found in any other book on the same subject. An important object of this one, on the contrary, is to give you what benefit may lie in my own personal, practical experience in both music and organ design. Since I am incontrovertibly being personal, little is gained by the awkward third-person style—except awkwardness.

I have tried to be fair minded but I do not believe that fairness requires the blind and bland implication that all things are equally good—certainly not fairness to the reader. I warn you, therefore, that when you read that something is good or bad or clever or stupid, remember it is only my opinion, with which after due experience, you may totally disagree.

One last very important and serious point. Despite all the electronics involved in organ design, as well as the many mechanical ingenuities, I hope you will keep firmly in mind that a good electronic organ is a musical instrument and not a design engineer's playtoy. The temptation toward musically useless gadgetry in organs is limitless because design engineers are always gadget happy, as I know by

looking in the mirror. If you are an engineer I beg you to keep in mind that an organ ought to be a medium of expression for perhaps the greatest and most universal art known to man, and not a technical curiousity.

Before ending these personal remarks let me indulge in a small tirade. You will find in this book no mention of Hz, kHz, mHz, or indeed any terminological reminders of the respected Heinrich. Despite the *diktat* of the National Bureau of Standards and the (in some cases expressedly reluctant) bending of the knee to a new Hertzian era on the part of practically all modern writers and publications, I am sufficiently antediluvian to believe that *cycles, cycles per second,* and *cps* have the inestimable value of intrinsic meaning (to say nothing of familiarity and lack of intricate capitalization). This meaning is clear and unequivocal to all technical English speakers, and any whining that "cycles" is inexact because it really means "cycles per second" is pure cavil, especially since Hertz means nothing (except perhaps to Heinrich's mother). To say that we should use the term to effect uniformity is nonsense, since its traditional users are not English speakers. Perhaps all technical writing should be in Esperanto; then we could really be uniform!

I am, I hope, forward looking in technical matters, but not everything new is better. As a confessed crank on the subject of English, I dislike split infinitives and anything that tastes good like a cigarette should. I prefer and carry a real fountain pen because I am silly enough to think it writes more smoothly and beautifully than a ball point. But I stick to the good old cycle on grounds of pure pragmatism. It means something and it is written and spoken without awkwardness.

Two important credits (to change the subject). First, my thanks to all the organ manufacturers whose products are described in the pages ahead. Every one furnished information promptly and cheerfully, and some went much farther with help of different kinds. Second, if your eagle eye does not detect more than the minimum number of errors, inconsistencies, and blurry explanations in the book, your thanks should go to Robert C. Avedon, who scanned the manuscript micron by micron with *his* eagle eye, and fought me to the death on every discernable deviation from the straight and narrow.

Richard H. Dorf
July, 1967

Chapter 1

Some Elements Of Music

Electronic musical instruments are at least as much electronic as musical from the designer's standpoint. As far as the owner and player is concerned, however, they are wholly musical, and whether there are transistors and capacitors inside the box or specially treated breakfast food he couldn't care less. The purpose of the instrument is to make music. Therefore, no matter how much the designer (and service technician and anyone else whose function it is to look under the hood) may know about electrons, he cannot work successfully with organs unless he understands their musical functions.

In this chapter, therefore, we shall take a look at the elements of music. Our examination will be a rather unromantic and unglamorous one, since we are less concerned with uplifting the soul than with reducing major musical elements to the physical characteristics necessary for an instrument which is to produce music. Music is a vast subject, of course, and we can only hit the high spots. If you are a musician, then, be prepared to forgive omissions, generalizations, and simplifications. If you are not a musician, however, read diligently, for you will be in no condition to understand electronic organs if you do not understand their end use—which is wholly and solely to produce music.

What Is Music ?

Music is one of the "fine arts," which are essentially highly refined works which express the highest and most intense emotions of man. Because man's emotions are so complex, it is just about impossible to define the products of an art form. Concerning painting, for example, one definition might be "depictions of scenes, people, and objects," but this definition would be incomplete. Many paintings are not depictions or reproductions of anything which exists in real life.

About all one can say to define music is that it is a succession of sounds designed to express the emotions of a composer and (hopefully) to affect those of listeners. One cannot say that the sounds must be of particular (or even any definite) pitches, for modern music includes sounds that under other conditions would be classified as noise, as, among many examples, the *musique concrète* of Edgar Varese. There is even, believe it or not, "music" which involves no sound at all—a "composition" by the extremist John Cage which requires a

pianist to sit at the keyboard for four minutes and some seconds and do absolutely nothing. And there is "music" (I wish I could emphasize the quotation marks) which does not even represent the expression of a composer–produced by programming a machine to serve up tones and other sounds with as complete randomness as possible.

Fortunately, conventional musical instruments such as the organs with which we deal in this book do not have to present the means for such a wide and chaotic expression of music as all this indicates. The music played on organs is, as a rule, "normal" in that its pitches and timbres are the same as those needed for the music with which most of us are familiar. This being so, we can make a much more usable definition of music for our purposes.

A glib definition, even of conventional music, is not easy and almost every musician will have his own version. Nonetheless, it seems evident that most music consists of a succession of tones of varying pitches, sounded sometimes singly and sometimes simultaneously, which pleases or otherwise affects the listener by appealing to his emotions. While the pitches cover a wide range, only certain specific ones are acceptable most of the time; and when played simultaneously, certain specific combinations are much more common than others. A great many different timbres or tone colors are commonly used, but these are also usually rather definite, corresponding to the characteristic sounds of what we call musical instruments.

Rhythm

Perhaps the most elemental attribute of music is rhythm, and this is not surprising because all the basic aspects of life and nature are imbued with rhythm. Rhythm may be called a pattern of regularity relating events to time. The four seasons are an example of rhythm. They always (in a specific part of the world) occur in the same order and at standard intervals of time. We normally walk in rhythm, placing one foot before the other with remarkable regularity. Babies howl in rhythm; dogs bark in rhythm; plants and crops grow and are harvested in rhythm. We chew, snore, and laugh in rhythm. In fact, when rhythm is missing and randomness takes over, as in stutterers and cerebral palsy sufferers, for example, disaster ensues.

Musical rhythm is obviously, therefore, instinctive, and even the most primitive civilizations practice it. The drum beats which aboriginal tribes use to accompany (and regulate) their dancing are a form of music, and in fact these drums show the first evolution of rhythm from its basic character of simple regularity in timing to something more sophisticated, as might be expected from the mind of man. In most aboriginal ceremonies the successive drum beats are not only regularly spaced in time; they also contain regular emphases. They may, for example, be divided into groups of four, with the first of every group emphasized by extra loudness: *um*-pah-pah-pah, *um*-pah-pah-pah. (The comma does not indicate any extra time spacing between the beats before and after it, but only the mental picture of separation of four-groups due to the emphases.) By varying the number of beats in a group, the number of chunks of time between certain pairs

of beats and other pairs, and the patterns of emphases, an enormous number of completely distinctive musical "compositions" can be played simply on a drum–and indeed civilizations with little or no melodic development do just that.

Rhythm in exactly the same sense pervades and dominates all the music we commonly hear and play today, though it is usually only one of several elements which make up the music. It serves two functions: (a) to satisfy our natural craving for regularity, and (b) as the framework which helps to *organize* music and contain it in a discipline, which is as essential in an art as in a science. The simplest minds with the least training and sophistication join all the rest of us in responding most quickly to the simplest rhythms–those with complete timing regularity, like *boom*-boom-boom-boom, *boom*-boom-boom-boom. You can confirm this by listening to any of the modern music adored by the teen-agers. More complex and interesting rhythms like *boom*-did-ee-um-pa-*boom*-dee-*boom*-dee require more experience for appreciation and appeal to more highly refined emotions and intellects. And some rhythms become so complex that the pattern is not even apparent to many people on first hearing.

Since rhythm is so important to music, all normal music must have it and all normal musical instruments or ensembles must have some way of producing it. In most ensembles like orchestras, bands, and the like, the principal rhythmic job is allocated to specific rhythm instruments such as drums and cymbals. In groups which do not contain rhythm instruments, like string quartets, and with orchestral instruments played unaccompanied, the rhythm must be implied in the timing of the melody notes and in appropriate loudness variations. However, with all instruments which are meant to be played principally by themselves, like pianos, harpsichords, organs, classical guitars, and so on, there is a specific means and technique for emphasizing rhythm, usually by combining it with the means of providing harmony and foundation bass.

A simple example is the waltz: *um*-pah-pah, *um*-pah-pah. A pianist often produces this beat by striking a single bass note for the *um*, and a higher chord for each of the *pahs*, with his left hand. This takes care of harmony and bass as well as rhythm, while his right hand plays melody.

The organist combines the functions in much the same way. On the *um* he plays a pedal, giving both the bass and the first-beat rhythmic emphasis. For each of the *pahs* he plays a chord on the keyboard used for accompaniment. And the melody is played with his right hand on the keyboard devoted to melody. All these are, of course, the simplest possible examples, and a very great deal of elaboration is possible and usual.

Melody

The invention of melody certainly came later in the history of human development than rhythm. Today, however, if we ignore certain very modern compositions and the things that go on in kindergartens (which sometimes have a remarkable similarity) melody is an indispensable part of music.

A melody is simply a tune–a series of consecutive tones of varying pitches. Melodies are by no means random; they are highly organized and intimately tied

in with rhythm. Melodies can be very simple or highly complex, and despite the fact that in conventional music the number of discrete pitches which may be used is severely limited, it is probable that new melodies can and will be invented until the end of time—or until some new system of making music takes over entirely. The scientific reason for this extreme proliferation of possibilities is the simple mathematical range of combinations and permutations. Even confining the available pitches arbitrarily to three octaves—36 or 37 different pitches—and the number of notes in a complete melody to perhaps 64, a pretty small minimum, you can imagine (or perhaps you *cannot* imagine) how many different combinations and sequences are possible.

Aside from mere sequences—and remember that notes can be consecutively repeated, too—rhythmic variations can add large numbers or orders of magnitude to the result. "America," for example, is pretty simple rhythmically, but notice that if you sing it so that all notes last the same time and are equally separated in time, it isn't the same at all. Next you might try rhythmic variations, like a long "My" followed by a fast "country 'tis of thee," then a prolonged "Sweet," etc., to show how an infinity of timing changes could alter the whole melody.

It is very nearly true that a random sequence of notes can be made to constitute a melody that makes musical "sense," as has been illustrated a great many times by musicians who specialize in improvisation and making melodies out of letters that spell someone's name, or a telephone number. The randomness is not really quite true, however. Such a musician organizes the notes into a recognizable rhythmic pattern, makes repeats of sections, begins and ends on appropriate notes, and does other things beyond the scope of what we are talking about.

To produce melodies, an instrument or ensemble of instruments needs two things. First, it must have on hand a source of tones of the correct musical pitches—the "accepted" ones—which it can select freely on demand. Second, if the instrument or ensemble can also furnish harmony, it must have a means by which the melody can be made to stand out and be heard clearly. In ensembles, melody is brought out by having it played on instruments which are audible above the others by reason of their loudness or of the "cutting" nature of the sound. In an organ, melody is brought out by this same means, because the organ has a multiplicity of available voices.

Harmony

There is nothing musically basic about harmony. Its development came rather late in human history, not more than a few hundred years ago in modern form; and in some cultures it is still used little or not at all.

A harmony is a group of pitches sounded simultaneously; this is as much definition as really ought to be given. We most often think of this group of simultaneous notes as one which is pleasant—*harmonious*—in and of itself, and which is also pleasing in combination with the melody note or notes being played at the same time. However, deliberate discords—harmonies with pitches that seem to clash and are, objectively speaking, less than pleasant—are far from uncommon in music and were, in fact, used with telling effect in many places by Bach himself. Much modern music, of course, uses practically nothing but discords, but all that is and will probably remain for some years outside the scope of the "accepted" musical conventions and thus does not influence organ design.

While the question is wholly subjective of whether a harmony is appropriate or not in the context of the composition in which it is used, there is a perfectly good physical basis for the pleasantness of the basic harmonies. The concept of harmony, therefore, was not simply dreamed up by some innovator.

The most basic "harmony" is the octave, two singers, for example, singing two notes an octave apart. As we shall see later in this chapter, any two notes separated by exactly an octave have a 2:1 frequency relationship. Middle A is 440 cps. The A an octave higher is 880 cps. The ear and brain sense an extraordinary "identity" between two frequencies having a small-number ratio—the smaller the better—and the combination sounds perfectly fine. The next closest feeling of identity is obtained from a 3:1 relationship, which amounts to notes an octave and a half apart, like C and the second higher G. Almost this same feeling can be had if the G is lowered an octave to give, say, the combination of middle C and the G just above it. (On a keyboard instrument, the tuning is such that the relationship is just a trifle less than 3:1 and accordingly the results are not perfectly pleasing.

The most basic chord of all is what our musical friends call the major triad, or simply the major chord. It is the one with which all conventional musical compositions begin and end. This most basic chord is simply made up of the notes having the lowest possible integral frequency ratios. Suppose we want a major chord in the key of C. The first note is therefore C itself. Two times the frequency of the C is another C an octave higher, which we can also play; but since we already have a C we don't actually need it. Three times the C frequency is, as we said in the last paragraph, a G; so that forms the next note of the chord. Four times the C is still another C. And five times the C frequency is an E (two octaves higher), so E is the last of the three notes of the chord. We can thus see that the simplest and pleasantest chord uses the three notes with the simplest frequency ratios—and selection of this chord by all conventional musicians is no accident or artistic license.

We can go on in this way analyzing other common harmonies. If we do, we will notice that the more complex the frequency ratios, the less obviously "pleasant" the chord is when we play it—though of course the chords are all useful since only an idiot would want everything he hears to be soothing.

Any musical group or instrument, then, to be capable of playing a great deal of music, must be able to sound several notes at the same time to form these harmonies or chords. In a group, each instrument (or each group of several instruments) sounds one note of the harmony, so that all the notes are heard when all the necessary instruments are playing. In a keyboard instrument, each key controls, in effect, a separate instrument tuned to only one pitch. We call upon several pitches at once by pressing several keys simultaneously.

Bass

To speak of bass as an "element" of music is a bit odd. For most music, however, it is just as essential an element as melody or harmony if the sound is to be pleasing. You will notice, for example, that a string quartet is not made up of four violins, though such a group would be fully capable of melody and harmony; not only is a viola added to the two violins for its darker tone color, but a cello, which can go down to two octaves below middle C is also added, largely to furnish that feeling of solidity and lack of constant shrillness which only a good foundation of bass tones can give. In a symphony orchestra and many popular bands bass is furnished by the doghouse (bass viol or double bass), which goes still an octave lower.

The requirement for bass is difficult to justify on strictly physical grounds. It would be appropriate, however, to say that its need is not surprising, for it furnishes a kind of balance—and balance is characteristic of all natural phenomena. Melody tends to be relatively high in pitch, and accompaniments are usually around the middle range. Providing bass satisfies our emotional need for a feeling of balance. That this need exists can be confirmed by anyone who has ever listened to music through an audio system or in a concert hall lacking bass. It is tiring and unsatisfactory.

This need for balance also, by the way, extends to the treble end of things. You will notice in a symphony orchestra or in a dance band that the sharp treble effects are equally in existence, coming from flutes and various brass and reed instruments. An instrument or ensemble which has no high, piercing tones is unsatisfactory and boring, however good a bass it may have.

This matter of balance is so basic to the ear that a "frequency product" was worked out long ago in audio engineering to take care of it. The application was the frequency response of audio reproduction systems. The idea says that if your system cannot reproduce the entire range of audible sound because one end is deficient, the other end must be curtailed to balance it. The magic figure is 400,000. For example, if the lowest available bass tone is 40 cps, the treble response must end at 10,000 (40 x 10,000 = 400,000). Or if bass goes down only to 100 cps, treble should end at 4,000 cps. If this theory is adhered to, an audio system, while it will not be "high fidelity," will not sound shrieky and unpleasant. But if the treble is allowed to go too high with respect to the bass, the *unbalance* will be hard to listen to. This is, in fact, exactly the situation with the majority of present home music systems. It is easy to obtain extended treble response, but with the widespread insistence on small speaker systems and the resulting lack of solid bass, the total effect is unbalanced toward the high end. No wonder most women, with the usual female special sensitivity to high-frequency sound, don't care much for "high-fidelity" sound reproduction. Unfortunately, the fault is mostly theirs for refusing to give houseroom to speaker systems of a size that can reproduce the essential low bass!

The same balance problem works the other way as well. If, for instance, the system gives full bass response to 30 cps and the treble only goes up to 4,000, the music will sound dull and flat. If, however, the bass is restricted so it only goes down to 100 cps, the over-all result is much more acceptable to the ear.

The philosophy extends just as well (though not necessarily with the same numbers) to musical instruments and ensembles. In a symphony orchestra the availability of very low bass makes use of some very high-pitched, high-harmonic-content instruments feasible. A string quartet has available only bass an octave higher than that, so the fact that the highest-pitched instrument is a violin makes for reasonable balance. A quartet composed entirely of cellos is not really practical (though I know of at least one that did exist) for balance reasons—no sufficiency of treble.

The reason for this rather lengthy prerogation is that a number of electronic organs on the market, especially in the lower-priced group, avoid higher pitched sounds and tone colors which are sharp, like reeds and strings. This results from technical deficiencies or economies, and from noting that musically untrained people (who are often organ buyers; many of them become quite musical after learning to play the organ) shy away at first hearing from any but the most syrupy, hooty sounds. Such organs may not frighten anyone, but neither have they the balance necessary for satisfactory musical expression over a period of time. They are just dull. Among the ideas I hope you will get from this book is the one that bold voicing pays off in terms of better instruments—assuming, of course, that the blander and bassier voicing is also there in proper balance.

Tone Color

Each of the instruments of the orchestra has its own characteristic sound or timbre* so that the orchestra is really a palette of widely varying tone colors from which the composer may choose for an enormous variety of effects. While tone color is not an essential element of music, most music would be pretty dull without its variations, and some would lose its effect entirely if played on a different instrument from the usual one.

Unique among instruments meant to be played alone, the organ, both pipe and electronic, has the facility of sound in many different tone colors, using one at a time, one for melody and another for accompaniment, or several at the same time for both. The organ was probably first referred to as the "king of instruments" about 600 years ago by Guillaume de Machaut, and probably because it was capable of such loud noises. It remains the king of instruments today because of the variety and subtlety of its many voices.

Chapter 5 is entirely devoted to the theory and practice of obtaining all these tone colors. Suffice it to say here that tone color variation is a normal and desirable element of nearly all music, and an absolute *sine qua non* of an organ.

Dynamics

Dynamics, in musical terminology, is the word which generally covers all variations in both speed and loudness during performance. Since speed is a purely

*You have my permission to pronounce this word like the warning for a falling tree rather than fooling with the awkward French.

musical matter controlled by the player, we shall take dynamics to mean only loudness variations.

We may very well say that a military band is loud and a string quartet relatively soft, but the absolute loudness or softness of an ensemble or instrument does not really very much affect our enjoyment of it. In a concert hall or a dance emporium there may be a difference in sound level of as much as 20 db between the front row and the peanut gallery, but auditors in both places derive just about as much (or as little) enjoyment.

What does matter is the moment by moment *variations* in loudness which the musicians make as they play. The precision of their control over loudness and the taste and expressiveness with which they employ it is one of the most decisive factors which separate the musical men from the boys. No music is satisfactory if it is played at constant volume all the time—with the possible exception of rock 'n' roll, which seems to me to be much too loud *all* the time and apparently satisfies somebody.

The string player varies volume by his bow pressure and velocity; the wind player with his breath. But the organ is much more of a "machine" and the organist (except when playing a tracker pipe organ) is completely separated from the actual sources of tone by the mechanism. The organ must therefore have other means of conveying the nuances of expression inherent in dynamics. It has two ways: selection of voices, which vary in inherent sound power, and a foot operated volume control known as a *swell shoe.*

Musical Frequencies

One aspect of music is subject to rigid, nonsubjective control, and indeed *must* be so controlled in an organ. This is the pitches or frequencies* of the musical tones which are selected when keys are played.

Any organized system of music requires an organized series of pitches from which selection must be made. Through the world's musical history there have been many different scales (as a standard succession of pitches is called). In those parts of the world which are outside what might be called the area of Western European culture several scales other than our own are still used, and some of them have a great many more available pitches than ours; that is, the differences or *intervals* between succeeding pitches are finer. Even in our own culture a few maverick composers, and even whole schools of them, have used nonstandard scales

All keyboard instruments, however, are tuned to the *tempered* scale, which was first popularized in the time of Bach. In the tempered scale, only octaves have exact small-number frequency relationships; all other intervals have been fudged so that a C#, for instance, and a D-flat can be played with the same key rather than requiring two separate keys which would complicate the keyboard immensely.

The tempered scale is constructed on an entirely logical basis. As we have

*Strictly speaking, pitch is a psychological phenomenon and frequency physical. I have, however, applied for and received permission to use the two terms interchangeably.

seen earlier in this chapter, two tones having a 2:1 frequency relationship (an octave apart, that is) bring to the ear a peculiar sense of identity. One automatically feels that the higher tone is the "same" as the lower one except that its absolute pitch is higher. Whether this identification is instinctive or conditioned I do not know; but people without any training in music and with just a bare understanding of the requirement can almost always say when two tones are an octave apart.

The octave or 2:1 relationship, then, is the basis of the scale. Please note that musical frequency differences or intervals are always a matter of *ratio.* The ear hears the same identity between 150 and 300 cps as between 440 and 880. The arithmetical difference in frequency is of no importance. This leads to the correct conclusion that *all* matters of musical pitch are in fact matters of ratio, except for those few unfortunate beings who by a psychological quirk have the "gift" of absolute pitch. Such people can identify an exact frequency; they also get quite upset when the frequency is off, and since few instruments are tuned exactly correctly for very long, they suffer often.

For most people, however, it would be adequate to sound a middle A at 400 cps or 500 rather than the International Standard of 440—provided that all other pitches of the instrument were tuned to the correct *ratios* with that 400 or 500. Naturally, too much of a deviation from standard pitches would make the whole thing sound too high or too low, especially to someone accustomed to hearing or playing music.

The drawing of Fig. 1-1 shows one octave of a standard keyboard plus a C key at the top which is a repetition, an octave higher, of the beginning C. The

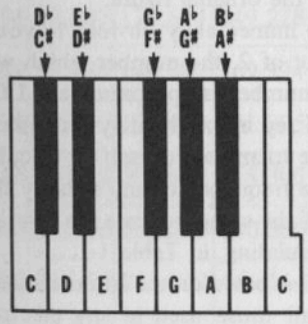


Fig. 1-1—One-octave section of a standard keyboard identifies names of notes.

term *octave* emanates from the Latin *octavus,* eighth. On the keyboard it is the eighth white or *natural* key which in the Western scale sounds the octave or doubled frequency of the first.

If we now count the number of keys within the octave, including both the whites and the blacks, but omitting the eighth white key because it is a repetition of the first, we will find 12 different notes. They are easily identified because the keyboard is one of the earliest surviving examples of what we call now human engineering—design for ease of human recognition and use. The pattern of the black keys is two and three, and this continues all the way up and down a keyboard. The note arbitrarily labeled C is the white just below (to the left of)

the group of two blacks. The D is the one *between* the two blacks. The B is just above the group of three blacks. And so on, each key always identifiable by its position with respect to one of the black groups. These identifications are learned with astonishing alacrity by any beginner.

The letters are applied only to the white keys, and since there are only seven different white keys, only the letters A through G are used (in English). Each black key is identified by reference to its white neighbors. The first black of the two-group is called C# (C-sharp) because it is just *above* C. The *same* black key may also be called D-flat (see symbol for flat in drawing) because it is just *below* the D key. Which term you use for identification is of no physical importance. It does matter in musical theory, but that is, as the saying goes, beyond the scope of this book.*

We have said that a keyboard, no matter how long, is always divided into repetitive octaves like that of Fig. 1-1. We have also said that the important factor in relationships between pitches is their frequency ratios. Finally, we can see from Fig. 1-1 that there are 12 frequency or pitch intervals in the octave–C to C#, C# to D, D to D#, etc., ending with B to C. In deciding what actual frequencies to assign to each note, the basic premises are (a) that the *interval* or apparrent pitch *change* from one note to the next must always seem the same regardless of which two notes we are talking about, and (b) no matter where we start, the last note at the end of the 12th interval must be exactly twice the frequency of the first note. To do these things we must find a constant factor by which we can multiply the frequency of any note to obtain the next one. The constant factor will make all intervals seem to the ear to be equal. And after 12 such multiplications the result must be twice the original figure.

The multiplying factor, which will be immediately obvious to you if you are mathematically inclined, is the 12th root of 2, the number which when multiplied by itself 12 times equals 2. The number is approximately 1.05946309. The number is irrational, which is why I say approximately, but that should be quite enough places unless you would like to amuse yourself by calculating more some long winter evening. Multiplying the frequency of any note by this number gives the next, and all the frequencies of the standard scale are now laid out for you, so you will not need to do any calculating, in Table 1-1.

Table 1-1 gives the standard frequencies, based on the International Standard of A-440, for eight octaves, covering all those used in any electronic organ (except the very few models with 32′ stops an octave lower than shown) and all other musical instruments. Just how organs are tuned to exactly these frequencies is explained in Chapter 8.

I should like to emphasize again that satisfactory tuning of an instrument is not a matter of certain exact frequencies but of ratios between frequencies. Table 1-1 is based on middle A at 440 cps, but that is by no means the only standard the world has used or does use. Many orchestras prefer the slightly

*Sharp designations are generally used in writing, simply because the number sign found on typewriters and in printers' fonts is a close approximation to the musical sharp sign, while nothing is usually available that looks like a flat sign.

brighter effect obtained from a standard A a few cycles higher and deliberately raise it. When they do so, naturally all other pitches are tuned correspondingly higher. In every case, however, except for players who have a tin ear, the pitches are related by the factor of the 12th root of 2, and the tuning still sounds correct. A few centuries ago it was common to use standard pitch very much lower than A-440–practically a half-tone lower, in fact. But whatever the standard, correct keyboard tuning still relates adjacent notes by the same factor.

Pitch Registers

When we play a certain key on the piano we expect only one pitch to be heard (we are not talking about harmonics of the string), and we know that no other pitch can be obtained except by striking another key. In the organ, on the other hand, a particular key or pedal is capable of giving two or more different pitches in response to settings of the controls.

If we press the middle C key on an organ and we hear the same pitch as we

TABLE 1-1: MUSICAL FREQUENCIES

NOTE	OCT 1	OCT 2	OCT 3	OCT 4	OCT 5	OCT 6	OCT 7	OCT 8
C	32.703	65.406	130.813	261.626	523.251	1046.502	2093.005	4186.009
C#	34.648	69.296	138.591	277.183	554.365	1108.731	2217.461	4434.922
D	36.708	73.416	146.832	293.665	587.330	1174.659	2349.318	4698.636
D#	38.891	77.782	155.563	311.127	622.254	1244.508	2489.016	4978.032
E	41.203	82.407	164.814	329.628	659.255	1318.510	2637.021	5274.042
F	43.654	87.307	174.614	349.228	698.456	1396.913	2793.826	5587.652
F#	46.249	92.499	184.997	369.994	739.989	1479.978	2959.955	5919.910
G	48.999	97.999	195.998	391.995	783.991	1567.982	3135.964	6271.928
G#	51.913	103.83	207.652	415.305	830.609	1661.219	3322.438	6644.876
A	55.000	110.00	220.000	440.000	880.000	1760.000	3520.000	7040.000
A#	58.270	116.54	233.082	466.164	932.328	1864.655	3729.310	7458.620
B	61.735	123.47	246.942	493.883	987.767	1975.533	3951.066	7902.132

Octave numbers throughout this book refer to columns above. Octave 1 is lowest 16′ octave of pedals or standard manuals. Tempered scale shown is based on A-440 (International Standard). Half-tone factor is 12th root of 2, approximately 1.05946309.

would when pressing middle C on a piano (261.6 cps) we are said to be playing in the *8-foot* (written *8′*) *pitch register.* This terminology is taken from the pipe organ; the open pipe which produces the lowest note for a 5-octave organ keyboard when playing in the 8′ pitch register is 8 feet long.

If we press the *same* middle C key and hear a tone an octave *higher* than middle C (523.3 cps) we are playing in the 4′ register. If we hear a tone an octave *lower* than "normal" we are in the 16′ register; and so on. An organ keyboard usually makes available at least two pitch registers and sometimes many more. In most cases each organ voice "speaks" in a particular register. If we turn on the Diapason 8′ we hear diapason tone at normal or *unison* pitch. If we turn on a Flute 4′ we hear flute tone an octave higher than normal from any key. And of course we may call upon all the voices and pitch registers singly or simultaneously as we wish, so that at any time pressing a single key may sound several octavely related pitches and several tone colors. There are also pitch registers which are not related by octaves; a "fifth-sounding" pitch register such as the 2⅔′, for instance, sounds the G an octave and a half higher when we press a C key. These are known as *mutation* pitches and are less common than octave-sounding pitches. (The actual sound in an organ equipped with a special mutation rank of generators is the third harmonic of the C, which is slightly sharp of the G, in our example. When the G is actually used, there being no special rank of generators, the mutations are said to be *borrowed.*

In working on and thinking about any organ there is always a practical problem of relating keys, pitch registers, and tone generators. We have therefore prepared Table 1-2, which eases the relationship. The normal organ keyboard has

TABLE 1-2: KEYBOARD VS GENERATOR OCTAVES

KEY OCTAVE	GENERATOR OCTAVES 16′	8′	4′	2′	1′
1	1	2	3	4	5
2	2	3	4	5	6
3	3	4	5	6	7
4	4	5	6	7	8
5	5	6	7	8	8
Top C	6	7	8	8	8

A key octave runs C through B. Octaves of short manuals to be counted as if all 61 keys were present. Example of use: What generator supplies tone for 3rd G key in 4′ register ? G5 generator; frequency 783.9 (from Table 1-1).

61 notes, or five octaves plus a top C. Each octave is assumed to begin with C and end with B. We also assume that the tone generating system of the organ has eight octaves of tone. Each generated octave is referred to by the number heading the column in Table 1-1. That is, the lowest generated E, for instance, is in Octave 1 and is at 41.2 cps.

To find out, then, the frequency of the E obtained when using the lowest E key (the E in Key Octave 1, Table 1-2) in the 16′ pitch register, we look down the 16′ column and across the Key Octave 1 column, finding the numeral 1. This means the E tone in question is in Octave 1, Table 1-1, and is 41.2 cps. Another example: What is the frequency required for the third G key in the 4′ register ? Looking down the 4′ column and across the Key Octave 3 column, the numeral is 5. Looking down the Octave 5 column in Table 1-1, we find that G is 784 cps. (Note that we are rounding off the frequencies shown in Table 1-1 to numbers much handier to deal with. We shall do that from now on.)

Many organs, especially smaller ones meant for the home, do not have full 61-key manuals. In using Table 1-2 you must then treat what does exist as a section of a 5-octave keyboard. To help you do that, simply remember that the middle C key is the start of Key Octave 3.

Also, most organs do not generate as many as eight octaves of tone. Where, for example, using Table 1-2 would indicate a tone in Octave 7, but the organ only generates tones through Octave 6, the tone actually used is simply the highest one available of the particular note—in this case Octave 6. The result is that the highest octave or two of a 4′ or 2′ or 1′ register may sound the same notes as the previous octave of keys.

Terms Concerning Pitch

Certain words used by musicians and in physics recur throughout any discussion of musical instruments. You may not know some of those in the following glossary.

Half-tone or **semitone.** The interval between any note and the next above or below it, whether black or white—for instance between C and C# or between F and E. Do not confuse a half-tone with *half-note,* an indication of timing (or *halftone,* a photograph rendered as a dot pattern for printing purposes).

Whole tone. The interval between any note and its neighbor next but one—for instance between C and D or between F# and E; in other words, two semitones.

Frequency. The repetition rate, in cycles per second (cps) of an a.c. wave, sine or complex, or the repetition rate of a sine wave to which a particular harmonic would be equivalent if the complex wave of which it is a part were rendered as a Fourrier series. A purely objective physical phenomenon not dependent on human perception.

Pitch. The aural sensation induced by the fundamental frequency of a tone. A psychological phenomenon which can be modified by factors other than actual frequency, such as loudness, motion, comparisons, etc. In this book we assume throughout that perceived pitch is the same as physical frequency and so use the two words interchangeably just for the sake of variety.

Fundamental. The lowest frequency present in a complex tone (one containing a series of harmonics); the repetition rate of any waveform. This almost invariably is the pitch as perceived by the ear.

Harmonic. Any frequency which is an exact integral multiple of the fundamental frequency of a complex tone. The fundamental is the first harmonic.

Overtone or **partial.** A component of a complex tone which may or may not be an exact integral multiple of the fundamental. Stretched strings, for example, have "harmonics" which are sharp. Bells of certain kinds contain overtones not related in any simple way to the fundamental.

Subharmonics. Integral submultiples of the fundamental. The word is sometimes bastardized as in *subthird,* meaning an octave below the third harmonic.

Sharp and **flat.** As nouns, the next note, white or black, above (sharp) or below (flat) any white key. As adjectives the words are used to indicate a pitch somewhat lower or higher than some reference. An out-of-tune note, for instance, may be said to be a little flat, meaning it is a little lower in frequency than it should be. Also can be used as verbs, correct forms for which are to *sharp* or to *flat* (to raise or lower pitch), not *sharpen* and *flatten.*

Chapter 2

What An Organ Is

If you are familiar with organs and their functional parts you need not read this chapter. Most people, however, have little opportunity to learn about the organ unless they own one, and the instrument is sufficiently different from all others to require some special knowledge, speaking strictly from the musical and visual standpoints and not of electronics.

Fig. 2-1 is a photo of one full size electronic organ, with numbers calling out the important parts so that as we discuss the parts you can identify them. All organs of full size have these items in roughly the same places, and smaller organs are not much different except that controls are usually on the blocks at left and right of the keyboards (the *cheek blocks*) and the pedals are shorter and hinged under the console.

The wood case which houses the organ is the *console*, regardless of the size or type of organ. The term "console model" has no meaning for organs, even though in pianos it refers to a small upright. A small organ is usually called a *spinet*, but there is no well defined term for one which is not small. I use the term *full size* to describe an organ which is of the form standard in pipe organs—two or more 61-note keyboards, called *manuals*, 32 (or sometimes 25) pedals pivoted behind the player (the *pedal clavier*), and stop controls above the upper manual or, not very often, drawknob controls on wings or *side jambs* at either side of the organist. The "full size" designation has no reference to the musical specification of the instrument, though of course full size organs are almost invariably larger musically as well as physically than spinets and in-betweens.

A third type of console is the theatre style, like the ones in Figs. 10-1, 11-1, 12-3 (a cross between a spinet and a theatre organ), and others. It is characterized by a curved stop board over the upper manual.

In the following text the bold-face numerals refer to the callout numbers in Fig. 2-1.

Manuals

The manuals of an organ **(1, 2)** are just like the keyboard of a piano or harpsichord when you look down on them from above, and anyone accustomed to playing a piano has nothing new to learn about key locations and interval distances when coming to the organ. An end view of an organ keyboard, however, is different from that of a piano. In Fig. 2-2 A is the profile of a piano keyboard; note the squared-off end of the white key and the horizontal top of the black. In B, an organ manual, the downward slope of the black and the front shape of the white have good reasons. They make it possible to insert a second manual C

below the first in such a way that the upper somewhat overhangs the lower, bringing the two closer together vertically and laterally than they could be if piano keys were used. This helps the player when he jumps a hand from one manual to the other and when, as sometimes happens, he uses different fingers of one hand on both keyboards.

The keys of B and C, described as *overhanging,* are standard for pipe organs and they are used on all full size organs except for some Hammonds. Smaller organs often have piano-style keys, presumably because they are cheaper, or, once

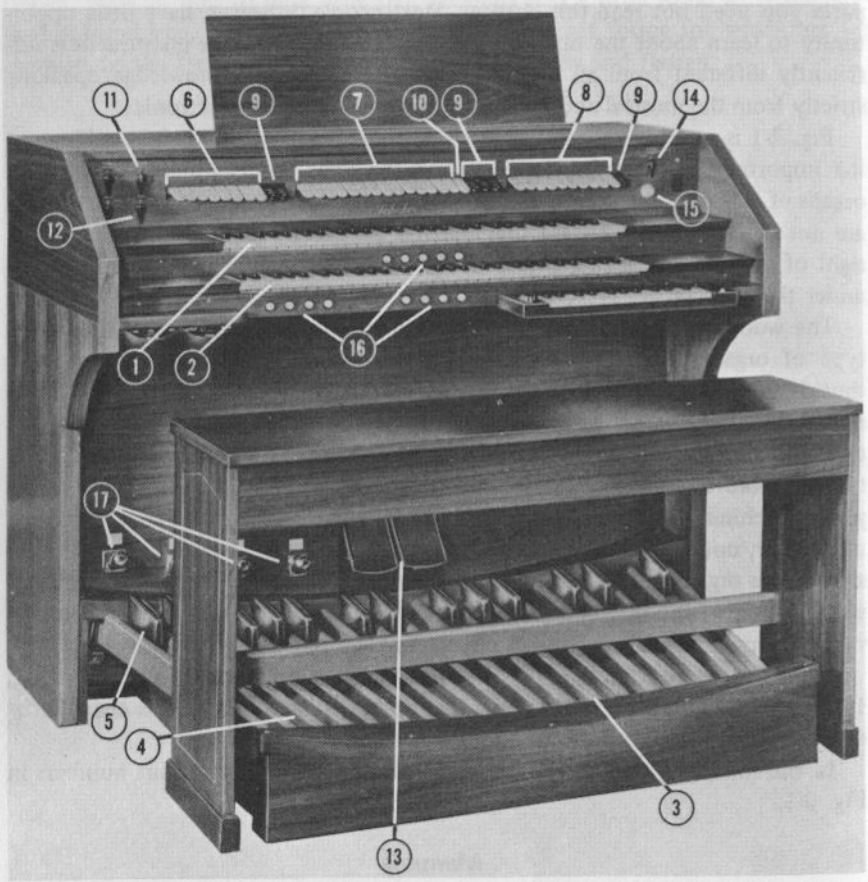


Fig. 2-1–A typical "standard" organ (the Schober Recital), with parts called out: (1) Swell or Solo manual; (2) Great or Accompaniment manual; (3) pedal clavier; (4) a "white" pedal key; (5) a pedal sharp (black key); (6) pedal stop tablets; (7) upper-manual and (8) lower-manual stop tablets, the two interchanged sometimes on theatre organs, all tabs (of a different type) located on cheek blocks at left of one or both manuals on most small organs; (9) coupler tablets; (10) vibrato tablet; (11) vibrato depth control; (12) control for imparting vibrato to either manual or both; (13) swell shoes; (14) percussion selector knob; (15) reverberation control; (16) combination action buttons or "pistons" and (17) toe studs. Not all organs have these controls and some have others. Placement varies, too.

in a while, a "waterfall" style in which the ends have a slight radius instead of being squared off.

Traditional keys are made of wood, the playing surfaces of the whites covered with ivory or plastic, and the blacks ebony or a cheaper black-dyed wood glued

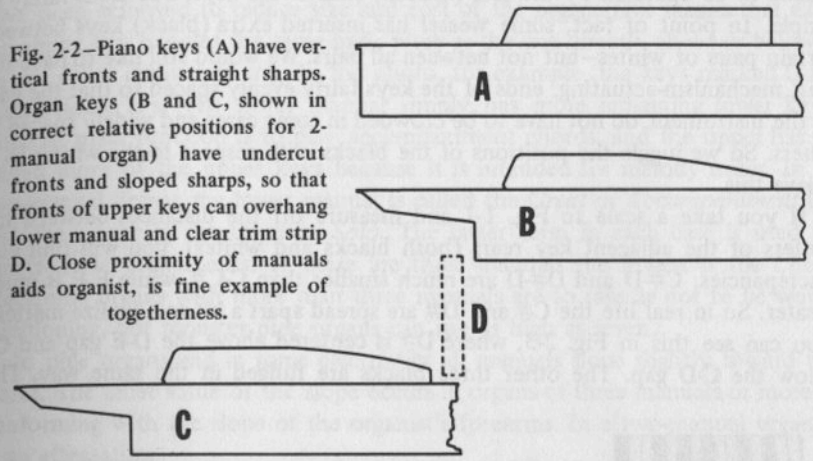


Fig. 2-2–Piano keys (A) have vertical fronts and straight sharps. Organ keys (B and C, shown in correct relative positions for 2-manual organ) have undercut fronts and sloped sharps, so that fronts of upper keys can overhang lower manual and clear trim strip D. Close proximity of manuals aids organist, is fine example of togetherness.

to ordinary wood bases like the whites. Modern keyboards, however, usually have plastic blacks on top of the wood sticks; they are cheap, won't break, and can't discolor or roughen.

A set of wood keys, by the way, is made by taking a slab of wood as big as the final keyboard and gangsawing out the keys. The quality of a wood keyboard is very much dependent on the quality and seasoning of the wood. Warping is far from uncommon.

That is one of the reasons for going to other materials, as several manufacturers have done. Some employ simple molded plastic keys mounted by flat springs and sometimes other elaborations to a frame. One of the best and most elaborate is made by Pratt, Read & Co., Inc., Ivoryton, Conn., who are also the world's largest makers of wooden keys. It is functionally of heavy steel and I have described it in Chapter 10.

The lateral separation of keys is pretty well standardized, based on an octave length of slightly less than 6½ inches. The real figures range from about 6.457 to 6.464, depending on the manufacturer, but these differences are too small to be noticed in playing.

A keyboard does have one intriguing peculiarity which is not obvious to most people until pointed out. If you will turn back to page 9 and look at Fig. 1-1 you will probably see nothing amiss. However, if you compare it to Fig. 2-3, an actual photo of some keys, you will see that the spaces between adjacent white key fronts do not necessarily coincide with the centers of the black keys, which they all–erroneously–do in Fig. 1-1 because it is easier to make a drawing that way. The reason for the discrepancy is that several requirements must be met at the same time.

First, there are seven white keys within an octave and their front playing surfaces should be of equal width so the organist will not have a nervous breakdown. If no black keys were needed this would simply mean that the seven switches, sets of strings, or whatever makes the instrument go, could be equally spaced to be operated by the rear (hidden) ends of the keys. But life is rarely so simple. In point of fact, some weasel has inserted extra (black) keys between certain pairs of whites—but not between all pairs. We would still like to have the rear, mechanism-actuating, ends of the keys fairly evenly spaced so that the guts of the instrument do not have to be crowded in some cases and widely spaced in others. So we juggle the positions of the blacks with respect to the whites to achieve this.

If you take a scale to Fig. 1-1 and measure off the distances between the centers of the adjacent key rears (both blacks and whites), you will find vast discrepancies. C#-D and D#-D are much smaller than C-C#, while E-F is much greater. So in real life the C# and D# are spread apart a bit to equalize matters. You can see this in Fig. 2-3, where D# is centered above the D-E gap and C# below the C-D gap. The other three blacks are fudged in the same way. The



Fig. 2-3—Actual photo of some keys. Note different dimensions of the white-key cutouts where they go around the blacks. Every white key is different.

white fronts are kept at uniform width and spacing and so are the blacks, but the end result is that each black is a slightly different distance above or below the adjacent whites. This difference does not bother the player because the black notes are known in music as *accidentals,* meaning that if he hits the right one it is an accident anyway.

There is only one unfortunate byproduct of this exhibition of cleverness. It is that each of the white keys has a slightly different shape and they are not interchangeable. As a result, any manufacturer who tools up to make his own plastic keys requires a total of nine different molds—one for each of the seven white keys A through G, an eighth for the top C of the manual, which has no indentation because there is a white key below it and no keys above it, and a ninth for the black keys, which are all (thank God) the same. The tooling bill usually comes to a fat five figures.

Methods of pivoting, springing, and upstopping the keys vary with manufacturer and require no enormous imagination. The AGO (American Guild of Organists) standard console specifications given in the Appendix call for a depth of touch of 5/16 inch to "3/8 scant," a woodworking term which always leaves me, with my engineering habits, gasping. Electronic organ manuals generally conform to this depth of motion (scantily) but they sometimes have a lighter touch than the 3¼ to 3¾ ounces the AGO mentions.

The standard organ manual, always present in full size organs, has 61 keys, C to C, representing five octaves of 12 notes each, plus a top C. Small organs more

often have short manuals, 44 keys F to C being most common. Short manuals can confuse momentarily because their placement is almost always "offset" as in Fig. 2-4. The confusion is eliminated if you will forget about the "offset" and consider each manual as a regular 61-note keyboard from which some end keys have been removed to reduce size and cost of the organ. This means that each remaining key handles the same basic pitch as the one directly above or below it on the other manual, if any. In the photo, for example, the keys marked C are both middle C's. The lower manual simply has more remaining lower keys, since its main function is to play accompaniment chords, and the upper has retained more of the upper keys because it is intended for melody notes. In all two-manual organs the lower manual is called the *Great* or *Accompaniment* (2) and the upper is the *Swell* or *Solo*. The latter term in each case is used on theatre-style instruments. If there are three manuals the lowest is the *Choir*. Electronic organs with more than three manuals are so rare as not to be worth mentioning, but monster pipe organs can run as high as seven.

In pipe organs and in some electronics all manuals slope slightly toward the Great. The chief value of the slope occurs in organs of three manuals or more in conforming with the slope of the organist's forearms. In a two-manual organ it is an affectation.

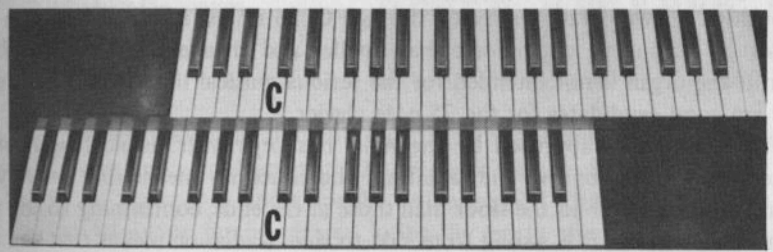


Fig. 2-4—Typical "offset" spinet manuals. Pitches of keys in a vertical line match, like the two middle C's shown.

Why should an organ have two or more manuals when the finest Steinway piano has only one? It is because the organ is a multivoiced instrument and at any moment the organist arranges matters so different voices or combinations of them are called forth by each manual. He may then play the melody on the manual having the more prominent voices and the accompaniment on that with the more subdued ones; or he may play entirely on one manual, then immediately shift to another to obtain a total change in tone color without delay.

The function of an organ key (and this is true of most pipe organs as well as electronics) is simply to close one or more switches which are actuated by one of many possible mechanical means and may be located under, behind, or above the visible or invisible parts of the key. In the electronic organ these switches cause signals of the correct frequencies to appear and to go to the desired circuit points. An organ key has no "touch" in the piano sense; it makes no difference whether the key is pressed hard, softly, or at high or low velocity. The only sig-

nificant exception to this rule is that a few organs employ switches in which one element is a plastic impregnated with a conductive material, so that pressure of a key-actuated finger varies the resistance of a contact. The resistance variation is limited and has a minor effect on signal transmission, and the organist is not meant to have control over tone loudness by controlling finger pressure. An actual variable-resistance keying system was used by Baldwin (intended, like the modern ones, merely to avoid too abrupt onset of tone, not as a control of loudness), but these switches are obsolete. Some theatre pipe organs did have *second touch,* a method of creating added effects on selected tones by pressing the key farther down than normal, against greater resistance, but I do not know of any manufacturer who uses second touch in electronics. This is not to say it might not be a fine idea, and I would not be surprised if it appeared before long, especially with the resurgent interest in theatre-type instruments. In any event, control of loudness is obtained by other methods.

Pedals

The *pedal clavier* **(3)** is actually a keyboard just like the manuals except for its Brobdingnagian dimensions. It always begins with low C, and it consists of black **(5)** and white **(4)** notes in the same order as the manual keyboards. A full size organ has 32 pedals if its pedal clavier is standard, two and a half octaves, C through G. A two-octave clavier (C through C, 25 notes) is not uncommon as a means of making an organ somewhat smaller and cheaper, and there is little harm if the organ is not intended for the serious musical literature. Theatre organs are good candidates for the 25-note clavier.

The pedal board approved by the AGO has keys with visible length of 27 inches and is both concave and radiating. The concavity means that the keys at the middle are closer to the floor than those at the ends, conforming to the position of the foot as it is swung from side to side. Radiation means that the heel ends of the pedals (the ends not under the organ console) come together toward the center, conforming to the line of the organist's foot as he pivots his legs from the hips. The details are given in the AGO Specifications in the Appendix. Less elaborate organs have flat claviers which are not concave, but all claviers having this type of pedal pivoted at the organist's rear do radiate. The concave pedals do not date from antiquity and standardization is comparatively recent. Except where the pedal work is fairly complex, as in the music of the masters, especially Bach, I doubt that a flat pedal clavier is much of a disadvantage.

Fig. 2-5 shows a different kind of pedal clavier which is common on smaller organs. The pedals are comparatively short and are pivoted underneath the console, as you can see by looking at any of the photos of spinet organs in this book. The usual spinet organ has 13 of these pedals, C to C, though a few more elaborate small organs such as the one in Fig. 10-3 have more. Such pedals are, of course, much cheaper than the standard type, and they contribute to the appearance of a small organ which is supposed to be compact.

The initial function of organ pedals is to provide foundation bass and rhythm, and after a surprisingly brief learning period they actually, contrary to the novice's expectations, help make the organ easier to play. Taking our overworked waltz again as an illustration, you get the *um-pah-pah* on the piano by playing a low note for the *um*, then dashing an octave or two upward to play chords for the *pah-pah*, your accuracy in this broadjump dependent on your skill and luck. On the organ you can find a pedal with your left foot and the chord with your left hand. Then, without moving, you can hit the pedal for the *um* and the chord for the *pah-pah*, a considerably less hazardous mode of operation.

The pedals are used in more advanced music, of course, in much more versatile ways, and can easily be compared to the double-bass or cello sections of an orchestra. Perhaps the zenith of pedal technique is required for polyphonic works like those of Bach, in which complete parts are played on the pedals against others on the manuals, sometimes at incredible speed. A good organist playing a Bach fugue is about the busiest man in town.

It is not nearly so awkward to play complex music with the feet as you might imagine, though you don't learn to do it in a day. A heel-and-toe technique is used for *legato* passages (those in which note follows note without a break).

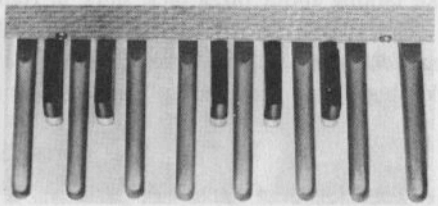


Fig. 2-5—Typical spinet pedals, pivoted under the console.

The heel presses one pedal and the foot twists at the ankle while the pedal is down. The next note is played with the toe, which has been placed above it during the twist, and so on. And since the well equipped organist has at least two feet, he can negotiate jumps without losing the *legato* effect by alternating the feet when necessary. The skilled musician does not have to look at the pedals while he does this, but most pipe organs have a light under the keyboards illuminating the pedals, from which we may infer that he does look at the pedals sometimes even though he is not supposed to. (This is like the *No Smoking* sign in my dentist's office, located 3½ inches from an ashtray.)

These fancy shenanigans are really possible only on a standard pedal clavier and not on a spinet type like that of Fig. 2-5. A few spinet makers provide somewhat longer pedals of that kind and claim they can be used for heel-and-toe work, but that is largely a delusion. Spinet organs cannot normally be used anyway for serious music because of their short keyboards and inappropriate voicing. For the popular, standard, and hymn music played, simple pedal work with the left toe is appropriate and adequate, and that is about all you can do with such pedals. *Toe pedals* is what these appurtenances are generally called (unofficially) and the term fits.

As you would expect, the pedals are tonally independent on most full size organs, so that you can play all 25 or 32 of them at once if you have 25 or 32 feet. Serious music calls for two simultaneous pedal notes once in a while. In less

elaborate full size organs, however, and in practically all smaller organs, the pedals are *monophonic* so that only one will sound at a time (generally the lower) if two or more are pressed at once. The reason for this lies in electronic design economy, but there is a playing advantage as well, especially for the novice, because humans cannot control their feet as well as their hands and this "interlocking" idea prevents the tonal mess which would be heard if two adjacent pedals accidentally pressed together could sound at the same time.

Most pedals are still made of wood, usually something hard like maple, finished with a clear protective coating. The blacks, like those of the manuals, are more often than not plastic. They are a great advantage as they cannot discolor, and I have yet to hear of one's breaking.

As a nut on English, I must take some space to beg you not to make two errors, one common and the other rare but funnier. Please do not use the phrase *foot pedals.* The word *pedal* comes from a Latin root which means foot, so addition of the English word *foot* is a redundancy, unneeded for clarity since you would hardly expect to find hand pedals or shoulder pedals. (It is something like talking about pizza pies, since in Italian the word pizza *means* pie.)

The second error was made by a company which apparently assumed that *manual,* which actually stems from a Latin root for *hand,* was a synonym for "keyboard." Its advertising literature carried references to the "pedal manual!"

Organ Stops

The principal characteristic which distinguishes the organ from all other instruments is that the player has a great deal of control over the tone color of its sound. In a pipe organ there is typically a separate complete rank of pipes covering several octaves of pitches for each different tone color or voice. In each rank the pipes are constructed differently to obtain the differing tone colors, and a rank in effect constitutes one complete musical instrument. The player can switch in one or several ranks for each keyboard. In the electronic organ the tone colors are created in a different way, but the playing effect is (or should be) the same.

Each voice is generally spoken of as a *stop* and each one is controlled by a *stop tablet.* Certain voices are available on the Great or Accompaniment manual and are switched in and out with one group of tablets **(8)**. Another group of tablets **(7)** controls an additional set of voices for the upper manual; and a third **(6)** controls a third set of voices available on the pedals.

The most common kind of stop tablet on full size organs is the *tongue tablet* like those in Fig. 2-1. A few of them are shown in closeup in Fig. 2-6. Each consists of a plastic tablet engraved with the name of the stop and its pitch register (see last chapter). The up position is off, and when all tablets for a particular manual are up pressing the keys produces no sound. When a tablet is flicked down, as is the Oboe 8′ in Fig. 2-6, that voice sounds when the keys are played. If two or more are pressed down, all the selected voices sound and their tone colors and pitches produce an additive effect in a well designed instrument. The tablets are mounted in a horizontal slot on a vertical wood piece known as a

stop board, with just the tablets protruding through the slot. Behind the slot is a mechanism for each tablet which (a) gives a very light toggle action so the tablet wants to stay up or down but not in the middle of its travel, and (b) one kind of switching or another to do the electrical job.

There are two other kinds of stop tablets. One is the *rocker,* commonly used on small organs. It is a flat-surfaced oblong of plastic, rows of which are let into cutouts in wood, usually on one or more cheek blocks as in Fig. 15-1 (and others). The rocker is pivoted underneath at its center so that it rocks back and forth when the front and rear parts of its top are pressed, actuating the switching mechanism underneath. Rockers mounted in cheek blocks eliminate the need for the vertical stop board above the upper manual and make a lower silhouette possible for the console. They are also often cheaper, especially in terms of assembly labor and replacement ease, as they can often be manufactured as integral parts of a whole electronic assembly (hidden beneath the wood) which includes electronic parts, then mounted as a unit in the console. They are not as easy to use as tongue tablets spread out over the upper manual, for the player

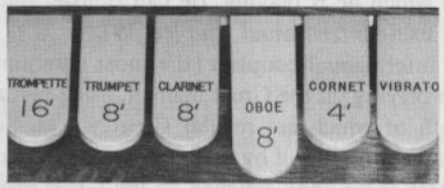


Fig. 2-6—Typical tongue tablets used on full size organs.

must always reach over to the left end of the organ for them. Sometimes, however, rocker tablets are spread out over the manuals as in Fig. 9-1, in which case they are almost as easy to use as tongues.

Drawknob stops like those visible on the jambs at both sides in Fig. 18-1 are uncommon in electronic organs, though they are very often found on larger pipe organs. Each stop is actually a knob which is pulled out to turn the stop on. They are not nearly so handy as tongue tablets, especially when the latter are at the player's fingertips above the manuals, and they take a great deal more space because they are relatively large (as well as expensive). My guess is that a few organists like them because they are reminiscent of large pipe organs and give a sneaky feeling of luxury.

The subject of voicing is a large one and Chapter 5 gives some basic information in preparation for the discussions of individual organs. Suffice it to say here that the number and assortment of stops for the organ and for each division (a division usually corresponds to one keyboard) knows no standard in either pipe or electronic work, and neither does the sound of any stop necessarily correspond to that of the same name on another organ. Voicing then is in the designer's province, and the principal judgement a musician makes of any organ is based on the quantity and quality (not necessarily in that order) of its stops.

Stop tablets which control voicing are usually white, with black fill in the engraving (though a few tablets are hot-stamped rather than engraved). However,

the theatre organ popularized a tradition of colored stops which is coming into more common use in electronic organs. The color system is useful because it helps the player identify stops of the four tonal families quickly—green for flutes or percussions, red for reeds, yellow or black for strings, white for diapasons and sometimes flutes. There is some variation from this. Whether the decor-minded ladies think colored tabs look too jazzy sometimes influences design and purchase, and personally I dislike those which are "marbleized," though I have no objection to solid colors. Colored fill is sometimes used in the engraving on formal organs, too, if only to the extent of filling the white tabs with red for the reeds.

Intermanual Couplers

Each of the organ keyboards has its own group of stops, which are different from those available on any other keyboard and can be used singly or in any combination. Sometimes, however, the organist wants to use the stops of one keyboard on another, in addition to those normally allocated to the keyboard on which he is playing. He can transfer the stops from one keyboard to another by using *intermanual couplers* **(9)**.

Intermanual couplers (the most common kind) work this way. Suppose you are playing on the Great manual, using perhaps an 8′ Diapason stop and a 4′ Flute both of which are regular Great voices. Suppose now that you would like to sharpen the sound by adding a reed, and the one you want to add is the Oboe 8′. There is no oboe assigned to the Great manual, but there is one available on the Swell. So you operate the tablet labeled SWELL TO GREAT 8′ as well as the Swell Oboe 8′. Now, as you play on the Great keys you will hear, in addition to the Diapason and Flute, the Oboe, borrowed from the Swell for the occasion.

The engraving on the coupler tablet has very definite and easily understood meaning. Swell To Great means Swell voices transferred to Great keys. When this tablet is on, any and all stops turned on in the Swell division will be heard when the Great keys are played. The 8′ designation does *not* mean that this applies only to 8′ stops. It means that any Swell stops selected will be transferred to the Great keys and will be heard in their own normal pitch registers. The Swell Oboe 8′ will be heard at 8′ pitch on the Great keys; the Swell Flute 4′ will be heard at 4′ pitch; the Swell Piccolo 2′ will be heard on the Great at 2′ pitch; and so on.

If the tablet were engraved SWELL TO GREAT 4′, it would mean that all Swell stops transferred to the Great keys would be heard *an octave higher* than normal—the Oboe 8′ at 4′ pitch, the Flute 4′ at 2′ pitch, and so on. A Swell To Great 16′ tablet would bring Swell voices to the Great keys an octave *lower* than their normal pitch registers.

We have used the word "transfer," but note that the voices "transferred" from the Swell division to the Great by the coupler will still be heard if you play the Swell keys; they do not pick up their feet and walk down to the Great—they split themselves like a paramecium and exist in both places.

Naturally, we may also have Great To Swell couplers that operate in the other direction, making available on the Swell keys any voices selected in the Great division. And there may be both Great To Pedal and Swell To Pedal couplers, so that manual voices can be added to those of the pedals. The variety and types of couplers differ on every organ model and are another means of adding versatility to the instrument.

In becoming familiar with the organ, the trick is to remember the exact meaning of the coupler nomenclature. The first word always indicates the division *from which* you are taking voices; and the second indicates the keyboard on which they will be heard. So Great To Swell means Great *stops* to Swell *keys*. The pitch indications on couplers are obvious if you know that 8′ is basic, normal, unison, and the starting point.

Normally couplers operate on all stops in the affected divisions. However the inner functioning mechanism is different on electronic organs from that of pipe organs, so it is often more sensible economically to limit this. For instance, a particular model may have only 16′, 8′, and 4′ stops on the Great, but 16′, 8′, 4′, and 2′ stops on the Swell. Usually on such organs if the Swell To Great 8′ coupler is used, the 2′ Swell stops are not transferred to the Great. Only those are transferred which are in pitch registers for which stops already exist on the Great. The reasons for this will become clear in Chapter 4.

Intramanual Couplers

The second kind of coupler, the *intramanual,* has its entire effect on one division or keyboard of the instrument. A typical one would be a Swell To Swell 4′. When it goes on, all voices of the Swell continue to sound in their normal pitch registers and *also* sound an octave higher. The 16′ voices, for instance, sound at 16′ pitch and also at 8′ pitch, so that if a single 16′ stop is on you can play a single key and produce the sound which otherwise would be heard only if you were playing two keys an octave apart. A Swell To Swell 16′ coupler would add to the normal pitches those an octave below normal. And so it goes.

As is true of the intermanual couplers, there is usually a limitation due to the capabilities of the instrument. If the normal stops of a manual are, for example, 16′, 8′, and 4′, a 4′ intramanual coupler will add 8′ tone to 16′ stops and 4′ tone to 8′ stops; but it will not add 2′ tone to 4′ stops because no 2′ tone is available from that keyboard. In the other direction, a 16′ coupler adds 8′ tone to 4′ stops and 16′ tone to 8′ stops, but no 32′ tone to 16′ stops because no 32′ tone is available. Also, if the organ has mutation stops with nonoctave-sounding pitch registers (fractional numbers like $5\frac{2}{3}$′, $1\frac{1}{3}$′, etc.) those stops are not affected by intramanual coupling, though they may be affected by 8′ intermanual couplers if the "receiving" keyboard has the necessary pitch registers.

A separate kind of intramanual coupler can be very useful; it is the Unison Off. When this coupler is turned on, it removes the normal pitches of all stops on the keyboard. Remembering that the normal intramanual coupler *adds* tone an octave or more higher or lower than the stop's own pitch, using one results in, say, both 4′ and 2′ pitches from a 4′ stop when the Swell To Swell 4′ coupler is on. If then the Swell Unison Off is used, the original 4′ pitch is removed, leaving

only the 2′, so that the total effect is not to add a second pitch but to *change* the stop's pitch (and that of all others on that keyboard) to one an octave higher. The Unison Off, used without any other intramanual coupler, can also be very handy as a fast way of killing all stops in case some separate effect such as percussion is wanted alone.

There are some electronic organs without couplers, usually smaller ones. Couplers are not absolutely essential equipment and an organ can be quite satisfactory for its size without them. Like many refinements with which an organ can be piled high, they add to value, but the organ still depends for its musical merit on a sufficient number of keys and an adequate number, variety, and quality of voices. Everything else is icing on the cake. How much icing you can afford to add depends on how much sugar you have.

Vibrato

There is almost no continuous-tone instrument which is only capable of producing "straight" tone of constant pitches. On almost all, the player can vary the pitch of each tone cyclically to produce what is known as *vibrato,* an effect which imparts a kind of warmth and drama. The most obvious example is the violinist who, as he bows a continuing note with his right hand, vibrates his left hand so that the finger which is stopping a string (determining its effective length) moves slightly up and down the string, varying its pitch. An equally obvious example occurs in singers, who almost never sing without vibrato. In fact, most people who cannot sing and are said to have "no voice" often have a very satisfactory voice indeed but simply lack the kind of muscular control necessary to create the essential vibrato. The rate of frequency variation in vibratos ranges from about 5 to 8 times per second and deviation (using the FM term) is around 3% as a maximum. (This is about a quarter-tone—half of a semitone—and is a very large vibrato indeed.)

In a pipe organ the effect is brought about by mechanically produced cyclic variation in the wind supply to the pipes. The result is correctly and invariably called *tremolo* (the mechanism and control are the *tremulant*), which by definition is a cyclic variation in *loudness.* Alteration in wind pressure does have an effect on pitch and thus contains an element of vibrato, but the principal effect is on loudness. There is no practical way to produce a purer vibrato.

Vibrato—frequency variation—is universally found to be more pleasant, however, which is why it is always used in electronic organs, where providing it is usually no problem since there are any number of ways of varying the frequency of electronic oscillations. A few electronic organs deliberately include a slight tremolo effect as well, but in almost all cases the use of the word "tremolo" or "tremulant" is no more than a polite bow to pipe tradition.

Vibrato is not always wanted, however; in fact it is frowned on in most classic music. There is, therefore, always at least one control which typically is a stop tablet (**10** and Fig. 2-6). There may also be other vibrato controls for depth, or intensity or both (**11**) and distribution of the vibrato effect to either manual or both (**12**). Some organs also have the vibrato speed control on the outside.

A few organs, like the Hammond with its rotating wheel generators, and the older Wurlitzers with their wind-blown reeds, have more trouble producing vibrato than the purely electronic instruments because the basic sources of tone are of fixed frequency. For both these instruments phase-shift systems were developed to add vibrato at a later stage, on the Armstrong phase-shift-equals-frequency shift principle. A phase-shift system is also used by Schober because its designers believe it sounds better, since the effect can be distributed to taste and upper harmonics need not be "vibrated" by the same percentage as fundamentals. A basic advantage of not applying vibrato to the oscillators of organs which must derive pedal and manual tones from the same generators is that vibrato can be omitted from the pedal tones. Vibrated bass tones are not especially pleasant, probably because the rate of vibrato is too large a percentage of tone frequency.

Expression Control

You will recall that we made quite a point of the element of dynamics in music back in Chapter 1. It is absolutely essential that the player have some good way of controlling the loudness of his sound and of varying it at will. The organ permits volume control in two ways.

The first method is the choice of voices from one moment to the next. In a well designed instrument voices are different from one another not only in tone color and pitch register, but also in volume. When playing melody against accompaniment with a pedal bass, the simplest form of *registration* (the word describing choice of stops for a particular piece of music or part thereof), the organist must use louder, or at least more penetrating, voicing for the melody than for the accompaniment, and must select pedal stops so that they furnish an adequate bass without overpowering the rest. Along the same lines, he may also allow the loudnesses of stops to influence his registrations for all purposes.

Selection of stops, however, is not a sufficiently fine, fast, or precise way of supplying continuing dynamics. Most pipe organs provide special chambers for some or all of the pipes with a sort of Venetian blind known as *swell shutters* over the opening through which the sound exits. A *swell shoe* on the console (**13**) can be rocked back and forth on its center pivot by the foot to control the shutters, each of which rotates on its axis so that the opening can be anything from fully clear to totally shut (but not actually acoustically wholly opaque).

On an electronic organ, the same kind of swell shoe is used but its operation is much simpler, since it can easily be geared or otherwise coupled to a volume-control potentiometer or some other electrical control device. In contrast to many pipe organs where at least some of the pipes are not in shuttered chambers and thus cannot be controlled, electronic organs almost invariably have all signals under control. In small organs a single swell shoe is the volume control for the whole organ. In larger ones like that of Fig. 2-1, there is often a separate swell shoe for each manual. with pedal-clavier volume also assigned to one or, as in the case of the Schober Recital Organ, assignable by a console switch to either shoe or partially to both.

A swell shoe remains where it is set after the foot is removed, unlike an automobile accelerator pedal which springs back to zero. That is essential because otherwise the organist would not dare to take his foot away for use on the pedals. In the extreme minimum position the shoe does not reduce volume to zero; its whole range is from maximum to around 20 db less. (There would be little sense to a zero-volume position, since that can easily be reached just by stopping playing!)

Too large a control range is also undesirable because the foot is not really capable of very fine control and too great a volume change per unit movement would bring danger of an unpleasant wavering effect. For the same reason it is important that the control be smooth–about the same volume change for each amount of apparent movement of the foot–which depends on the taper of the potentiometer or the functioning of whatever device is used. It is interesting that a potentiometer with an audio taper (the usual semilog employed for hand-controlled volume adjustment) is not suitable for swell shoes. When the foot is moved in the low-volume range (toe high) rather slight movements feel like large ones because the effort required is relatively great; when it is moved in the high range (toe down) much larger movements take place per unit of desire. To complement these tendencies we have found over the years that a linear potentiometer gives the right effect and the player feels he is getting practically uniform volume change over the range of foot movement. The swell shoes on some organs were designed strictly from book larning (not *this* book) with semilog pots, and as you play them you are quite disturbed to find that most of the (apparent) shoe movement produces little effect, while the last bit makes the organ suddenly blare.

It is also desirable to give the swell shoe considerable frictional resistance to movement so that slight motions of the foot, which, as we have said, is not the most delicate of apparatuses, do not make the volume shift.

Percussion And Sustain

The items we have discussed to this point are those which may be expected to appear on practically every organ. There are others, however, which appear when (and if) the engineering and sales departments think them desirable. This slightly snide approach stems from the fact that although none of the points to be listed are necessary, some of them are more sensational and more appealing for the moment to the unknowledgeable purchaser than the necessary and desirable attributes of an instrument. Unfortunately, that moment is generally the one in which the salesman is pitching, and too many devices have been designed for organs almost solely with that pitch in mind. As a result, there are organs in which the production money has been spent for relatively useless frippery, to the neglect of the important keys and good voices which more than anything else determine an owner's long-term satisfaction.

Nonetheless, gadgets and effects can be interesting and useful when they do not detract from the organ's main virtues. One of the most common effects to be found in today's organs is sustain or percussion, or both.

Sustain causes keys to bring forth normal continuous tone as long as keys are pressed; but when the keys are released the tone dies away slowly instead of ceasing almost immediately. Obviously, sustain allows you to approximate the sound of a percussive instrument such as a guitar or celesta by just touching a key and releasing it immediately. With moderate decay times, sustain also tends to help the novice player a bit by covering the pauses when he lifts his fingers to seek the next note and before he learns to play *legato* by pressing the next key just before he releases the first. It is especially useful when applied to pedals in organs with toe pedals, where the toe must be lifted from one before it can press the next. Sustain effects vary in detail; some have special voices for sustain only, while others have controls which can apply sustain to the regular organ stops or to groups of them. Usually some kind of control of the decay time is also provided. An unfortunate number of organs also have short-sustain effects which they term "reverberation." The effect does not sound like reverberation, though I suppose it may be better than nothing.

A drawback of sustain when it is, as most often happens, substituted for a real percussion, is that one cannot have percussive effects along with regular organ stops, since sustain does not sound percussive unless everything is played with a very *staccato* touch which will not allow the organ voices to sound fully.

With the real *percussion* (or *attack percussion*, as some makers call it to differentiate it from sustain percussion approximations), pressing a key causes a sudden onset of tone immediately followed by a decay, whether or not the key is held down. Percussions of this kind can be used along with organ stops, the percussive effect being heard at the time of key strike and the organ stop sounds remaining as the keys are held down after the percussion has died away.

Percussions and sustains may be more or less elaborate and lifelike, depending on design. They may be controlled by a rotary selector switch **(14)** or by regular stop tablets as in Fig. 2-7.

Combination Action

In an organ having a lot of stop tablets, and especially when the organ is used professionally, it is desirable to have a system for changing the registration which is quicker and less breathless than flicking many tablets by hand. Pipe organs have traditionally had *combination actions* and some electronics like that of Fig. 2-1 have them too. A number of pushbuttons **(16)** are provided, and pushing one button causes stop tablets to go up and down automatically and remain set ac-



Fig. 2-7–A few modern electronic theatre organs have very elaborate percussion divisions, with convincing realism and effects of various kinds in addition to different voices. These tablets control the percussion described in Chapter 10.

cording to whatever registration was predetermined for that button. The number of buttons provided is highly variable and so is their disposition. Usually one set is provided for each manual and another set, known as *generals,* is provided for the organ as a whole. The organist may thus elect to select a new preset combination for one manual only (or, more often, for one manual and the pedals), or for the whole organ. Sometimes, however, all the combinations are generals. Toe studs **(17)** may also be provided in case the hands are too busy to push buttons.

Types of action vary. All the good ones actually move the tablets so that not only can the organist see exactly what the registration is at any instant, but he can also alter it in part after the automatic selection simply by moving tablets. There are, however, some "blind" actions, such as the Rodgers described in Chapter 17, which simply do the stop switching under the hood and leave the tablets alone. The Hammond presets (Chapter 16) are also blind since they do not move the drawbars.

Among those which do move the tablets, methods of presetting vary. The best types are set up in the simplest way. For the organ of Fig. 2-1, for example, a combination is set by holding in a button and forcing the selected tablets up or down; pushing the button thereafter reproduces the setting. Other actions employ *setterboards,* which are banks of pull, toggle, or slide switches, one per stop per combination button, which are preset to taste. The organ described in Chapter 10 employs a setterboard which, instead of switches, uses small diodes which are inserted into terminals set into a printed circuit which ends up looking like a fakir's bed.

The traditional pipe organ used (and many still use) an action motivated by air taken from the main wind supply. Pushing a button admits air to a pneumatic mechanism which pushes the tabs up and down. This is why a combination button is traditionally called a *piston.* More modern (but not necessarily better) actions are electromechanical, utilizing combinations of solenoids, rods, swinging cams, and the like, with a generous dash of Rube Goldberg thrown in. A new action, very promising because of its basic simplicity and low cost, has been devised by Paul Klann of Klann, Inc., Waynesboro, Va., and is available to the industry. It uses a puff of air supplied by a solenoid-driven cylinder to inflate small plastic bags on each tablet mechanism to push the tab on or off. The organ of Chapter 10 is the first to use this new action.

Reverberation

An electronic organ in the home suffers from the "dead" acoustical setting which almost any living room gives. A very essential feeling of spaciousness may be obtained by a good artificial reverberation system. Hammond developed a spring-operated device for this purpose many years ago and now also sells smaller versions of it to other manufacturers (of both organs and other audio systems) through a subsidiary.

The Schober Reverbatape Unit is a more elaborate tape-recorder type of device whose reverberation effect can be controlled **(15).** Reverberation is discussed in detail in Chapter 7.

Pipe-Organ Devices

Two devices often found on pipe organs are sometimes, but seldom, seen on electronics.

The *crescendo* shoe looks just like a swell shoe from the outside. However, instead of controlling volume, it switches on stop after stop as it is advanced, in a selected but invariant sequence, just as if stop tabs were being pushed down by hand. (The tablets do not actually move.) The crescendo is useful in the pipe organ, at least in large part, as a means of increasing volume quickly and readily by adding ranks of pipes to the ensemble. Better ways of doing this are available electronically, hence the infrequent use. If a crescendo is used it must have an indicator of some sort in front of the organist's eyes, usually a mechanically coupled dial. Otherwise it can be on without the player's knowing it and he will get effects he did not bargain for.

Much the same philosophy applies to the *sforzando,* a pushbutton or toe stud which causes an effect sometimes called *full organ*—lots of stops and a big noise. To kill the effect the player pushes the button again. Sforzandos are rare on electronics.

Rhythm

A reasonably recent trend is toward the inclusion of so called rhythm devices, which make noises more or less accurately like those of rhythm instruments such as drums, wood blocks, castanets, and the like. These (real ones) were not uncommon on the old cinema organs. They were wired so that playing lower-manual or pedal notes also caused a strike on the rhythm instrument switched in. On electronics the sounds are created electronically.

Two of these devices are described in Chapters 14 and 15, but they are not the only ones in the industry.

Other Gadgets

The electronic organ is a fertile field for the gadget-minded engineer, and various organs contain circuits, devices, and controls covering a wide field of possibilities. Slides, glides, this-does-that, repeats; all are interesting and sometimes even useful. Where they are of sufficient interest they are described elsewhere in this book.

Chapter 3

Tone Generators

In Chapters 3 through 8 we shall look into the various functional sections into which an electronic organ naturally divides. We shall examine the need for each function and varied ways of filling the need; but we will not go very much into the exact details. The purpose of this section of the book is to give you the background you will need to understand what the individual manufacturers have done as described in the chapters on specific organs.

Every musical instrument has at least one tone generator—a reed or two, the player's lips, vibrating strings, and the like. In most orchestral instruments there is just the single generator, and the player is able to vary its pitch to produce one note at a time. In the organ, however, many notes must be playable at once; so a great many generators are provided, each "permanently" tuned to one pitch. Pressing keys may turn them on or switch their outputs to a common point; that we will see in the next chapter.

Generators in electronic organs fall into three general categories. The first is *nonoscillating* generators, a catch-all class which includes all those that do not have vacuum tube or transistor oscillators. The second is *individual-note* systems, which use a separate tube or transistor oscillator for each pitch, and sometimes several ranks of them. The third is *frequency dividers*, a system in which only 12 master oscillators are provided, one for each note of a single octave, and each oscillator is followed by a string of slave circuits which divide by 2 any frequency fed to them and thus reproduce the scale in lower octaves. *Monophonic* generators are a sort of subclass, producing just one note at a time, usually for the pedals of less expensive organs, thus saving the cost of a generator for each pedal.

How Many Generators?

Tone generators are what make an organ seem complicated. Most organs produce at least four octaves of pitches, and even on a small organ this amounts to at least 48 individual circuits. Forty-eight of anything (and more usually a much larger number, ranging up to 85 and sometimes several times that) is rather a lot. But since they are all more or less the same, the organ is really compound rather than complex. If you have understood one generator in most organs, you have understood them all.

The "standard" organ (those similar in minimum console facilities to pipe organs) have 61 keys per manual. At 8′ pitch this requires 61 different pitches for completeness. (You can see the pitches involved in each case discussed here by referring to Chapter 1.) If each manual also has 4′ pitches available, this adds

another necessary 12 pitches, for a total of 73 (or sometimes 72, when the top C is not provided at 4′). And if there is a 16′ register, still another 12 are needed at the bottom. This makes a total of 84 or 85. The lowest 16′ octave is also used for the pedals. Frequency-divider organs can usually employ the same set of generators for all keyboards, so the usual maximum remains at 84 or 85. But individual-oscillator organs often use a rank of generators for one keyboard only. So in the cases we have cited, such an organ would require at least 84 or 85 generators for each manual and another rank figured the same way for whatever number of pedals there are.

Different organs and models vary both up and down on this quantity assessment. A very elaborate organ may have several ranks of individual or frequency-divider generators for each manual, to create celeste and chorus effects or to supply different voices. A very inexpensive organ may cut corners by supplying fewer than the "minimum" number of generators and having adjacent octaves of keys sounding the same pitches under certain circumstances.

All this can easily amount to a good bushel basket of generators—and all of them, even in a cheap instrument, must be acceptably stable and not require frequent retuning. A good deal of care is lavished on generator design, therefore, to make them both good and as cheap as possible; and much of what you pay for an organ goes for the generators.

Rotating Tone Generators

Most of the nonelectronic tone generators comprise wheels of some sort, each one rotating at a constant speed and producing a repetitive pattern of electrical pulsations or oscillations. The action may be electromagnetic, electrostatic, or photoelectric, but all rotating generators have certain basic requirements.

The first requirement is that for each wheel, the number of whatever-makes-it-work must be integral and the spacing between all of them in degrees must be exactly equal. The point can be illustrated with the disc in Fig. 3-1, and without even bringing electronics into the discussion.

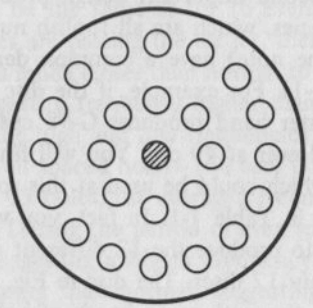


Fig. 3-1—Metal disc with 16-hole outer band, 8-hole inner, illustrates principles of rotary generators.

Let us assume that we have this metal disc shaped like a phonograph record, which spins on a shaft. We provide a pipe whose inner diameter is about the same as the diameter of each hole in the two circular bands on the disc, and through the pipe we direct a steady stream of air against one face of the disc.

The pipe is so placed that at any one of 16 positions of rotation the air blows right through one of the 16 holes in the outer band.

If the disc is rotated at a steady 10 rps (600 rpm), 160 puffs of air will pass through holes each second, and, standing at our side of the disc we will hear an audio tone of 160 cps. We can arrive at a very simple formula for the frequency of the tone produced: $f = sn$, where s is the speed of rotation in revolutions per second and n is the number of holes. Obviously, the same thinking applies whether we use the air system described, a photoelectric method which lets light shine through the holes onto a photocell, or any other rotary idea.

In reducing such a system to practice, however, the theoretical simplicity gives way to some problems. While it is true that the average frequency obtained depends on the number of stimuli per unit time, it is not sufficient merely to provide 160 stimuli over a period of a second, because the ear can distinguish pitch in a far, far shorter time than one second, or even a tenth of a second (one revolution in our example). We actually distinguish pitch in terms of repetition rate, which in practice means that our brains measure the difference in time between each impulse and the next. As long as all the time differences are exactly equal, as they would be if the holes were perfectly evenly spaced, and the rotational speed of the disc exactly constant, we will hear a steady tone. But any inequities in hole spacing or variations of disc speed will register as phase modulation and will be readily (and usually annoyingly) apparent. Small variations in hole spacing will register as unpleasant noise. A sizeable inequity in hole spacing at one point on the disc will cause a loud apparent click or dropout, depending on speed and frequency. Actually, the only thing we can allow is a small, smooth and steady speed variation 5 to 8 times per second, which would merely give a vibrato effect. It is actually very difficult indeed to produce a disc with precisely equal element spacings, which is one strong reason why so few organs using rotary generators have been successful.

A second important design problem is caused by the peskiness of musical frequencies, which are all foolish numbers, no two of which (except octaves of the same note) have a common denominator high enough to be practical (see Table 1-1). For example, if the disc is rotated at 6.125 rps (367.5 rpm), the 16-hole outer band produces G-98 cps and the 8-hole inner band yields the G an octave lower at 49 cps. You will find that there is no other integral number of holes which could be used at this speed to produce any other musical note appearing in Table 1-1. In fact, you will have to have discs going at 12 different speeds to produce the 12 different notes. You can, however, do the whole job with only 12 discs. The disc in Fig. 3-1 has two bands of holes. You might also make a disc with seven bands of holes, the outer band having 512, the next half that or 256, and so on, each band having half the number of holes as the previous one, ending with a band of 8 holes near the center. If the disc is then rotated at the 6.125-rps rate, the outer band will produce 3135.95 cps, near enough to the frequency of G in the seventh octave, and the inner one will produce G in Octave 1, 49 cps, the remaining bands producing all the G's in between. You can then use 11 more similar discs, rotating at various speeds over a range of a bit

less than 2:1, to produce seven octaves of the remaining notes. If convenient, the whole system could also rotate faster and have fewer holes in each band, or *vice versa.*

It is irritating in the extreme to have to go through all this mechanical work to obtain 12 different accurate speeds, when the temptation is strong to figure out a way to have all wheels driven at the same speed or even have all the notes of the organ on a single wheel. Mathematically, of course, you can do it. If disc speed, for instance, is 10 rps and you would like 155 cps, the simple answer is to have 15½ holes. Any wise guy will snicker at the idea of half a hole (which, unlike the loaves, is not necessarily better than none), but when you are dealing with photoelectric patterns, for instance, it is quite practical to install half a one. Unfortunately, none of this will work.

Suppose we try to make a disc with 15½ holes (or patterns, or whatever). We lay out radials at intervals of 360/15.5 or 23.225806 degrees and at each point where a radial intersects a circle we have drawn, we drill a hole. As we get to the end of the job, we will have to decide whether to drill 16 holes or 15, or we might even drill half a hole, which would look like a semicircle. But what actually determines the perceived frequency is the interval between holes, of which there must be 15.5 for each rotation, and since 15.5 is not an integer, at some point on the disc there will have to be at least one interval between one hole and the next which is not the same as the rest of the intervals—in other words, half an interval (or an interval and a half).

Now, when we spin the disc we will hear pulses from the equally spaced holes for most of the rotation, establishing the 155-cps frequency. But when we come to the uneven spacing there will be a sudden discontinuity of phase which will be heard as a loud noise. The result is totally unacceptable.

The strange fact, too, is that we will also get an impression of a steady frequency (with the noise superimposed). If we have actually made 16 holes (counting half a hole as a full one since for this purpose there either is a hole or there is not), the over-all frequency will be the same as for 16 evenly spaced holes. Why? While the evenly spaced holes are passing the air jet, their spacing suggests 155 cps; but since the 16th hole is much closer than normal to the next, the interval between those two alone suggests a frequency higher than 160 cps. Over a period of time the ear integrates the high and low frequencies and averages them to the same frequency as 16 well spaced holes—160 cps. Too, as you might expect, if you elect to make only 15 holes, the averaged frequency will sound like that of a normal 15-hole disc. During the period of even spacing the holes are closer together than normal, suggesting a higher frequency. But the distance between the last two holes is larger than normal, suggesting a lower frequency. Again they average and produce the sensation of exactly 150 cps.

This integrating effect is interesting but not at all useful, and it works only when the disc speed is reasonably high as in this example. It is accompanied by a loud noise each revolution. You can easily experiment along these lines, as I have. You will find that a band of holes or patterns to be detected any way you please will very definitely produce a tone frequency, even if the spacing is rather

offhand. But it will also produce far too much noise and garbage unless you space elements by some method, such as engine dividing, which is far more accurate than any layout man or draftsman can employ.

You will also have some trouble keeping the disc in one plane at all points of rotation, and in keeping speed exactly constant, remembering that you cannot use plain gears because the tooth-engagement frequency would modulate the audio. One way of licking the plane problem is to make the discs semiflexible and rotate them at high speed so that centrifugal force keeps them flat. Electro-Voice, which makes an electrostatic organ, uses belts to get around the gear problem. Hammond uses resilient couplings.

Photoelectric Generators

For a great many years, as evidenced by the patent literature and plain common sense, one of the most attractive basic ideas for tone generation has been the photoelectric, largely because it potentially eliminates the need for synthesizing anything. Theoretically, photoelectric recordings of actual organ pipes (or any other instrument) can be made, affixed to a disc, and played back. Unfortunately, life is never so kind and in practice photoelectric organs are so difficult to make that not one remains on the market. Nonetheless, some reader of this book with the necessary imagination and ingenuity may well find the key to unlock the rich possibilities of photoelectrics. To my knowledge, three photo organs have been marketed. The German Welte was produced before World War II and never revived afterward. Baldwin produced a model in the fifties and shortly withdrew it. And Kimball made one for a while, then stopped and went to electronic generation.

The idea is illustrated basically by the two frames of 35-mm motion picture film in Fig. 3-2. Notice at the left of the pictures the vertical sound track, which

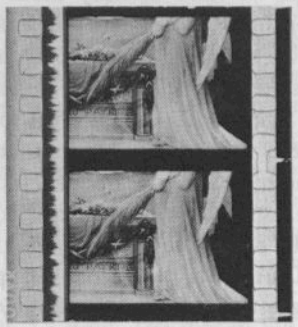


Fig. 3-2–Two frames of movie film with a variable-area sound track at left.

consists of waves based at the left with their peaks toward the right. When this track passes across a very narrow horizontal line of light, focused on it from a slit over a light source, a photocell picks up varying light amplitudes from the other side of the film and the resulting audio signals are translated by amplifier and speaker to sound. The sound track illustrated is of the *variable-area* type, less common on modern motion picture prints than the *variable-density*, in

which the full track is always modulated in varying intensities to correspond to instantaneous signal amplitudes.

The variable-area idea is carried directly across to the Welte tone disc of which a segment is reproduced in Fig. 3-3. The entire disc is clear except for the black waveform pictures in 18 concentric bands. When a radial slit of light is focused on one band and a photocell is placed on the opposite side of the disc, rotating the disc varies the light reaching the cell in accordance with the waveform. The fundamental pitch depends on disc rotation speed and the number of waveforms in the band.

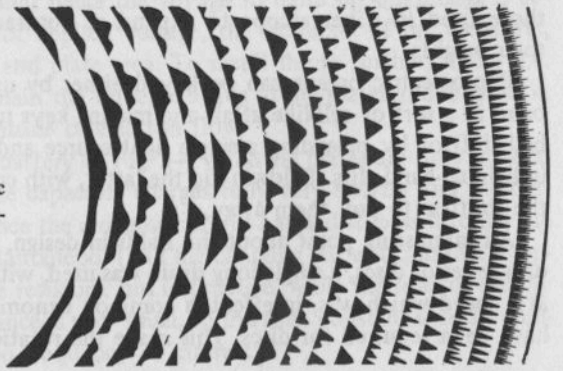


Fig. 3-3–Segment of a glass disc carrying photographic tone patterns.

While this system conforms with the dictum requiring a total of at least 12 discs, one for each note of the scale, each running at a different speed, each single disc can provide not only the various octave repetitions of its note but also different waveforms. Counting from the center (left in the photo), you can see that of the first three bands the innermost has the longest and thus fewest waveforms and this produces the lowest frequency. The second has twice as many and the third again twice as many, so that the first three bands produce three octaves, all of about the same tone color. Very likely this would be one of the pedal stops.

The fourth and fifth bands produce the same frequencies as the second and third, in a different voice, possibly a diapason because of the strong second harmonic content. Bands 6, 9, 12, 15, and 17 are four octaves of triangles, which I would guess to be some sort of flute, and the "diapason" tone is also continued into higher octaves. In all, I count seven octavely related pitches and four different waveforms. Your glasses may be better than mine.

Obviously a disc like this cannot be made just by optical recording, since the patterns must be very well defined and very sharply reproduced and–above all–reentrant, with an integral number of equally spaced, identical patterns in each band. Undoubtedly the desired waveshape for each voice was determined by optical recording, but the actual disc patterns must have been carefully drawn in ink and then made, after reduction, into a highly stable photographic master, probably on a glass plate. Other problems exist as well. For one thing, the patterns must be "bent" to conform with the circular band. For another, best re-

sults from a light slit which must be of finite width and thus attenuates upper frequencies and harmonics through gap effect as with a tape recorder, are obtained by fiddling the patterns a bit to preemphasize slopes, for example. The problems here are of the same kind as in the Electro-Voice organ, even though the latter scans its patterns electrostatically. The inventor had to devise many unusual techniques, including, for example, the use of anamorphic lenses like those used in Cinemascope to expand or contract patterns without affecting their heights.

Photoelectric organs can be keyed either by using a separate lamp for each band in a sort of cell-like affair and making keys turn the lamps on (which Kimball did) or by providing a single light source and a series of shutters covering individual-band slits. Baldwin did the latter, with cords running from the keys to the shutters to pull them away.

An interesting point about the Baldwin design, by the way, was that instead of a series of discs, a single long drum was used, with all patterns on it. It rotated at a speed which was, in effect, a common denominator of all those that would have been required for discs. This made the rotational speed very low, and the

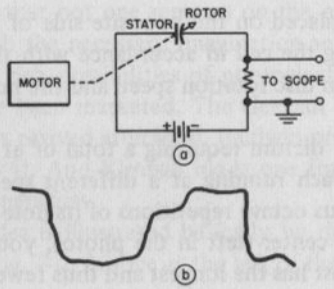


Fig. 3-4—Experimental setup illustrates electrostatic principle.

speed was controlled by a band of patterns at one end which was scanned and the frequency passed through a discriminator, forming a sort of servo system which continuously supplied corrections to keep speed constant.

A much simpler photoelectric organ could be made by furnishing only one or two different types of patterns, such as a sine and a sawtooth, from which organ voices could be synthesized by distribution and filtering. It is doubtful, however, especially today, that the mechanical hazards of a photo system would justify the idea when transistor generators are relatively cheap and much more reliable.

Electrostatic Generators

The electrostatic generator, which exists in rotating and in other forms, is based on the change of capacitance of a capacitor at an audio rate. The principle is easily illustrated by a setup like that diagrammed in Fig. 3-4. A variable tuning capacitor capable of continuous rotation has its shaft driven by a motor (I used an old phono motor) and the two sets of plates are connected to a battery and resistor as shown.

A capacitor is a pair of conducting surfaces in proximity, separated by a non-

conductor. When a voltage is applied to the plates, there is a sudden rush of electrons from the plate connected to the positive battery terminal, through the battery, to the other plate. Once all the electrons that can rush have rushed, the capacitor is charged; there is an excess of electrons on the "negative" plate and a scarcity on the positive.

The number of electrons that make the voyage to build up this charge is determined by the magnitude of applied voltage, the character of the dielectric, distance between the plates, and plate area. To distill things further, if the d.c. voltage and the dielectric remain the same, the area of the plates and separation distance determine the magnitude of electron flow.

Assuming we connect the battery of Fig. 3-4 while the plates are fully meshed and the motor turned off, the capacitor charges fully very shortly and the current flow comes to a stop. Since the charging current has had to pass through the resistor, charging was not instantaneous (and the charging current caused a temporary voltage drop across the resistor), but in practice, while the resistor usually has a high value, the capacitance is very small, so the time constant is small compared to a cycle of any audio frequency of interest.

Now we manually give the capacitor shaft a half-turn or so. The effective plate area is now much smaller and there is less capacitance. Some of the electrons on the negative plate must go back to the positive plate, and in so doing they create a temporary voltage drop across the resistor, in a direction opposite to that of the initial charge. We will find, in fact, that every time the capacitor is readjusted there will be a current flow and a resulting voltage drop across the resistor, its direction depending on whether we have increased or decreased capacitance.

Now we turn on the phonograph motor and the capacitor plates rotate continuously. With each revolution capacitance reaches maximum and minimum, and electrons pass back and forth through the resistor, creating an a.c. voltage across the resistor, the frequency corresponding to the rotational speed and the

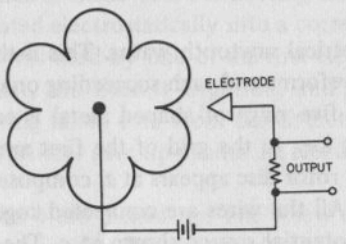


Fig. 3-5—Scalloped disc is a possible electrostatic generator.

waveform depending on the shapes of the capacitor plates. In this experiment, which I actually performed some years ago, the resistor was 15 megohms, the capacitor 10-30-pf ($\mu\mu$f they were back then), and the d.c. supply 90 volts. The motor shaft went around at about 900 rpm and I saw the resulting 15 cps on an oscilloscope with the waveform shown.

Fig. 3-5 shows a simple-minded way of putting the theory to work. A scalloped metal disc is substituted for one capacitor plate and a pointed electrode

for the other. By shaping the disc properly we could make any of a number of different waveforms. Actually, the capacitance here would be so small that we would run into bad noise problems unless we used enormous voltages. There are fringeing effects as well, since the electrode has a capacitive relationship not only to the disc edge point right in front of it, but also to those parts above and below it.

A more practical realization is illustrated by Fig. 3-6. This is taken from an old patent (2,522,923) of L. E. A. Bourn and represents the principle used in the British Compton organs.

The figure shows the two essential parts at *a* and *b*. A disc of insulating material *b* has five annular rings of metal mounted on it, each shaped as a series of

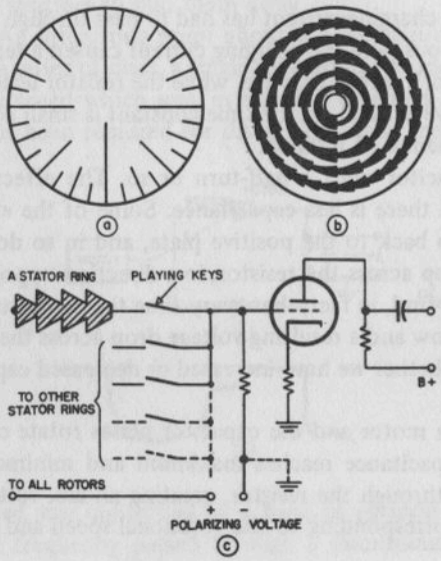


Fig. 3-6—A practical electrostatic generator patented by the designer of the British Compton organ.

symmetrical sawtooth waves. This is the stator disc. Note that the outer ring has 32 waveforms and each succeeding one has half that of the preceding band. Each of the five rings of shaped metal is connected through a switch operated by a playing key to the grid of the first amplifier tube as in *c*.

The rotor disc appears at *a*, composed of a network of rigid wires mounted on a rim. All the wires are connected together and to the positive end of the polarizing potential source shown at *c*. The rotor is placed parallel to the stator and very close to it.

Notice that in the rotor there are radial wires of five different lengths. As you can see by comparing the two discs and imagining them in their face-to-face position, for every band of patterns on the stator there is a group of radial wires separated by exactly one metal wavelength. As the rotor turns, all its wires form one "plate" of a capacitor while the other plate is divided into five separate parts—the rings on the stator. We thus have, in effect, five capacitors, all of

which have one common plate, which is, of course, the rotor of the arrangement.

Let us assume that the outermost ring of the stator is the one connected to the topmost key switch in diagram *c* and that the top playing key is pressed. Positive voltage from the supply is connected to the rotor, one plate of our capacitor, and negative voltage is now connected through the tube's grid resistor to the other plate, the patterned metal ring of the stator. As the rotor goes around, the capacitance at any instant depends on which part of each stator waveform is opposite a rotor wire. If it is the wide parts of the waveforms, capacitance is maximum; if it is the narrow parts, capacitance is minimum. Capacitance thus changes from minimum to maximum and back 32 times for every revolution of the rotor, and charging and discharging current passing through the grid resistor causes grid-ground voltage drops accordingly, as in our previous experiment. If the speed of the disc is such that 32 times rps is equal to a musical frequency, that is what is fed to the tube grid.

Obviously, a frequency one octave lower may be obtained by closing the key switch connected to the metal ring having 16 sawtooth patterns—and so on. And a series of such pairs of discs, with rotors running at the correct speeds, will produce five full octaves of all notes in the chromatic scale.

It is not actually necessary to have the network of wires shown for the rotor; a single radial scanning wire would theoretically do the same thing. However, providing a wire for each pattern of each band increases the actual value of capacitance so that the back and forth transfer of electrons is greater in amplitude, which helps overcome the noise level. It also helps to average out any variations the discs may have from perfect flatness. It is obvious that the waveforms could be other than sawtooth, although, as with all systems of this sort, the finite width of the scanning element tends to reduce high-frequency content.

Electrostatic Reeds

Electrostatic generators can be made in other ways than with rotating elements. In fact, the motion of anything which is made of a conducting substance and moves at an audio rate can be translated electrostatically into a corresponding voltage and amplified. This principle was used by one of the first electronic organs to appear on the American market, the Everett Orgatron, and later in Wurlitzer electronic organs, Wurlitzer having taken over from the Everett Piano Company (now owned by Hammond). Wurlitzer gave up producing electrostatic reed organs only recently.

Fig. 3-7 shows how strings *a* and reeds *b* are picked up. The screw and the string or reed form the two plates of the capacitor. Polarizing voltage (shown as a battery) is connected to the two "plates" through a resistor. As the string or reed vibrates (vertically in the drawings) the capacitance changes at the audio rate and an audio voltage is produced across the resistor. In the organs where reeds were used, they were windblown as in a parlor harmonium, but closed up in a soundproof box so that the actual acoustical sound of the reeds was not heard. Earlier organs of this kind actually had the reeds played by opening air valves, but speech was very slow and unsatisfactory so later models had the

reeds blown continuously, while the keys operated switches which applied polarizing voltage, through time-constant circuits for gradual attacks and decays, to the pickup screws.

The waveforms of the tones picked up in this way are not the same as the acoustic outputs of reeds and strings, since they only detect the motion of a small area of the moving element. Wurlitzer organ pickups produced mostly a waveform which was almost sine, though there was also a group of pickups which were thin strips of metal out at the edges of the reeds, which gave complex waves. The effect was not that of a harmonium and, especially in the later and smaller models, tone coloring was obtained by harmonic synthesis—building up tones by simultaneously keying reeds corresponding to (almost) harmonics, much as in the standard Hammond organ. Full details of an organ of this kind are in the second edition of this book.

There has never been any significant sale of instruments using electrostatic pickup of other musical elements. I do remember that Ansley Dynaphone electronic pianos made a brief appearance in about 1939 but nothing much came of it. They didn't sound very good, just loud.

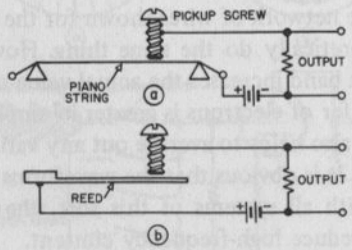


Fig. 3-7—String (a) and reed (b) motions are picked up electrostatically.

Only two organs are still made in the United States using generators which are not purely electronic—Hammond with its phonic wheels and Electro-Voice with electrostatic generators. I firmly believe that no more organs using electromechanical generators will be successful and that there is no need for them. With the advent of transistors, and with further developments to be expected along the lines of integrated circuits and perhaps techniques of which we know little or nothing today, purely electronic elements are too cheap, simple, reliable, and effective as tone synthesizers to encourage anyone to diddle with these fancy but musically limited ideas. True, Hammond, at least, long ago achieved a highly reliable and relatively inexpensive system which, like the internal combustion engine, was so aggressively engineered that it worked. But there is no longer any need to endure the tonal limitations of such a system, with its unnatural harmonic synthesis and its lack of all the important higher harmonics, a thesis with which Hammond apparently agrees in its sporadic attempts to come out with purely electronic instruments (Novachord, Solovox, chord organs, Extravoice, and now a new series of, for the first time, straight organs in electronic form which I think will succeed and may eventually supplant the weary old whirling discs).

Let us, therefore, pass on to the more fruitful study of purely electronic tone generators.

Electronic Generators

Electronic tone generator systems as used today divide themselves into two classes. The first is *individual-oscillator* generators, which contain at least one separate, individually tuned and controlled oscillator for each desired pitch. The second is *frequency-divider* systems, in which there is a self controlled oscillator for each of the highest desired 12 notes, each one followed by slave stages which divide the frequency of the oscillator and of each other by two to obtain the lower-octave notes.

Before getting to the nuts and bolts of the two major classes, we may as well settle down to a little comparative discussion, for perhaps the major controversy among organ designers (and educated users) is the one type *versus* the other; and selection of one or the other is almost the first decision a designer and his company must make.

First, neither type is categorically superior to the other—which is the reason for discussion—unless the organ designer knows that his organ must either (a) have the cheapest possible design without regard to quality or (b) have the most subtly superior musical characteristics without regard to cost. It is rare indeed that things are so easy, for no one (almost no one, anyway) deliberately sets out to make a bad product, and the best product in the world is useless if no one can afford to buy it.

The question hinges more on a matter of art than science. In a frequency-divider organ all voices and all tones are usually derived from a single rank of generators providing one, and only one, signal for each pitch. Furthermore, divider systems operate in such a way that all notes of a particular letter—all C's, all G's, etc.—have exactly precise octave relationships without the slightest variation, and are all in phase; they are literally locked together. For any note, therefore, no matter how many different voices may be produced by filtering or waveshaping, all the voices stem from a common source and are locked together in frequency and phase. Moreover, the same is true of all octaves of that note.

In an individual-oscillator organ, on the other hand, where a single rank of generators is used for the entire organ, the above is true for any single note, but octaves of the note are independent in frequency and phase and may be expected to wander a bit in both respects. When playing two different C's, then, for example, one can obtain much more of a sense of two different tone sources than with frequency dividers.

One can, of course, provide two complete sets of tone generators of either type. If this is done, then the frequency-divider organ still has only two independent sources for any note and all its octaves, but the individual-oscillator organ has as many independent sources as there are octaves of the note, times 2.

How important are independent sources? The classic example is the vocal chorus which may be composed of individual singers none of whom has a decent voice, only the ability to stay on pitch; yet the collective effect of the chorus

may be very listenable and musical. It appears that many independent tone sources, each usually very slightly different in pitch from the others, produce a feeling of slight pitch indeterminacy and also large numbers of very low-order beat frequencies, all of which lends an interest and warmth to the sound which renders it pleasing. It is for this reason that an orchestra includes many violins, for example—not simply to achieve greater sound level. It is why a large pipe organ sounds so vibrant and interesting in *sforzando* passages where many different ranks of pipes are speaking at once.

There is no doubt, therefore, that multiple sources of tone sound better than single sources. Because the more sources there are the greater the number of slight beat frequencies are created—and also beats between beats to a point of incredible complexity—the greater the number of sources the more interesting the tone, up to a point where adding more sources creates little or no more effect that can be perceived.

With a single source of tone for each pitch, adding more voices for a single note does nothing more than make a change in the single composite waveform. An instrument built on this basis depends for its musical interest almost entirely on the quality of its voicing, on the important fact that single notes or octaves are rarely played and other notes not locked to each other are sounding usually, and on extraneous matters such as the acoustic environment.

It follows, therefore, that an electronic organ with an adequate number of ranks of independent generators will, other things being equal, sound better than one using a single frequency-divider rank or even one with a similar number of divider ranks. In fact, such an organ which copies pipe design by supplying a rank of oscillators for each rank of pipes, each oscillator controlled as to speech transients in the right way, could sound so much like the pipe instrument as to be indistinguishable to most experts. Such organs have indeed been built (by Allen), and when the available funds are as great as are necessary for an organ of that class, the designer's choice is easy.

Individual-oscillator arrangements do have drawbacks, however, and none of the foregoing should be interpreted to mean that frequency-divider instruments do not sound highly musical when well designed. Individual-oscillator instruments require a very stable tuned circuit for each oscillator, while a frequency-divider stage needs no tuned circuit at all and is consequently a great deal cheaper. Even a run-of-the-mill individual-oscillator organ may have 200 such tuned circuits, with consequent much greater chance of getting out of tune and much more work and time in the tuning than a 12-oscillator divider instrument. Space for a frequency-divider system is much less than for the other, since no large coils are required.

Musical flexibility is usually sacrificed in individual-oscillator organs, except those costing several tens of thousands of dollars. It is usual practice to key the oscillators on and off and to use any rank for more than one pitch register, as in the Conn and Rodgers unit flute generators described in Chapters 19 and 17. Thus no extra tone is added when a particular oscillator is keyed as part of stops in two or more registers. If the oscillators are left running, as in the Conn, and

electronic keyers are used, the effect is the same since it would be too expensive to use many ranks of keyers. Flexibility, in terms of large numbers of stops with greater differences in tone colors, certainly favors frequency-divider design as it is usually practiced.

A thousand other points could be brought out which favor one system on one account and the other for some other reason. To confuse the matter further, the possibilities for canny design are great. For instance, suppose you provide three divider ranks and arrange matters so that succeeding octaves in any pitch register draw upon different generators. For any stop that will give about the same effect as independent oscillators, but will still be cheaper, smaller, and easier to service.

To sum up, individual-oscillator design, as generally practiced, gives the most subtly fine musical results but generally limits the tonal flexibility of the instrument. Frequency-divider design, again as most usually carried out, is indisputably cheaper and easier, but musically is less satisfactory, especially for those who seek as close as possible a replica of pipe effects. Each designer and each purchaser can only try to decide which results are the most important and weigh the factors to make a decision. I have personally always used frequency dividers in my designs. But if I were to embark on the design of an organ to be priced at more than, say, $7,000 or $8,000 (or the kit-form equivalent) I would probably go either to individual oscillators or to some multiple-rank divider system with interwoven octaves. The breaking point would be determined by the price at which I could provide the subtle beauty of multiple sources without sacrificing the tonal variety and pitch register control of the ordinary divider system. Many musicians and engineers will be both with me and against me.

Oscillators

One item common to generators of both types is the self-controlled oscillator, which is used for the top octave in divider systems and for all notes, with changes of L and C values, in others. There is little point in going into oscillator theory since it is covered in any basic book on electronics. Any stable L-C oscillator will work, and in the chapters on specific organs you will find that each maker has his own little variations of standard designs, sometimes to produce a particular output waveform, sometimes to allow frequency deviation for vibrato purposes, and sometimes just because the engineer involved thought it would be pleasant.

R-C oscillators have always been attractive because of the possibility of dispensing with the space and cost for the inductor. No present organ uses them, however, for they are not sufficiently stable, as Minshall once actually proved on the market, to its cost. I am certain that every manufacturer has calculated and experimented extensively on coilless oscillators; I know we have at Schober. In finally giving up we are in good company. (But if you finally hit on a really good R-C oscillator do let me know!) The problem with R-C oscillators is that in most cases the bandpass of the selector circuit is too wide and oscillation frequency has too many possibilities, and in all cases frequency is controlled in part by resistors, which are sensitive to heat.

The most usual type of tuning coil is simply wire wound around a form having an adjustable powdered iron or ferrite core. The coil of Fig. 3-8 is not untypical. Sometimes the form looks a bit different for mounting reasons; and the

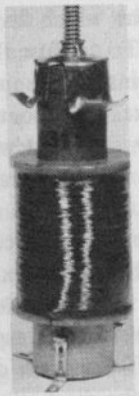


Fig. 3-8—A common type of oscillator coil, especially in frequency-divider organs where large inductances are not needed.

winding may be either single and tapped or double for transformer effect. A single coil value can easily be used over a range of an octave (with three or four capacitor value changes), and may thus be the only type furnished for a frequency-divider organ.

A second type of inductor, used in different values for individual-oscillator organs, appears in Fig. 3-9 and looks like an ordinary transformer, and with excellent reason because it is built like one. It employs E and I laminations of suitable steel, and it varies in only one very significant way from the usual transformer construction. The I lamination stack is effectively hinged where it joins one end of the E. A spring-loaded screw holds the movable laminations against the rest of them. When the nut on the screw is loosened the movable laminations



Fig. 3-9—A tuning inductor on a transformer core. The nut and spring allow adjustment of end stack. Conn uses these.

are pushed by the spring away from the E laminations, causing an air gap in the magnetic circuit. The nut may be adjusted to get just the air gap required for a given inductance within the available range, and this is how the coils are tuned.

Because the magnetic circuit is nearly complete, these coils also give high inductance for a given size and amount of wire, and they can be manufactured at low cost from standard transformer materials. The use of laminations rather than solid iron or steel core material reduces core losses as in any normal transformer, but not to the extent that powdered iron would. The continuous (almost) mag-

netic circuit also helps to confine the magnetic field to avoid interaction between coils, but not nearly so well as the cup core.

Tuning these coils is not quite so easy as adjusting the slug in a coil like that of Fig. 3-8 because the spring action plus friction does not allow the core-piece movement to conform exactly to turns of the nut. Customary procedure is to loosen the nut until frequency is a bit high, then tighten it until frequency is just right. The trick is easily learned. It is also important to proportion the capacitor so that the gap is not almost closed, for here inductance is more sensitive to slight gap changes because they are large percentages of the total gap. When the gap is open beyond a certain point precise tuning is easier and jars are less likely to cause mistuning.

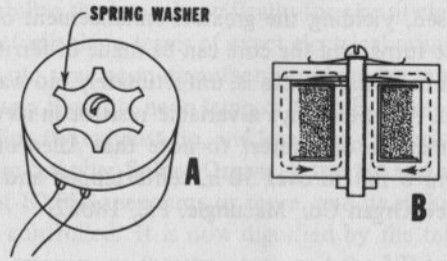
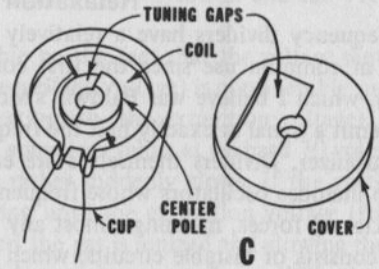


Fig. 3-10—Field of cup-core coil is wholly contained within core.



Both of these coils have one disadvantage—formation of and sensitivity to an external field—and this can make life difficult because they interact when placed too close together. The cup core coil of Fig. 3-10 avoids this problem because its field is contained wholly within itself.

The cup core, which is used at the moment by Rodgers and Gulbransen, is rather interesting. It is actually a cup made of ferrite, with a cylindrical core in the center. A small bobbin on which the coil is wound fits inside the cup, over the core. Then a flat piece of ferrite is used to close the cup, or a second similar cup is up-ended over the first. The ferrite forms not only a core inside the center of the coil but a closed magnetic path around it. One advantage is, of course, that the character of the ferrite and the closed magnetic circuit greatly increase the inductance of a given coil so that wire resistance remains low and Q high (and space is saved). Another is that many such cup-core coils can be placed in a relatively small space, close to each other. Because the cup is closed, the magnetic field generated by the oscillations in each coil is almost totally confined and cannot emerge to cause interaction with an adjacent coil.

Cup-core coils can be made tunable by slant-grinding or slotting surfaces so that rotation of the cover or one of the two mating halves causes an increasing gap between the halves and an air gap in the magnetic circuit. Adjustment is limited, of course, to a half turn. This means that either adjustment is very touchy physically or the range of adjustment in terms of frequency has to be very small. The ferrite material is somewhat temperature sensitive—at least the variety usually used.

The greatest tuning stability can be obtained in a coil when it has an air core (no iron core to cause any losses) and large wire is used to make for minimum resistance. Allen did this at one time. Allen now employs toroidal coils, which probably are the next best thing since the magnetic circuit of the core is closed, yielding the greatest enhancement of inductance for a given number of wire turns, and the core can be made of ferrite, powdered iron, or any other suitable material. There is, unfortunately, no way to vary the inductance of a toroid coil. Allen employs a variable resistor in its circuit for tuning over a small range. It may be of interest to note that Allen offers these toroids for sale in values from 6 mh to over 50 h, centertapped and untapped. For information address Allen Organ Co., Macungie, Pa., 18062.

Relaxation Oscillators

Frequency dividers have a relatively long history in the organ field, having been in common use since the first commercially successful purely electronic organ, which I believe was Baldwin's Model 5. The basic job of any divider stage is to emit a signal at exactly half the frequency of that supplied to it as trigger or synchronizer. Dividers themselves are easily divided into two classes. The first group includes oscillators whose frequencies are easily influenced and controlled by external forces, meaning almost any sort of *relaxation oscillator.* The second class consists of bistable circuits, which we will get to a little later.

Relaxation oscillators employ R-C or R-L circuits for frequency determination, but they are also subject to the effects of any other parameter changes, such as supply voltages and tube or transistor gain. They are inherently unstable, which is why they are so subject to external control. They invariably produce waveshapes other than sine, and that fact, too, makes them useful in organs; a sine wave is useful only for flute voicing or harmonic synthesis, while complex waves can be filtered to produce voicing of many kinds. The term "relaxation" is apt because each of these oscillators operates with brief periods of action followed by (usually) longer periods of inaction or relaxation. Phase-shift, twin-T, and Wien-bridge oscillators, by the way, do not act that way and are not relaxation oscillators even though they do not have L-C tank circuits.

Neon Dividers

The neon-lamp oscillator is a classic and very simple relaxation oscillator. Originally designed as an indicator, the neon lamp is a glass envelope filled with neon gas (normally a nonconductor), containing two electrodes. If d.c. is applied to the electrodes and the voltage gradually raised, the gas ionizes and conducts

when a certain voltage is reached, and it glows with an orange-red color, the glow surrounding the negative electrode. If the lamp is energized from an a.c. source (as when it is used as an indicator right across the power line), one electrode glows on each alternation and to the eye both seem to glow at the same time.

It was recognized many years ago that neon lamps could be used as circuit elements, and as far back as 1933 (at least) Nicholas Langer has patents on their use as oscillator components in organs. But not until Signalite, Inc., deliberately

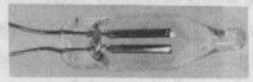


Fig. 3-11—A small neon lamp designed specifically as a circuit element, made by Signalite.

set out to manufacture, age, and stabilize the lamps specifically for circuit element use were reliable lamps available (reliable in terms of exact electrical ratings) generally. Now General Electric, an old time neon manufacturer, does the same, at least to an extent. Fig. 3-11 shows a Signalite neon lamp of the NE-2 class. It is slightly over 3/4 inch long, including the exhaust tip, and less than 1/4 inch in diameter. This particular one, used in Schober Spinet Organs, is treated and selected for interelectrode resistance of 80,000 megohms or more, and its striking and extinction voltages are closely controlled. It is now dignified by the term "gas tube" and belongs in the same category as the thyratron and the VR tube gaseous regulator.

Its use as an oscillator is made possible by the fact that the voltage required to ionize the gas and ignite the tube (the *striking* voltage) is considerably greater than the voltage at which it will then extinguish (the *extinction* voltage). That is, if the voltage is gradually raised from some low value, at perhaps 70 volts (depending on the particular lamp) the gas rather suddenly glows. If the voltage is then lowered, the lamp keeps on glowing until the extinction voltage (for example perhaps 55 volts) is reached. When the gas is ionized and glowing the resistance between the electrodes is fairly low and current flows. With the tube extinct the resistance is extremely high and for practical purposes no current flows.

With this in mind, examine the simple, classic neon oscillator of Fig. 3-12. With the switch open and no B+ applied capacitor C is uncharged and no voltage appears across neon tube N, which is therefore not ionized and does not glow.

When we close the switch, B+ (of some value greater than the striking voltage of the lamp) causes C to begin charging through R. The charging process is actu-



Fig. 3-12—The basic, classic neon-lamp relaxation oscillator.

ally a transfer of electrons from the upper plate to the lower plate through R and the power supply. At the outset the number of electrons to be transferred is very large because the capacitor was initially at zero charge, so C tries to get rid of them all at once. However, these electrons, passing through R, create a voltage drop across R, so that most of the B-voltage appears across R and not across the

capacitor. Since little of the B-voltage is thus applied to the capacitor, the charging action is very much slowed up. Remember that all this time N is effectively an open circuit and does nothing.

As time goes on—the amount of time depending on the values of R and C—C gets more and more charged. That is, it gets closer and closer to the point where all the electrons that can be transferred are transferred. Much as when we eat a full meal and eat more and more slowly as our stomachs get fuller and our appetite decreases, the transfer of electrons gets slower and slower. The rate of charge of the capacitor therefore decreases, and the charging curve of the capacitor (typical whenever a capacitor is charged through a resistor) is shown by the A section of the graph.

After a certain time has passed, the capacitor is finally charged to a voltage sufficient to cause the gas in the tube to ionize—voltage S in the graph. When this happens tube N suddenly conducts. Since it is directly across the capacitor, it short-circuits C and causes C to discharge by restoring excess electrons on the lower plate through N (*not* through R and the supply) to the upper plate. The discharge is a very great deal faster than the charge, because the effective resistance between the electrodes of the ignited lamp is very much smaller than R. The lamp resistance is not zero, however, and the discharge is not a vertical line on the graph but one with some slant, indicating that it takes at least some time. The igniting of the lamp, you must note, is not like closing a switch right across the capacitor; it is liking closing a switch with some resistance in series with it.

As the capacitor discharges and the voltage across it diminishes, that voltage reaches the extinction value at which the lamp winks out. Notice on the graph that this is not zero voltage—zero capacitor charge—where the capacitor was before the switch was closed. It is, to go to the example we cited several paragraphs ago, the 55 volts at which the lamp extinguishes; and this point is indicated by dashed line X.

Since the lamp is now extinguished, it no longer affects the circuit and the capacitor begins to charge again through R from the B-supply. And again when it reaches the lamp striking voltage, S on the graph, the lamp ignites again and C discharges until lamp extinction point is reached and it goes out. This action continues indefinitely. The graph shows the voltage appearing across the capacitor. Obviously this is a pulsating d.c. and when passed through a d.c. blocking capacitor on its way to the next stage it becomes a.c. oscillations. The waveform, for obvious reasons, is a *sawtooth.*

The difficulty of controlling the exact frequency is easily demonstrated. Obviously both R and C are instrumental since they determine the time required for the capacitor to charge from the extinction voltage X to the striking voltage S. Temperature and voltage stable capacitors are not hard to come by, but any resistor, at least the ordinary composition kind, changes value significantly with heat. The exact supply voltage is critical, too. A higher voltage would cause the capacitor to reach point S faster, and *vice versa.* And the neon lamp, like all gas tubes, is "history sensitive." That is, the voltages at which it fires and extinguishes vary according to how recently and how long the gas was ionized—as

well as with temperature and even depending on the presence of light.

Now let us see how this oscillator can be caused to lock into step with a synchronizing signal. Assume that a small capacitor is now connected to the junction of R, N, and C. The other end of the capacitor is connected to a source of large positive pulses; and each pulse comes shortly before the charge on C reaches the point S where the lamp will fire. When the positive pulse comes along it will add to whatever charge is on the capacitor at the time, making the total voltage at the top of the lamp sufficient to fire the tube and initiate a new cycle. Since the new cycle thus starts whenever a synchronizing pulse comes through, the frequency of oscillation of the neon circuit now becomes the same as the frequency of the sync signal.

To synchronize a neon oscillator, therefore, we design it so that its own free-running frequency is lower than that at which we wish to synchronize it. We then inject sync pulses to fire the lamp before it would "normally" fire; that is, the sync signal raises oscillator frequency to that of the sync source.

Synchronizing the neon oscillator to the same frequency as the sync signal is very easy. It really does not matter how positive the sync pulse is, above a certain minimum value. But suppose we want to cause the neon oscillator to oscillate at exactly *half* the frequency of the sync signal—fire, that is, once every two cycles of sync.

First we must make sure that the free-running frequency of the neon circuit, as determined by the lamp type, the R and C, and the supply voltage, is lower than the ultimate frequency we wish, but only somewhat lower, not very much. Then we must control the amplitude of the sync pulse so that the first time one comes along, sync amplitude plus the amplitude of the charge on C is *not* enough to fire the tube, while the second time it comes the capacitor charge has increased enough so the sum of the two *is* enough to fire the tube. In this way, for every odd sync pulse the sum of sync and capacitor charge does not fire N, and at every even pulse the two add enough to fire it. When this is true, the sawtooth output of the neon circuit will have exactly half the frequency of the sync.

As a practical matter it is very difficult to design a synchronizing circuit for the arrangement of Fig. 3-12. The source of sync signal must be either the L-C master oscillator or a similar neon circuit. If either is connected to this neon circuit for sync purposes, some of the frequency of the synchronized stage feeds back to the synchronizing stage, and some signal of the lower octave appears in its output, which is unpleasant. For years ingenious attempts were made to devise synchronizing circuits that did not involve some sort of vacuum-tube isolation stage between each pair of neon oscillators, thus destroying the economy of the neon idea. There are patents galore, but none of them were practical in commercial terms. The breakthrough was an invention of J. D. Bick, U. S. Patent 2,680,198, assigned to RCA. Bick's principle was employed by a development group (of which I was a member) in the design of an instrument which later became the Kinsman Organ (now defunct but not for technical reasons). The neon circuit to be described is still used in the Schober Spinet Organ. In both the Kinsman and Schober applications these generators have been entirely

reliable and, to my knowledge, they are still about the cheapest dividers extant.

Fig. 3-13 shows two successive stages of such a divider string. Ignoring synchronization for the moment, let us see how the first stage is really an ordinary neon oscillator just like that of Fig. 3-12, despite the apparent doubling up of components. R1 and R2 in series simply result in a single total resistance equivalent to R in Fig. 3-12. A part of this resistance, R1, is made variable to set the free-running frequency within the range which can easily be synchronized.

The two neon lamps N1 and N2 in series represent no difference from lamp N in the previous diagram, except that double the voltage is required to fire them. And capacitors C1 and C2 in series across the lamps result in a single net value which performs the same function as C in Fig. 3-12. The reason for having two is to isolate the output from the oscillating circuit to some extent, since keying the organ introduces some resistance between the output point and ground. The lower capacitor, across which output is taken, has only 1/6 the capacitive reactance of the two in series, so the effect of the keying resistance on oscillator timing is minimized. The output signal is, of course, reduced as well, but since the

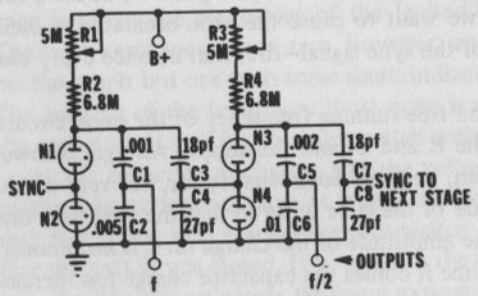


Fig. 3-13—A pair of neon-lamp frequency dividers.

voltage division is purely capacitive there is no change in the waveform. We thus have exactly the same kind of oscillator as in Fig. 3-12, and it works the same way. For the moment take my word for it that this oscillator has been synchronized by the oscillator for the octave above. Now let us cause our first oscillator to synchronize the second identical one, which consists of similar components except for C5 and C6, which have twice the value of C1 and C2 because the output of this second oscillator is to be an octave below the first.

At any time when lamps N3 and N4 are not glowing—when the voltage between the top of N3 and ground has not reached striking value S in the graph of Fig. 3-12—N3 and N4 are effectively open circuited. C3 and C4 are very small capacitors. But since they are connected in series across lamps N1 and N2, the full output of the first oscillator appears across them, and since they have no load and constitute a capacitive voltage divider, something less than half the output of the first oscillator appears at their junction. This junction is connected to the junction of N3 and N4.

Suppose that at a particular instant C5-C6 has charged nearly, but not quite, to the point where N3-N4 will fire. And suppose that at that instant N1-N2 *does* fire. Due to that firing, a rather steep negative pulse appears across C1-C2, and

the same pulse appears across C3-C4. A little less than half the negative pulse is applied to the lower end of N3. The top of N3 is already almost positive enough to fire. The addition of the negative pulse to the lower end of N3 is enough to make it fire and become a fairly low resistance. This immediately applies positive voltage to the top of N4 and it, too, fires. Thus, the firing of the first oscillator has triggered the firing of the second.

Now events continue in both oscillators as expected, and, having fired, both discharge their capacitors C1-C2 and C5-C6, and both sets of lamps go out. Both sets of capacitors begin to charge again, but because C5-C6 is twice as large as C1-C2 it charges only about half as fast. When N1-N2 fires for the second time, C5-C6 has charged to a point considerably below the firing point of N3-N4. While the firing of the first oscillator again applies a negative pulse to the bottom of N3 through the C3-C4 voltage divider, the negative pulse and the positive charge at the two ends of N3 do not add up to enough to fire N3; so N3 sits this one out.

However, by the third time N1-N2 fires, the charge on C5-C6 is nearly at the firing point, and the sync pulse is enough to cause N3-N4 to fire again.

It is obvious, therefore, that N3-N4 fires only once for every two cycles of N1-N2, and the output of the second oscillator is at exactly half the frequency of the first.

Now we come to the reason for the dual neon lamps. If there were only one lamp as in Fig. 3-12, the capacitor carrying sync to it would have to be large, since it would be the series leg of a voltage divider having as a shunt leg capacitor C. By large I mean it would have to be comparable in value to C. The output of the second oscillator would then couple back to the first, and output f would contain a component of $f/2$, something which is very apparent to the ear and sounds as though a 16′ stop is always present along with the 8′.

Using two lamps and applying the sync between them, at a point which is an open circuit except for the brief time when the lamps are ionized, a very small capacitor can be used, on the principle that there is no voltage drop across an impedance, no matter how high the impedance is, unless current flows. C4 is really not even necessary. However, we use C4 and make a voltage divider simply because the divider, and not the unpredictable capacitance between the electrodes of N4 (or that of wires or printed-circuit lines) then rigidly controls the applied sync amplitude.

The virtue of using small capacitance to connect the two oscillators together is that when the pulses of oscillator 2 which appear at the junction of N3-N4 try to get back into the output of oscillator 1, they must pass through a voltage divider consisting of C3 and C1 in series and C2 in shunt. The net value of C3-C1 as the series leg is so very small compared to that of C2 as the shunt leg that the fed-back signal from oscillator 2 is attenuated by a factor of around 150 times, or around 45 db, and becomes inaudible. And at the same time, neither the capacitor through which sync is applied, C3, nor operation of oscillator 1, has any effect on the operation of oscillator 2 except at the instant of sync.

Neon dividers of this type, while highly practical, are not readily compatible

with transistor circuitry because no neon lamp operates at less than about 70 volts for striking, much higher than usual transistor supply voltages. (B+ for Fig. 3-13 is actually 255 volts.) As can be seen, neon circuits require very high impedances, which make them at least potentially subject to the effects of extreme humidity which might saturate a printed circuit or deposit on the lamp envelope; and the divider outputs are of much higher impedance than other divider types, giving some possibility of problems in very elaborate organs. Despite the fact that the neon lamps have proven every bit as reliable and trouble-free as tube and transistor frequency dividers, they will probably die out eventually except perhaps in quite small organs where their low cost, even if a special power supply has to be furnished for them, outweighs the transistor trend. Neon-generated sawtooth waves are also slightly less useful for tone coloring than those generated in other ways, since flyback time is somewhat longer than with other circuits and less high-harmonic content is produced.

Blocking Oscillators

Two principal types of tube or transistor relaxation oscillators have been used for frequency division, *blocking oscillators* and *multivibrators*. They operate on the same basic proposition as the neon dividers. That is, the termination of a cycle or initiation of a new one depends on a sort of unmistakable "terminal condition" when something very definite happens (like the ionization of the gas in the neon lamp), and an injected synchronizing signal can bring about this happening sooner than the timing circuit of the oscillator normally would.

Fig. 3-14 shows a simple and workable blocking oscillator used by Schober in models now obsolete. Baldwin had a circuit which differed slightly in a little more complication in the R-C timing circuit and a fixed positive voltage applied to the cathodes, as well as in sync method.

The transformer used is a pulse transformer. While this may conjure images of expensive and sophisticated transformer design, the opposite is true. The transformers used by Schober were standard open-frame units measuring about 1 3/8 inches high, 1 3/4 inches wide, and 7/8 inch thick, with relatively little steel in the core and windings of 51 and 101 turns. There being little steel and low-impedance windings, the inductance is low and efficient transmission is limited to higher frequencies—or pulses, which consist of high-frequency harmonics of lower-repetition-rate signals. Manufacturer's cost of such a transformer is roughly the same as half a twin triode such as a 12AX7. The inductance of the transformers, lest you form a wrong impression, is of no importance as long as it is not too high, since it is not used for timing or resonance but solely for phase reversal. Actually, very nearly any transformer at all can be used to make up a blocking oscillator of this kind!

Suppose that at the moment power is applied from the B-supply a small random positive signal appears at the grid. It is amplified and reproduced at the plate as a negative signal. The transformer is so phased that anything at the plate is transferred to the grid in opposite phase; so the negative plate signal gets back to the grid as a positive signal, adding to the original small positive signal. This

positive feedback action snowballs—builds up so fast that the result is a very steep, high-amplitude positive grid pulse. The grid, of course, draws current since it is positive, and the grid current passes through R, creating a large voltage drop across R, with the negative end of the voltage drop at the top end of R. C1 and C2 in series have a net capacitance slightly less than .01 μf (this varies proportionately with the octave) and this net capacitor is quickly charged by the high-current pulse to the same voltage as the drop across R.

All this happens very quickly indeed—in microseconds—so fast that the voltage rise across the capacitors and resistor cannot be seen on an ordinary 5-mc oscilloscope. When the capacitors have charged, the potential on the grid is strongly negative with respect to ground, so the tube is well below cutoff and no further plate or grid current can be drawn. We then have a charged capacitor shunted by a resistor, one end of the combination grounded and the upper end effectively connected to nothing. The capacitor combination gradually discharges through R. The rate of discharge is determined entirely by the values of R and C operating as a classic time-constant circuit.

As the discharge continues, a value of voltage is finally reached between the top of the R-C combination (the grid) and ground which is less negative than the

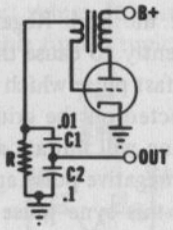


Fig. 3-14—The blocking oscillator yields an excellent sawtooth.

grid cutoff voltage. At this point electrons begin to flow again from cathode to plate and the tube again becomes an amplifier. Since the capacitor charge between grid and ground is steadily becoming less negative due to discharge through R, the grid sees a positive-going voltage. So as soon as the grid sticks its nose above cutoff, the entire action begins again. Left to its own devices, the blocking oscillator will oscillate in this manner at a rate determined by the values of R and C, the applied B+, the turns ratio of the transformer, and the amplification factor of the tube.

The waveform of the oscillations as obtained at the output terminal shown is easily deduced. In the first place, as in the outputs of the neon dividers, the capacitance is divided into two parts to isolate the key-switching resistors from the frequency determining components of the blocking oscillator. Note that the lower capacitor has 1/10 the reactance of the upper; so for all practical purposes the upper, which is not shunted by the output and what follows, mostly determines oscillator operation. The two capacitors form a voltage divider which, being capacitive in both legs, does not have frequency discrimination and therefore does not alter output waveform. The waveform appearing across C2 is the same as that across the combination of C1-C2.

When the capacitance charges at the start of the action (when the grid draws current) it charges negative and does so very quickly. This forms a very steep negative-going pulse in the output. As the capacitance discharges it does so exponentially. When discharge is sufficient to allow the tube to conduct again another steep negative pulse forms—and so on. This pattern of gradual positive-going and very fast negative-going voltage forms a series of sawtooth waves, just as in the graph of Fig. 3-12, but with almost truly vertical flybacks.

How one of these blocking oscillators is coupled to another to synchronize the two is not very important; practically any kind of coupling can be used, including even accidental coupling such as through the power supply or by close transformer proximity. Unlike the situation of the neon oscillator, this one has the benefit of an amplifying element, so only a small amount of sync need be injected and the problem of back-coupling need not exist. In any event, the starting point for a frequency-divider string intended to divide the frequency in half from each stage to the next is to adjust the values of C1 and C2 so that for each succeeding stage of lower frequency the R-C product approximately doubles. This produces a free-running frequency about half that of the preceding stage.

The principal action which initiates a cycle is the first appearance of a positive voltage at the grid. Regardless of whether or not the capacitance has discharged sufficiently to cause the tube to begin conducting, which would cause it to initiate the fast pulse which starts a new cycle on its own hook, enough positive voltage injected on the grid from outside by any method to make the tube begin conducting will initiate a new cycle. Effectively, therefore, the necessary sync signal is a negative pulse applied to the grid.

In practice, this sync pulse has an amplitude designed to be within certain limitations. The amplitude is such that if it comes within the first half of the normal discharge period of the capacitance it will not be enough to cause the tube to conduct and nothing will happen. But if it comes during the second half of the discharge time the capacitor charge is close enough to grid conduction potential so that when the sync pulse is added the tube conducts. In this way only every other sync pulse is effective, and synchronization occurs at half the frequency of the sync source.

In Baldwin's design all the transformers for all the divider stages of a particular generator were wound on the same core. This caused sufficient coupling between stages to provide reliable sync. In the Schober design the special transformer construction was avoided and a small trimmer-type capacitor was connected between the plates of successive stages. The trimmer was adjusted by the kit constructor to give the proper sync amplitude for correct 2:1 frequency division, simply by ear.

Blocking oscillators can easily, of course, be made with transistors, and the Gulbransen organ described in Chapter 13 has them. The circuitry is the same, except for power supply polarity if PNP's are used, but on bias must be furnished the base. A simple method which does not interfere with the values of the timing components is to connect the bottom of the R-C combination (or just

the R) to a low-impedance and well bypassed bias source rather than to ground.

The Multivibrator

The multivibrator is actually a generic class of 2-stage resistance-coupled amplifiers having the output of each stage connected to the input of the other, so designed that when one stage is cut off the other conducts fully, these conditions being swapped between the two stages either spontaneously or in response to triggering pulses.

There are three kinds of multivibrators. The *monostable* multivibrator or *one-shot* (once in a while called a *univibrator*) has only one stable condition—one specific stage conducting and the other cut off; and it will return to this condition sometime after triggering pulses are removed. Since it produces an output pulse of standard amplitude and length (determined by circuit values) in response to trigger pulses which may vary over a range, it is sometimes used as a "pulse standardizer," though rarely in organs.

The *bistable* multivibrator, better known as the *Eccles-Jordan flipflop*, changes state once for each triggering pulse applied and will stop with either of the stages conducting or cut off. More of this later.

The *astable* multivibrator is actually an oscillator in which the stages swap states spontaneously at a rate dependent on circuit values. It is, in fact, another relaxation oscillator. In simple but entirely workable form a typical multivibrator is shown in Fig. 3-15. (In common terminology the word "multivibrator" usually indicates this oscillating, astable type. The terms "one-shot" and "flip-flop" are used for the other two.)

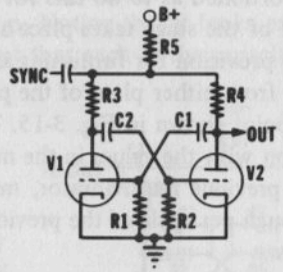


Fig. 3-15—A simple but workable astable multivibrator.

Since we have to assume something at the outset, let us assume a small positive potential at the grid of V2. This causes negative voltage at V2 plate, which is transmitted to V1 grid. The resulting positive at V1 plate goes back to V2 grid, augmenting the positive already there. This positive feedback action snowballs and very quickly V1 grid goes very negative and the negative charge at the grid end of C1 is enough to considerably more than cut off V1. At the same time V2 conducts fully. At this time voltage at the plate of V1 is as positive as the top of R3 because there is no current through R3 and therefore no drop. Voltage at V2 plate is much more negative because of the V2 plate current passing through it.

Since no more signal transfer is taking place C1 begins to discharge through R1. After a time it has discharged enough so that V1 grid is no longer negative

enough to cut off the tube. At this time it is positive-going; and immediately the action repeats except in the opposite sense, driving V2 to cutoff and V1 to full conduction. V1 plate is now negative and V2 plate positive.

A signal taken from either plate is a square wave with very steep sides (usually invisible on the normal oscilloscope) and perfectly flat tops and bottoms. The square wave is the result of the extremely fast changes of state between the two tubes, followed by periods of complete inaction (except for the gradual discharge of a grid capacitor, which cannot be observed in the plate circuit because the tube is cut off).

The free-running frequency of operation obviously depends on component values (the time constant of R1-C1 and R2-C2, which are identical), amplification of the tubes, and supply voltage. An assymetric rectangular wave can be produced by making the time constants different. As so far described R5 does nothing, since the transient currents through it are equal and opposite from the two tubes and cancel.

The presence of R5 does, however, provide a point above a.c. ground at which sync pulses can be injected from the previous stage. The pulses are positive. They have no effect on the stage which is conducting, but if the timing components have brought the nonconducting stage near enough to conduction, raising the plate-supply voltage in this manner can cause the tube to conduct, thus initiating a new changeover. As with the previous systems, sync amplitude is so proportioned as to do this for only every second sync pulse, so that synchronization of the stage takes place at half the frequency of the sync. In practice, no special provision for furnishing and sizing sync pulses is necessary. The output taken from either plate of the previous multivibrator is simply connected to the sync point shown in Fig. 3-15. The sync capacitor is a small one so that in conjunction with the values in the multivibrator it differentiates the square wave from the previous multivibrator, making it a pulse. Actually, two such pulses come through per cycle of the previous stage, but only the positive-going one has any effect.

Like the previous relaxation oscillators, this one must be within a reasonable range of the right frequency. This means components and voltages must be chosen right and tolerances must be carefully considered.

The Flipflop

While relaxation oscillators have been widely used, especially the neon and blocking oscillator types, which have the advantage of producing a sawtooth wave directly, they have the great disadvantage of being periodic; that is, each one will work only within a specific range of frequencies. They are thus not true frequency dividers, but only synchronized oscillators, and they can go wrong if component values or supply voltages change somewhat.

The flipflop, on the other hand, does exactly what it is told, neither more nor less. It will not oscillate by itself but acts only in response to specific commands; and a flipflop can be designed with a single set of component values with which it may be used at any frequency over the audio spectrum and more. Particularly

since the advent of cheap transistors, the flipflop has effectively made the relaxation oscillator obsolete, even the neon type with its inherent cheapness. Except for organs designed before the latest in transistor technology, flipflops have preempted the field.

The flipflop is basically a pair of amplifier elements (tubes or transistors) which are used as switches. The output of each is connected to the input of the other, but spontaneous oscillation cannot take place. One of the two transistors (let us forget about tubes from now on) is always fully conducting and the other is always cut off. When a sync pulse—more correctly called a *trigger* pulse—is supplied to the circuit, it causes the cut-off transistor immediately to conduct and the conducting one immediately to cut off. If output is taken from either one of the two transistors, two trigger pulses must be applied to cause two changes of transistor state or a complete cycle. Since a trigger pulse is supplied only once per cycle of the previous stage, only one cycle of the second stage is obtained for two cycles of the first, and we have frequency division by 2.

The output is always a square wave since it is composed only of two voltages, one each representing on and off conditions of the transistor. The trigger pulse required is quite sharp and narrow and is obtained simply by passing square output of one stage through an R-C differentiator to the next. As the vertical portions of the square waves are identical regardless of repetition rate, the differentiated output is the same in all cases and no value changes are necessary to correspond with frequency, making the device truly aperiodic.

Fig. 3-16 shows a simple, typical flipflop. Notice that it looks exactly like the astable multivibrator of Fig. 3-15, except that each of the capacitors coupl-

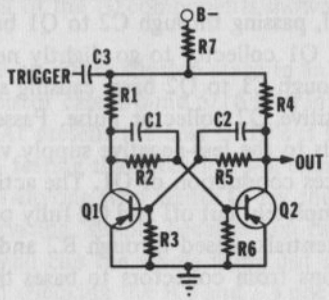


Fig. 3-16—The flipflop. Some in use are actually like this; others have a capacitor between bases or some bias on emitters. Occasionally trigger pulses are delivered directly to the bases through capacitors.

ing a collector to a base is paralleled by a resistor (and of course it uses transistors instead of tubes). Note that the transistors are PNP's (in our illustration, not in all flipflops), meaning that negative base voltage turns them on and positive turns them off.

When the power is first turned on, negative potential is applied not only to both collectors, but also to both bases through R2 and R5, tending to bias both transistors on. Actually, however, the turning on of power causes an initial *pulse* of on bias to be applied to the bases through C1 and C2. Due to circuit inequities, a slightly larger pulse will go to one of the bases; let us assume the favored transistor is Q1.

As Q1 goes more quickly than Q2 toward full conduction, its collector potential also goes more quickly in a positive direction. This positive-going pulse passes through C1 to the Q2 base, overcoming the lesser initial negative voltage there. Thus as Q1 goes toward full conduction, it drives Q2 toward cutoff.

The voltage at any time on the collector of one of these transistors depends on voltage-divider action. The emitter-collector path in the transistor is the shunt leg of the divider, and the collector resistor (R1 or R4) the series leg. With Q1 fully conducting, it is a very low resistance, so that its collector voltage is almost the same as ground. Since the bias for the base of Q2 must come from the Q1 collector through R2 after the initial pulse action is over, Q2 base is practically at ground potential and Q2 cannot conduct. This makes Q2 a large resistance, so that its collector voltage is almost the same as the B– appearing at the bottom of R7, which is a relatively small resistor. The highly negative Q2 collector voltage is applied to the base of Q1 as on bias and stays there. The circuit has now reached one of its two stable states–Q1 conducting and Q2 cut off–and in the absence of any external stimulus it will stay there.

Let us suppose now that we insert a negative pulse at the trigger point, through C3. It will make the bottom of R7 even more negative than before. Since it will pass down through R4 and C2-R5 to the Q1 base, it will make Q1 conduct even more heavily, if possible (for the duration of the pulse), bringing Q1 collector even closer to ground potential and cutting off Q2 even more firmly if possible. In other words, a negative trigger pulse only confirms the stable state at which the circuit has arrived.

Next let us pass a positive pulse through C3. Its effect is to make the supply voltage for both transistors less negative. If the pulse has sufficient amplitude, it will, passing through C2 to Q1 base, slightly reduce the conduction of Q1, causing Q1 collector to go slightly negative. The negative Q1 collector pulse passes through C1 to Q2 base, causing slight conduction of Q2 and a resulting slightly positive Q2 collector pulse. Passed to Q1 base through C2, this positive pulse adds to the less-negative supply voltage caused by the trigger and still further reduces conduction of Q1. The action snowballs and in microseconds Q1 becomes completely cut off and Q2 fully on, both maintained in that condition by steady potentials passed through R2 and R5. The functions of the parallel R-C combinations from collectors to bases thus becomes clear. The *change* of state occurs very quickly because of the pulses passed through the capacitors. Once the change has been made, the steady potentials applied through the resistors keep the new condition in force.

If we take our useful output from the Q2 collector, as shown, two trigger pulses are required to obtain a complete output cycle; the first may make Q2 collector negative and the second positive, while the third makes it negative again, initiating a new a.c. output cycle.

The trigger is obtained from the previous flipflop stage, which is exactly like the one shown. C3 and R7 both have small values and form a simple high-pass filter, also known as a *differentiator.* Such a filter transforms each full cycle of square wave from a previous flipflop to a narrow negative and positive pulse, of

which, as we have explained, only the positive pulse is useful. Since only one positive pulse is produced per full cycle of the previous stage and two such pulses are needed to produce a full cycle of output of the stage shown, each succeeding stage produces square-wave output at half the repetition rate or frequency of the previous one.

Because no changes in the flipflop values are needed, regardless of frequency, all components of each stage except the transistors are frequently made in the form of a flat package dipped in a hardening insulating material. The resistors are usually printed or screened on the base material with resistive paint, and the capacitors, which in these circuits are conveniently small, may consist of plates printed in silver paint. The whole operation is conducted by automatic machines and is relatively inexpensive, cheaper sometimes in large quantities than discrete resistors and capacitors. Fig. 3-17 shows one of them, made by Centralab, which calls them PEC's (Packaged Electronic Circuits). Organ manufacturers can buy just one type in great quantities, since one can be used for each note of the

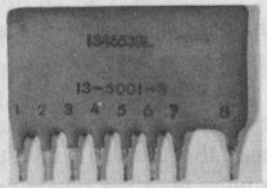


Fig. 3-17–A PEC containing a complete flipflop except for the transistors.

organ except the master oscillators. The PEC is valuable in saving assembly labor, since it can be inserted into a tone generator printed circuit board in a single thrust, as opposed to individual insertion of the 10 components shown in Fig. 3-16.

An even more modern technique is to place an entire flipflop stage in an integrated circuit (IC) contained in a TO-5 transistor case around 5/16 inch in diameter; this includes the transistors as well! At present the cost of IC's is too high to make organ use practical, but I have a feeling that situation will change rapidly. Flipflop IC's are certainly available to anyone who wants to try them. PEC's (and IC's) are also of some help in servicing, simply because you *can't* service them. If any part of one goes wrong, you clip the whole thing out and throw it away. This is considerably easier than checking the R's and C's one by one–and probably cheaper, considering the serviceman's labor time.

Waveshaping

Square-wave tone signals are useful for filtering to produce certain kinds of voices, as you will read in Chapter 5, but sawtooth waves are much more desirable for general use. In fact, an organ which produces and uses only square (or other symmetrical) waveforms exclusively cannot be well voiced. If either square or sawtooth tones can be dealt with by the keying system and following circuits, but not both because of some desired economy, the sawtooth is preferable by far, especially since in any frequency-divider organ with at least two pitch registers it is quite easy to reconstitute a symmetrical wave from the sawtooth by *out-*

phasing (Chapter 10) and original square waves are not even needed at all.

One method of obtaining sawtooth waves from the square ones which invariably come from dividers such as the flipflop and multivibrator is to use a simple waveshaper for each stage, like that diagrammed in A of Fig. 3-18. The input is connected directly to the collector of a flipflop transistor, which may be the same collector which feeds trigger signal to the following flipflop. C1 is small, as is R2, so that the two form a differentiator for the square waves. The only time this differentiator passes a signal to the base is during the time the

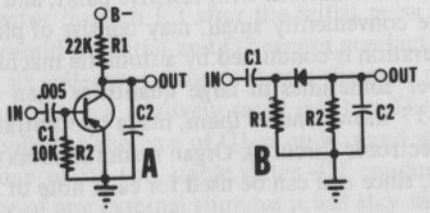


Fig. 3-18—Transistor (A) and diode (B) sawtooth waveshapers.

flipflop is changing state. When the square wave goes from negative to positive, a positive pulse is placed on the base. Since the transistor is cut off anyhow, not having any bias applied to it, the positive pulse does nothing. But when the square wave goes from positive to negative and a negative pulse appears at the transistor base, the transistor conducts fully—just for the very brief period of this negative pulse once per square-wave cycle.

During all the time when the transistor is cut off, it has no effect on R1 and C3. In this time, C3 charges negative through R1 from the power supply. The values are chosen so that the time required for C to become fully charged would be greater than one signal cycle. When the negative pulse comes along and causes the transistor to conduct, it acts as a short circuit directly across C3, discharging it very quickly.

The sequence, then, following any negative pulse, is (a) slow charge of C3 in the negative direction at the characteristic logarithmic rate of an R-C time constant, and (b) almost instant discharge of C3 when a negative pulse comes. This gradual charge and fast discharge makes the voltage across C3 a sawtooth at the same frequency as the driving flipflop, and the stage is thus a waveshaper. The value of C3 should be inversely proportional to frequency, since the lower frequencies mean a longer available charging time, and C3 must never be allowed to reach full charge by the time the discharging pulse arrives. If it were fully charged even slightly before discharge time, the wave would be flat-topped. To obtain equal amplitudes and slopes at all frequencies, each C3 should have just time to charge to a given common, fixed percentage of the B-voltage between pulses. As capacitor values determine charging time (if resistors are all the same), higher frequencies, with their shorter periods, must have smaller capacitors.

A cheaper waveshaper appears at B in Fig. 3-18. Since it has no amplifier all its power comes directly from the flipflop connected to its input, and its output is much smaller than that of A. Again C1 and R1 are a differentiator. Positive pulses have no effect as the diode polarity is against them. When a negative ex-

cursion of the square wave occurs and a pulse is presented to the diode by the differentiator, the negative pulse passes through the diode and almost instantly charges C2. During the remainder of the period C2 is free to discharge through R2, which it does slowly and exponentially. Then the next negative pulse suddenly charges C2 again and so on. The voltage across C2, sudden negative charges and slow discharges, is a sawtooth with negative-going flybacks. The sawtooth may be had in reverse phase, of course, if there is any reason for a choice, simply by reversing the diode.

Waveshaping is something of a nuisance because, first, it requires extra elements, including, desirably, a transistor. It does not, in addition, provide a linear sawtooth, but instead the customary swayback one. A linear rise could be obtained just as in some oscilloscope and TV sweep circuits by utilizing only the very early part of the charge, but this would result in uneconomically low signal amplitudes or large capacitors.

The reason the sawtooth is valuable in tone coloring is that its harmonic structure is very straightforward. The amplitude of each harmonic is inversely proportional to its ordinal number; that is, the amplitude of the second harmonic is 1/2 that of the fundamental, that of the 7th is 1/7 the fundamental, and so forth. This is actually true, however, only of a linear sawtooth like that at A in Fig. 3-19. The sawtooth actually obtained is much more like that at B, and the result is that the fundamental and lower harmonics are attenuated to less than



Fig. 3-19—Linear sawtooth (A) has exact harmonic amplitude relations. Exponential one (B) has lower harmonics with less than theoretical amplitudes. Waveshape at C approaches a pulse and has emphasized upper harmonics.

their linear-sawtooth values. Why this is true can easily be seen at C, in which we have merely accentuated the hump of the rise—as indeed may happen if we place some resistance across the waveshaper capacitor, such as following key-switching circuits. The waveform now looks much more pulselike than at B, just as B begins to approach a pulse in contrast to the pure sawtooth of A. A pulse is, of course, composed of higher harmonics, and the sharper and narrower the pulse the less the power of the lower harmonics. Indeed, as we have been discussing, pulses are deliberately produced from square waves for certain purposes by passing them through high-pass filters—differentiators.

While I do not for a moment mean to say that a waveform like B, which is obtained from shapers not overloaded with following-circuitry resistance, and also from blocking oscillators, cannot produce perfectly fine tone colors by the methods discussed in Chapter 5, the attenuation of fundamental is not helpful. When, for instance, one tries to obtain a moderately bright flute voice, somewhat heavier low-pass filters must be used to assure satisfactory fundamental

body, and they make the flutes duller than might be desired, as well as making scaling slightly more difficult. Reed voices, which are created by tuned bandpass circuits, may also have somewhat less body than ideal without special filter elements to correct the problem. These difficulties are very nearly, but not quite, inconsequential, however, as long as the natural nonlinearity of the capacitor charge is not accentuated by resistive loading.

A difficulty of the opposite sort appears with neon-produced sawtooth waves, which look the same as B, Fig. 3-19, except that the vertical portion is not so vertical and the *higher* harmonics of the ideal sawtooth are attenuated.

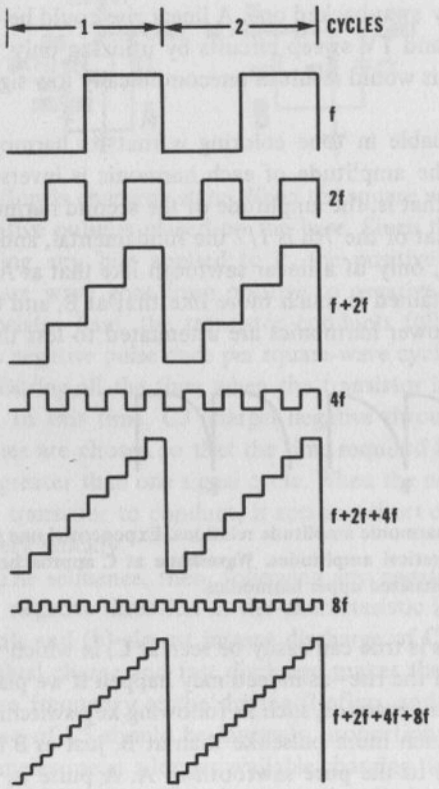


Fig. 3-20—Adding square waves of frequencies f and 2f gives simple staircase. With each octave addition, steps become finer and resemblance to a linear sawtooth becomes more complete.

Flute voices come out very easily. But strong string voices must be produced by high-pass filters which emphasize treble harmonics more than otherwise, and the result is a lack of body. An ideal sharp string voice has some buzz but also has some body furnished by lower harmonics, and it is best produced from a sawtooth with the idealized harmonic amplitude relationships stated above—that is, a linear sawtooth as in A. It should be noted that in B and C, waves formed from flipflop square waves, the vertical strokes are almost always invisible on the usual oscilloscope because they take such little time and at the treble end of things, at least, the idealized harmonic relationships are found in practice out to very high

numbers—the 30th harmonic at the very least, and often very much higher.

Staircasing

There is a simpler and cheaper method by which square waves can be robbed of their symmetry and made to contain even as well as odd harmonics, called, for reasons which will become obvious very soon, *staircasing.* Simple forms of staircasing, which are most often used, add some even harmonics but leave out many others entirely. Simple staircases leave in the tone some of the "hollow" tone quality characteristic of symmetrical waves and thus are not conducive to the best voicing. Full staircasing, however, can produce a final waveshape which includes almost all harmonics in the audible range, and the result is in no way inferior to a linear sawtooth. However, even simple staircasing gives a tone exactly the correct theoretical amplitudes of whatever harmonics *are* included; there is no attenuation of fundamental.

The method of staircasing is to mix together square waves of two or more frequencies which are octavely related and synchronized, with amplitudes inversely proportional to frequency. An organ tone generator naturally puts out such signals from its divider stages. (Amplitudes are usually equal, but signals may be mixed in any desired proportions, of course.) On a C generator, for example, the lowest divider may produce 32.7 cps. The next produces twice that frequency, the third twice the frequency of the second, and so on. The effect of staircasing can be shown graphically and numerically, and we shall do both.

At the top of Fig. 3-20 we show two cycles of output of the lowest divider of a generator, which we shall call frequency *f.* The dimension arrows marked 1 and 2 show the two cycles. Below the *f* wave is the output of the next higher divider at frequency 2*f,* drawn exactly below *f* and with amplitude half that of *f.*

The third waveform shows the result of linearly mixing or adding those shown for *f* and 2*f,* easily constructed on graph paper if you are familiar with that sort of thing. This is exactly what an oscilloscope will show (assuming the two original divider outputs are true square waves and contain no overshoots, which they sometimes do but which does no harm aurally). You will notice that the *f* + 2*f* waveform is no longer symmetrical and, in fact, if you averaged out the slope by ignoring the steps of the staircase, you would get a linear sawtooth.

The figure next shows the output of the third divider from the bottom, at frequency 4*f,* with its amplitude half that of the previous one. And then you see the result of mixing *f,* 2*f,* and 4*f.* In the same way, 8*f* is shown, again cut in half in amplitude, followed by the result of mixing all four of the lowest frequency dividers of the generator. Notice that each time you add another generator output to the mix, in the proper amplitude relationship, the stairs get finer and you get closer to a true sawtooth, but always at the repetition rate of *f.*

Fig. 3-21 shows how staircasing is done. In this diagram we show a generator in which only the simplest staircasing has been carried out. Mixed output *f,* the lowest final output, is obtained from dividers *f* and 2*f,* both signals obtained from voltage dividers having a resistor of value 0.1R as the common shunt leg. Signal from divider *f* goes through a series leg of value R, while that from divider

2*f* goes through a series leg of 2R. The final *f* output therefore contains square wave at a given amplitude from divider *f* and square wave at half that amplitude from divider 2*f*. The result is a staircase with three steps, the simplest one of Fig. 3-20. Other outputs shown are obtained in the same way and have the same waveform but higher repetition rates or fundamental frequencies.

It is quite easy to make the staircase have steps as fine as desired, limited only by the number of stages in the generator. To obtain the 7-step staircase shown next in Fig. 3-20 for final output *f*, we would add to the diagram of Fig. 3-21 another resistor from mixed output *f* to divider 4*f*, and its value would be 4R. And to obtain a still fuller staircase, one could add yet another resistor of value 8R from divider 8*f*. One can keep this up for as many stages as are in the generator, and do the same for each output.

One thing becomes immediately obvious, of course. If, for example, the generator has seven stages, the lowest-frequency one can have the finest staircase, drawing on all seven frequencies. The next can draw on only six frequencies, and so on until the topmost frequency has nothing above it to mix with and must remain a square wave. However, as we will see when we play the numbers game

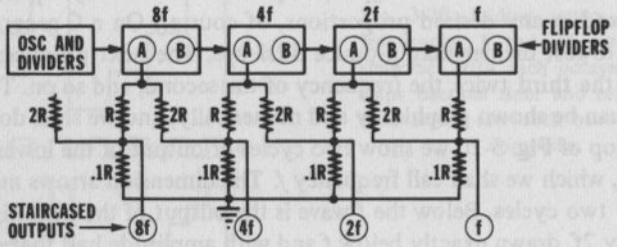


Fig. 3-21—How frequency-divider outputs may be added to obtain the simple staircase resulting from f + 2f.

in a moment, doing the fullest staircasing job possible still makes sense. Making the stairs finer fills in missing harmonics, but as the frequency of the fundamental gets higher and more and more harmonics must be left out, the missing harmonics become so high that they are either above the limit of hearing or are impossible to detect anyway. This is especially true when the top generated tone is in a high octave such as Octave 8 in Table 1-1.

The resistor relationships shown in Fig. 3-21 are typical for good operation and you may wonder why it is necessary to take such a loss of signal level in the voltage division.

One absolute essential in any organ is to prevent any audible tone from any generator stage from appearing in the output of the next higher-frequency stage. Even a suspicion of this will make the higher tone sound as though it had some permanent content of tone an octave lower. This is easily detectable and highly objectionable. (A tone with some content of a higher-frequency stage is not nearly so bad, since the ear merely interprets that as a little extra harmonic development.)

34571

In Fig. 3-21 it is obviously possible for some tone from the *f* divider to get into the final 2*f* output through the resistor network. That network must be proportioned to keep such leakage lower than audibility, as well as to proportion the staircase mixing correctly.

A typical case is the Schober Theatre Organ described in Chapter 10. Considering only the mixing of the fundamental frequency and the next higher one, the value of R (referring to Fig. 3-21) is 22,000 ohms. 2R is therefore 47,000 (the nearest RTMA value) and 0.1R is 2200. The collector resistors of the flipflops are 1,000 ohms. To see what is happening we need only figure the attenuation of signal from divider *f* by the time it gets to the collector of the transistor in divider 2*f*, since from that point to mixed output 2*f* both the *f* and 2*f* signals undergo identical attenuation by an R-0.1R voltage divider.

Divider-*f* signal reaching the junction of *f*'s R and 0.1R is reduced to 1/11 of its initial value by the voltage divider. From there it passes through a second voltage divider, of which the 2R leading to frequency divider 2*f* is the series leg of 47,000 ohms and the collector resistor of divider 2*f* is the shunt leg of around 1,000 ohms. Call this voltage division roughly another 48 times, and the product of the two, 11 and 48, makes a total of 528 times, or in the area of 54 db. This, believe me, is none too much; it effectively kills the curse of the lower tone in the higher-frequency output but not with much to spare. Obviously, if more components are added to the staircase mix, they come through still higher resistors and represent no additional threat of this kind. But no staircasing system can be tolerated without adequate attention to this point.

A slightly simpler staircasing system is possible for simpler organs, but at the cost of complicating the key switching; it is used in some Thomas organs. Resistors R and 2R are still present, but the 0.1R is not. The key switches are so arranged that when a key is not pressed and the mixed tone is therefore not terminated across a low enough resistor to provide the attenuating effect of 0.1R, the switch and the junction of the R and 2R are grounded. If two or more ranks of switches must be fed by the generator, as with multiple pitch registers and/or two manuals, a separate R-2R combination must be provided for each rank and every contact must ground in the off position. The advantage of this is that some of the attenuation can be avoided.

It is also possible to staircase somewhere other than in the generators, as Baldwin does in the organ of Chapter 9. Square waves directly from the frequency dividers are keyed in several pitch registers. To provide staircasing where it is wanted, two or more pitch registers are simply combined at the inputs of the stop filters in the correct amplitude proportions. For example, a filter for an 8′ stop is fed both 8′ and 4′ square wave keyed and collected tone, with the 4′ reduced to half amplitude.

This system has its advantages. First, tones are keyed at high level, without the 20-db attenuation shown in Fig. 3-21. Second, using only one keying system, some stops can be fed square waves and others staircased signals—and in the latter case, the staircases may be more or less complex depending on the available pitch registers and can even be varied from the normal amplitude mixing

ratios whenever the designer thinks that will improve the sound of a stop.

The disadvantage is that whatever the uppermost keyed pitch register is, it cannot be staircased and must be square-wave from top to bottom, since there is no higher pitch register of tones to mix with it to produce a staircase. When the job is done at the generators, instead, it is only the top *octave* of tones that find nothing to mix with.

You will notice in Fig. 3-20 that the staircasing resembles sawtooth waves of positive polarity—slanting up to the right and with negative-going flybacks. This is always what is obtained when the frequency divider outputs are taken from the triodes designated A in Fig. 3-21, those, that is, which do not supply trigger pulses to the next divider stage. If all outputs are taken instead from triodes B, the phase will be reversed, with negative-going rises and positive-going flybacks. It is hard to imagine why it should make any difference which one chooses, but you must take all outputs from corresponding triodes to avoid chaos. The choice is usually based on the master oscillator circuit, and is made in such a way that divider phases with respect to the oscillator output will be correct. The reason for this little detail of performance is that a pulse of only one polarity emitted by triode B can cause the next stage to change state. For an interesting few minutes, take a piece of paper and postulate different initial phases at power turn-on and subsequent phase relationships of output square waves. You will see that the above rule is always correct for PNP transistors and exactly reversed for NPN's or tubes.

The Numbers Game

Graphic analyses are pretty but superficial. Analysis of harmonic content tells a much more thorough story about how closely staircased square waves approach the theoretical ideal of a linear sawtooth, with the latter's beautiful progression of all odd and even harmonics exactly inversely related in amplitude to their ordinal numbers. To start with, keep in mind that the harmonic amplitudes in a square wave also have this idyllic amplitude relationship with the fundamental, the only difference being that even harmonics are completely missing. Still, the third harmonic is 1/3 the fundamental amplitude, the fifth 1/5 the fundamental, etc.

To simplify our analysis, let us assume a 7-octave tone generator, of which the frequency of the tone supplied by the lowest octave frequency divider is 1 cps. The remaining six octaves then produce 2, 4, 8, 16, 32, and 64 cps. Initially, we shall assume that the tone we are analyzing is that of the lowest divider, 1 cps; we are determining which of the remaining signals supply which of the even harmonics normally missing from the low divider, when they are mixed with the 1-cps tone in a staircasing arrangement. Selection of the 1-cps frequency is convenient because the numbers representing frequency and those representing the harmonics of 1 cps are identical. That is, 2 cps is the second harmonic, 5 cps is the fifth harmonic, and so forth.

There is also the convenience that the desired amplitude for each frequency is simply the reciprocal of that frequency. The 16th harmonic of 1 cps is 16 cps,

for instance, and in the ideal sawtooth we seek its amplitude should be 1/16 that of the fundamental.

Let us leave aside for a moment the question of the amplitude of harmonics and see how the harmonics of the lowest tone are added by the higher-octave generator stages. Table 3-1 shows the seven octavely related outputs of our hypothetical 7-octave frequency-divider generator. The columns at the top are

TABLE 3-1

Generator Octaves						
1	2	3	4	5	6	7
1	2					
3		4				
5	6					
7			8			
9	10					
11		12				
13	14					
15				16		
17	18					
19		20				
21						

Generator Octaves						
1	2	3	4	5	6	7
	22					
23			24			
25	26					
27		28				
29	30					
31					32	
33	34					
35		36				
37	38					
39			40			
41	42					

Generator Octaves						
1	2	3	4	5	6	7
43		44				
45	46					
47				48		
49	50					
51		52				
53	54					
55			56			
57	58					
59		60				
61	62					
63						64

headed 1 through 7 to indicate the octaves, octave 1 being the lowest frequency (1-cps) output. Accordingly, the first figure to appear in the octave-1 column is 1 (cps). The succeeding figures are the frequencies of all the harmonics of 1 cps which appear in this 1-cps square wave—that is, 3, 5, 7, and all the rest of the odds.

The octave-2 column is made up in this same way. The top figure is 2 (cps) the fundamental frequency of octave 2. The third harmonic of 2 is 6, the fifth is 10, the seventh is 14, and so on. And so goes the octave-3 column, its fundamental at 4 cps, third harmonic at 12, and fifth harmonic at 20. The only "fiddling" we have done with this straightforward information is to space out the frequencies vertically so that if we run a horizontal straightedge down the sheet, all frequencies (and harmonics of 1 cps) will appear in sequence. This is an aid to the eye but alters none of the information.

Let us suppose we do the simplest possible staircasing job by adding tone from octave 2 to that of octave 1, then examining the result to see what harmon-

ics of octave 1 are present in the result. The fundamental (harmonic 1) is furnished by octave 1, the second harmonic by octave 2, the third by octave 1. The fourth harmonic is missing, and to add it, as well as the 12th, 20th, 28th, 36th, and so on, we would have to add tone from octave 3 to the mix.

If we do add octave 3 to the mix, we find that all harmonics are present until we come to the 8th, to furnish which we must also take tone from octave 4. You can continue your own analysis, and you will see just which harmonics are missing with a particular degree of staircasing and which octave's tones must be added to supply given missing harmonics. You can also begin to make some judgements about completeness *vs* economy. For a reasonably complete job, I would say that octaves 2, 3, and 4 should be added to octave 1. Octave 5 is less important because the lowest harmonic it adds is the 16th, octave 6 much less vital because its first addition is the 32nd, and octave 7 minimally important, as the first missing harmonic without it is the 64th.

As a practical matter, the majority of organs, which add tone only from one stage higher, the simplest possible staircasing, do not suffer fatally from lack of the harmonics shown in the 3 through 7 columns, though the tone color which pervades all the stops is a degree of hollowness which makes really good and well varied registration difficult or impossible. An organ which also includes a third octave in the staircasing can, if well designed, have quite satisfactory voicing, however, as the one extra ingredient makes a very substantial difference. Note well, though, that whatever one continues to add to the staircase can very definitely be heard as long as (a) the fundamental is low enough so that each added tone contains at least one harmonic within the range of hearing, and (b) the organ is not voiced with nothing but mushy stops containing no upper harmonics. Beyond a certain point, however, the law of diminishing returns certainly does take over.

There is one case in which it is highly desirable to put as many tones in the staircase as are available—when an outphaser is used to convert the staircased waves back to square form. The outphaser and its use are described in Chapter 10. Essentially it is a method of mixing tones of one pitch register with those an octave higher in such a way that components of the higher tone which are of the same frequency as even harmonics of the lower cancel those even harmonics, leaving only odds and the resultant hollow tone quality required for some stops. Obviously, the cancellation cannot be complete if some harmonics of the upper staircased tone are missing, and the basic tone will not be as hollow sounding as would be desirable.

The question of harmonic amplitudes is easily settled by a simple example with the first couple of harmonics in a basic octave-1-octave-2 staircase. In the square-wave output of octave 1 divider, the 3-cps 3rd harmonic component has 1/3 the amplitude of the 1-cps fundamental and the 5-cps component has 1/5 the 1-cps amplitude. And so on down the line.

In the octave-2 output, the 6-cps 3rd harmonic has 1/3 the amplitude of the *2-cps* fundamental, and the 10-cps 5th harmonic has 1/5 the amplitude of the *2-cps* content.

In the staircasing system we use resistors to make the total amplitude—and thus the amplitudes of all harmonics—of each output added to the mix half that of the previous octave. With these resistors, therefore, we reduce the 2-cps fundamental of octave 2 to 1/2 the amplitude of 1 cps. Since the 6-cps octave-2 component is 1/3 the amplitude of the 2-cps component, that makes it 1/6 the amplitude of the 1-cps octave-1 fundamental. Since the 10-cps 5th harmonic of octave 2 is 1/5 the amplitude of the 2-cps fundamental, it is 1/10 that of the 1-cps fundamental. It is obvious, therefore, that for every tone added to the mix through the halving resistors, the amplitude of each harmonic is the reciprocal of the number shown in the table, in relation to the 1-cps fundamental of the octave-1 tone which we are staircasing. This is exactly the right relationship for a linear sawtooth, and everything thus falls nicely into place.

For practical use, the numbers in Table 3-1 should be taken to indicate harmonics rather than actual frequency, of course. And we may always take the octave-1 column to represent whatever tone we are staircasing, no matter what its frequency, while octaves 2, 3, and so on, merely represent the octaves above it. If, for instance, we are considering A-440, we can see that the simplest staircasing, mixing in the square-wave tone whose fundamental is 880 cps, will supply the second harmonic at 880 and the sixth at 2640, but not the fourth at 1760, for which we would need to mix in the tone from the second higher generator stage at 1760 cps. If we did add these two higher stages to the 440-cps tone, the earliest missing harmonic would be the eighth, at 3520.

In a practical tone generator, it is quite impossible to have the same number of higher tone outputs in the mix at all frequencies, simply because we run out of tones. If the simplest staircasing is used, just two stages, we do not run out until the master oscillator, and perhaps its waveshape can be set so it has some natural even harmonics. However, if, as in the Schober Organ of Chapter 10, we add for each tone as many higher octaves as exist, we find that more and more harmonics are lacking as fundamental frequency gets higher. The actual *frequencies* of the harmonics that are missing, however, remain about the same. This is merely equivalent to spectrum shaping, which is an integral part of the formant voicing method, so the ear does not hear any *change* in the tone color, which is the important thing. Doing the maximum possible staircasing job does pay off in terms of audibly better tone because, with no significant harmonics missing, full advantage may be taken of formant filtering (Chapter 5) to produce the full tonal subtlety of which the method is capable. With very simple staircasing and a missing fourth harmonic—plus the definite hollow quality left by the failure to add enough even harmonics—the raw tone fed to the voicing circuits has a definite character of its own which cannot be erased by filtering and gives most voices at least some element of sameness. The situation is much like a less than ideal loudspeaker, which tends to color all music to at least some extent with its own personality.

Chapter 4

Keying And Coupling

Every organ contains a multiplicity of tone generators, one for every pitch at the very least. In some manner, the playing keys must enable the organist to select the desired pitches for transmission through the rest of the instrument. Every key causes one or several switches to operate when it is pressed. Both the electrical functions and the mechanical arrangements of these key switches are important elements of design, not only because there are so many of them, but also because they must function reliably over a great many more cycles of operation than is true for switches in almost any other kind of apparatus. This leads to careful consideration of the keying actions and switching materials, for which each manufacturer has his own design.

In this chapter we will pin down the basic requirements which apply to the switching of generator tones in all instruments. Because there is so much variety among manufacturers, we will go into real detail only in the chapters on the individual organs.

There are basically two kinds of switching schemes in use, and they depend on the generator philosophy of the organ design. The first type, common to the large majority of organs, is found in instruments where the tone generators run continuously. This includes all frequency-divider organs and some where each frequency is generated by an independent oscillator. In these cases, the switching system switches the desired generator outputs directly or through electronic keyers to following voicing circuits.

The second type is found only in some instruments which employ individual oscillators that are normally turned off, usually by depriving them of some necessary operating voltage for the tube or transistor. In these organs the key switches supply the missing voltage to selected generators and make them "speak." Usually the outputs of all generators of each individual rank are permanently connected together and fed, usually through voicing controls, to following circuits.

Coupling systems go right along with keying systems for the purpose of discussion because they, too, deal directly with tones which come from the generators before they are introduced into following voicing or preamplifier circuitry.

The Musical Scheme

The first question is: What must each key do? Let us for the moment ignore the question of how it is done and look at the necessary results. What is the basic scheme of key control of pitches?

In capsule form the question is answered by Table 1-2, page 12. Let us take a

single key, middle C, as an example, in conjunction with Tables 1-1 and 1-2.

In a standard 61-note manual, middle C is the first key in the third octave of keys, and it is therefore located in Key Octave 3 in Table 1-2. In the most usual kind of 44-note spinet manual, which begins with an F key, middle C is the first C on the upper manual and the second C on the lower. On manuals of other designs, it has other relative positions, though anyone familiar with keyboards can always locate it by eye, simply because it tends to be at the center of things. In Table 1-2 it is still in Key Octave 3 because it would be impractical to number the octaves for pitch purposes in any other way than according to the only recognized standard keyboard—that having 61 notes.

When middle C is pressed it does not control a single tone as on other keyboard instruments like the piano; it controls several. This is because the organ always has two or more pitch registers. The available pitch registers for a particular manual of a particular organ can always be noted by examining the stop tablets associated with that manual. In most organs, each stop sounds at its own pitch register, which is marked on the tablet. (In a very few organs, especially some older ones, individual voices have no particular pitch registers; available registers can be noted by looking at the coupler tablets.)

If we assume that the manual we are examining has 16′, 8′, and 4′ stops, then each key must switch three tones. First it must select the 8′ tone corresponding to that key. Looking at Table 1-2 under the 8′ column and opposite Key Octave 3, we see the figure 4. This indicates that for the middle C key, the 8′ frequency is in the Octave 4 column of Table 1-1, where the C is 261.6. We can also deduce that if the tone-generator system supplies tones down through the 16′ octave, as shown in Table 1-1, the 8′ generator output for the middle C key is the fourth from the lowest generated C. In short, then, the middle C key switches the 262-cps generator output to the 8′ voicing circuits (or simply turns it on, depending on the kind of organ).

It also switches the appropriate 16′ tone. From Table 1-2 the 16′ tone for Key Octave 3 is generator Octave 3, 130.8 for the C. And it also switches the appropriate 4′ frequency, 523.3.

The middle C key in this example therefore affects at least three separate circuits. And if, as on some elaborate organs, the manual handles six or more pitch registers, that number of switching or circuit operations takes place each time the key is pressed.

Key Switch Requirements

For several reasons key switches are among the most difficult elements of an organ to design well. The first problem is caused by the fact that there are so many switches—at least one for each key and usually many times that. The second difficulty is that the pressure required to operate the switches must be light so as not to affect the key action and cause playing trouble. But despite the low pressure, switch contacting must be positive, preferably with enough pressure and wiping to assure self cleaning, especially where the organ design calls for switching low-level audio signals which have no ability to cut through poor con-

tact, dust, or surface corrosion, all of which cause noise during intermittent contact even if good contact is finally made.

The switches must not make physical noise, since no one wants his playing to be accompanied by a constant series of little metallic clicks. And where audio is being keyed, capacitance between the contacts or blades of an open switch must be very small to prevent tone leakage. This generally makes blade-type switches impractical. Because there are so many of them, the switches should be as cheap as possible, and to avoid unnecessary labor time they should, if possible, be mounted in some way that does not require individual placement and attachment for each switch.

Above all, the switches must be *reliable.* They must work initially, later, and practically forever without special adjustments or frequent cleaning or any other attention. Nothing is more frustrating to an owner than keys which work sometimes or not at all. And few things make a service technician chew his nails down so close to the bone, because key switches are almost invariably hard to replace or even clean or adjust. In the end, anything that makes the switches reliable, even if it causes other desirable attributes to walk the plank, is worthwhile. Designing key switches really requires two different kinds of engineers. The first should be ingenious, imaginative, even flighty, as it is his job to evolve the basic scheme, which requires a good quota of intuitive gymnastics. The second should be a sour schoolmarm type, never satisfied with anything until he has proved it will work under water and will withstand both exposure to an H-bomb at 5 yards from ground zero and the attention of small children (which is worse).

Switch Materials

One key to reliability is using precious metals for contacts, but not just any old precious metal. Coin silver, which is the most common material for run-of-the-mill (non-organ) switches and relays, may get you by if you are switching enough voltage so that even a pair of amber rods would make adequate contact. Sterling silver is a little better. But silver acquires a black coating of silver sulfide whenever there is any sulfur in the air, and the sulfide is an insulator with just enough dielectric strength to prevent good contact for audio and even some of the d.c. often used in organs. The sulfide is purely a surface coating and can be wiped off, so you may get by if there is enough wiping action, which is rare in a switch that must be operated with light pressure and not very much travel.

A much more satisfactory contact material is palladium, or more specifically, a palladium-ruthenium compound. This material is not attacked by anything likely to be in the air and its contacting properties remain good. It can easily be welded because it softens below its melting temperature, and for this reason it is commonly available from blade-switch manufacturers such as Switchcraft and Guardian in the form of welded crossbar contacts. These are small bars of the compound welded to each of the blades which are to contact, with the bars at right angles to each other. Palladium is also available as spring wire, but because it is quite expensive it is almost necessary to use just a small piece for each contact, welding it to some other part of the mechanism. Platinum is also an excellent contact material, but it is slightly softer than palladium and is more expensive since palladium, in addition to being found in platinum ores is also found in significant amounts in nickel ores.

In specifying palladium contacts it is a good idea to stipulate that there should be no silver in the compound. I know of one manufacturer who found this out the hard way some years ago. The switches eventually failed because the silver migrated to the surface of the palladium and then acted as silver can be expected to. I am no metallurgist and for all I know other elements in palladium compounds may do the same thing. The material used by Switchcraft and Guardian is quite satisfactory. If it is obtained elsewhere, the designer should specify that it is for switch contacts and note especially (if it is the case) that it is for "dry" switching—where voltage and current levels are so low that conduction can never be dependent on the breakdown of any surface dielectric.

My own experience indicates that where contact pressure is light, nothing is so satisfactory as gold. As everyone knows, gold does not corrode or tarnish. In practical use, the only way of marring its contact properties is to let dust accumulate on its surface and thus physically keep the conducting surfaces apart. Gold is soft and malleable, of course, and cannot be used in as many forms as palladium. Excessive contact pressure would give undue wear. Probably the most popular gold alloy used for contacts is the so called Western Electric No. 1, which is hard enough to give good wear.

It would be unnecessarily expensive to use anything but a surface coating on a base metal for contacts. Gold is usually used in wire form, the base material being one of the copper compounds, often brass. The gold cannot be plated on the base material, since gold will not electroplate onto itself and thus only a thin coating could be built up—perhaps a few microns. This amounts to what is known as a gold flash, and it would wear off in no time in most applications. Instead, the base material is often *clad* with a gold alloy, a mechanical process in which the wire is drawn through a die along with the gold and the gold is physically drawn onto the surface. In this way coatings as thick as .001 inch are commonly clad on the base, and this is plenty.

Gold against gold is the best contact system I know of, as long as the pressure is moderate and the system does not permit arcing. Actually, if there is arcing, voltage and current are high enough to make use of the more expensive metals rather silly. Gold is usually used only for dry switching.

To my knowledge, there are no successful grandmother's-remedy-type switching materials for dry circuits. Nichrome wire, for example, has gone the rounds of the rumor factories. As far as I know, you can forget it.

One other kind of switching material is in actual use—and though it is a part of at least one popular commercial organ, the thought of it (plus data I have had from a research source I respect highly) makes me nervous. It is a resilient vinyl compound impregnated with a dispersed conductor such as carbon. The idea is that the contact resistance decreases with increased pressure as the key goes down, and the material thus allows a somewhat gradual contact rather than a simple yes or no affair. I strongly recommend that any designer contemplating

its use investigate very carefully before commercial production and exercise rigid quality control thereafter. In any event, the range of resistance between no pressure and full pressure is not very large, and I suspect the tolerance allowed by circuit design for the final full-pressure resistance ought to be large, as this is very likely to depend on which particular point on the material surface the contact happens to be striking.

Audio Switching Schemes

In all organs in which the generators run constantly and electronic (diode or transistor) keying is not used, the purpose of the key switches is to select the desired generator outputs and pass them on to the following circuits, usually voicing circuits which operate on the raw generator tones to produce the various tone colors of the organ. The general scheme of the switching system is the same

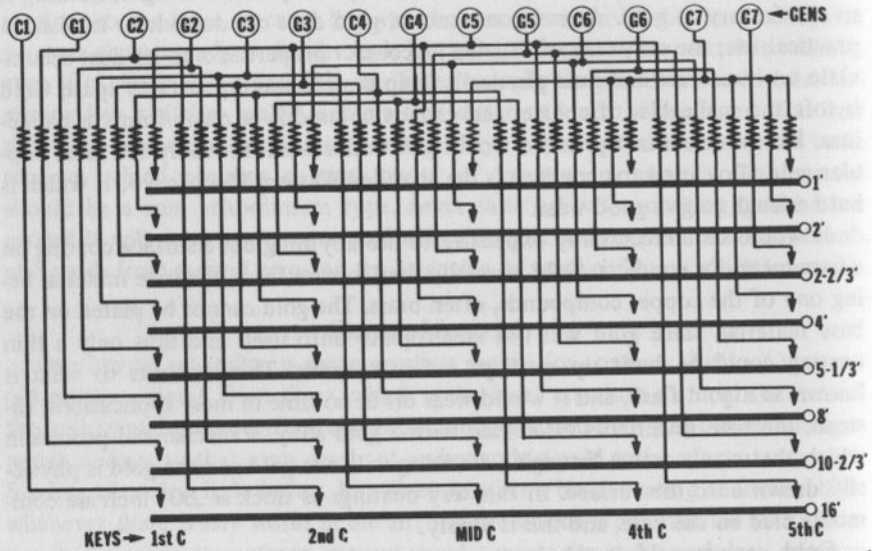


Fig. 4-1—Layout of a direct keying system shows how different pitch registers are produced on the keying busses.

in all cases, though each organ may vary in detail. An understanding of this basic scheme, which is shown in Fig. 4-1, is basic to an understanding of organs.

Fig. 4-1 shows a part of the switching system operated by one keyboard. In the diagram are shown all the switches operated by all the C keys of a 5-octave keyboard (except for the top C). The purpose is not to illustrate the C's especially, but to illustrate the idea for all the keys of any note.

Five vertical rows of switches are shown, one for each of the five C keys. Eight transverse busses are shown, one for each of the eight commonly used pitch registers. Few electronic organs provide all eight registers, and some of the pitch registers shown at the right are never used on manuals but only on pedal keyboards, such as the 10 2/3', and perhaps 5 1/3'. Showing them all, however,

will give you a correct idea of the relationships they bear one another.

The short vertical lines with arrows are moving switch contacts, while the long horizontal lines are stationary busses. When any of the five keys is pressed, all eight movable contacts in the vertical row above the key designation come down and each comes in contact with the bus just under the arrowhead. We have omitted the dashed lines customarily used to indicate switch ganging to avoid further cluttering an already messy diagram.

We have assumed that this organ has a set of tone generators which provides 7 octaves of tone for each of the 12 notes of the chromatic scale. The lowest, generator octave 1, designated in the figure by the circled symbol C1, provides the 32.7-cycle pitch shown in Table 1-1 as the C frequency for Octave 1. The remaining circles contain similar symbols keyed to Table 1-1. While the keys operating the switches shown are C keys, the G generator outputs must also be shown to provide for the mutation registers 2 2/3', 5 1/3', and 10 2/3', as you will see.

In all we say, note that the C's discussed are typical. If you want to consider, for example, the C#'s, then the same scheme would apply except that (a) the keys at the bottom would be 1st C#, 2nd C#, etc.; (b) the C1 through C7 generator outputs would be C#1 through C#7; and (c) the G generator outputs would become G#1 through G#7, maintaining the relationship of a musical fifth above the basic note.

Instead of showing each generator output directly connected to a switch contact, we have shown a resistor in series with each contact. The resistor provides isolation and we will discuss that in detail a little later.

Consider first the 8' register, which is basic. Tracing this for the middle C key, first find the 8' bus from the markings at the right. Trace the bus to the left to find the contact just above it in the MID C row. From this contact, trace the wiring, which goes through a resistor to generator output C4. This means that when the middle C key is pressed, and all the middle C contacts touch their respective busses, this particular contact passes signal from generator output C4 into the 8' bus and the correct frequency of 261.6 cps appears on that bus.

If we now trace the origin of the signal appearing on the 8' contact for the 2nd C, we find it is generator output C3, an octave lower as would be expected. The 8' contact for the 1st C carries signal from C2, and so on. If we connected an amplifier to the 8' bus, then, we would expect to hear all the C notes in correct octave relationships for the 8' register. And since the keyboard actually contains similar contacts for all the notes, we could play melodies and harmonies and hear the whole thing in the same pitch register as though we were playing piano keys.

Now let us look at the 16' register. The 16' contact for middle C is connected to generator output C3, which provides tone an octave below that present for the same key on the 8' bus. Examination will show that the 16' contact for *every* C key puts into the 16' bus tone an octave lower than that put into the 8' bus by the 8' contact. So if we connected the amplifier to the 16' bus we would hear the same music, but everything would be an octave lower.

The same reasoning applies for the 4′ register, and at this point you should trace the origins of the 4′ contact for each key. You will find for each key it derives tone from a generator an octave *above* that used for the 8′ contact. Thus, the amplifier, connected to the 4′ bus, would provide the music one octave higher than the 8′ bus and two octaves higher than the 16′ bus.

Next, trace the 2′ contacts. You will find that the generator signals on them are still an octave higher than on the 4′ contacts. Here, however, we encounter the first instance of running out of generators. For the 1st, 2nd, middle, and 4th C keys, the 2′ sequence is C4, C5, C6, and C7, which accords perfectly with the scheme we have shown. The 2′ contact of the 5th C key, however, does not go to a generator C8, as it should, because this organ generates only seven octaves of each note and not an eighth. To have the 2′ register silent whenever the 5th C key is pressed would be abhorrent, though it has been done in some pipe organs when a rank runs out of pipes. So, being ingenious, we simply feed the highest available C, C7, to the 5th C 2′ contact.

In the 1′ register only the first three C keys can be fed by the correct generator outputs, since the last two would require a C8 and C9, respectively. So again we just connect the highest two 1′ contacts to the highest available C, C7.

Now, picture what that means. In the 2′ register it means that not only the 5th C key but all 12 keys of the highest octave produce the same pitches as the previous octave. That is, the top 2′ octave *repeats* the previous one. And in the 1′ register the top *two* octaves of all notes are the same as the third octave.

Different organs vary as to what registers contain repeats of this kind, since some have as many as eight octaves of generators and some have as few as three. Most organs do not have pitch registers as high as 1′ or even 2′ and this relieves the need to have repeats, though it also reduces the versatility of the instrument. Some organs may have 7-octave generators but the tones produced may be octaves 2 through 8 instead of 1 through 7.* In such cases a 5-octave keyboard would have a repeat at the low end; the lowest octave of keys in the 16′ register would produce the same pitches as the next octave up.

Repeats of this kind are perfectly legitimate if not carried too far. They are common in pipe organs where pipes at the upper end of the high registers would simply have to be too short to be practical. Carrying, say, a 1′ register to its full conclusion would end in tones of around 15,000 cps, which a great many people cannot even hear, though we could easily enough produce such tones with electronic generators. More important, from a musical standpoint it would be practically useless. Stops of the higher pitch registers are never used by themselves, at least near the upper end of the keyboard; melodies or harmonies at such high pitches would sound silly (except perhaps in some very modern music). The high-pitched stops are used instead in combinations or ensembles of stops which consist basically of lower-pitched voices, and the purpose of the high-pitched voices is only to add brilliance to the ensemble. As you approach the top of the keyboard with such an ensemble, the heightened pitches of the fundamental

*Octave 1 would be produced by a pedal generator, for the pedals only.

voices tend to make the music more brilliant and the effect of the 1′ and 2′ voices is no longer needed.

The 1′ and 2′ stops, however, are very valuable in the middle of the keyboard and lower, for here their presence is very obvious and can make a very substantial difference in the effect of any ensemble. Again, however, the size and cost of the organ determine which pitch registers are included and where the repeats begin. I believe the absolute minimum for both manuals of any organ includes 8′ and 4′ registers, with the 4′ pitches carried right to the top of the manuals without repeats (except possibly for the topmost, 6th, C), and at least one of the manuals should have 16′ pitch as well, though there is no harm in having it repeat in the lowest octave since chords and melody notes played in the 32-65-cps region are rarely effective anyway. Pedals should always have at least a 16′ register and preferably 8′ as well. Anything past these requirements is highly desirable but its absence need not cause condemnation of the instrument if it is well voiced within its limits. A 32′ pedal register is not uncommon on pipe organs but is very rare on electronics.

All the pitch registers we have discussed so far are octave-sounding registers. The mutation registers used in pipe organs are of two kinds, *fifth-sounding* and *third-sounding*. A fifth-sounding register is one which is a musical fifth above an octave-sounding register. For instance, when we play a C key and listen to the 5 1/3′ bus, the pitch we hear is a musical fifth above the 8′ C—a G. The same is true of the 2 2/3′ register with respect to the 4′, and the 10 2/3′ with respect to the 16′. A third-sounding register such as 1 3/5′ would provide pitches a major third above the octave-sounding, for instance the pitch of an E when a C key is played.

Musical authorities argue over the merits of third-sounding registers, some claiming they are unpleasant. They are rarely provided at all on electronic organs except in extremely large, usually custom-designed organs. We have not shown any third-sounding registers in the diagram.

Isolation And Loading

In most organs the outputs of the individual busses shown in Fig. 4-1 go eventually to voicing circuits, which are usually formant filters (Chapter 5). Each individual bus is fed (usually through amplifier stages) to a set of voicing filters which can be turned on or off by the stop tablets. Since each bus is handled individually, it follows that each voice, in addition to having its own tone color and loudness, also speaks in its own pitch register. A filter, for example, which derives its signal from the 4′ bus produces a voice one octave higher in pitch than one fed by the 8′ bus. The aural result is in accord with standard pipe-organ practice and gives the same kind of variety and ensemble.

It is not sufficient, however, merely to do the switching we have described and illustrated. Two kinds of interaction must be prevented: (a) no tones must be allowed to leak through and be heard in addition to the ones being played at any moment, and (b) the volume level of any tone must not be affected by whether other tones are also being keyed at the same time or whether the tone

may be in use simultaneously in more than one pitch register or on more than one keyboard. In a direct keying system like that shown, these requirements are easily met, though satisfying them usually costs a severe reduction in signal level by the time signals reach the busses.

Fig. 4-2 shows a pair of tone generators A and B feeding the grid of a tube through switches. Naturally, each generator has some source impedance, which could be represented by a resistor in series with the generator, but that impedance does not matter and changing it would not prevent the bad effects we are about to describe. The point at which both switches come together represents any one bus of a keying system like that of Fig. 4-1, and the tube may be a typical *bus amplifier,* from which tones will be fed to a group of voicing filters. You may assume that all the generators necessary to provide tones for that pitch register are connected through switches to the bus and grid and that all generators

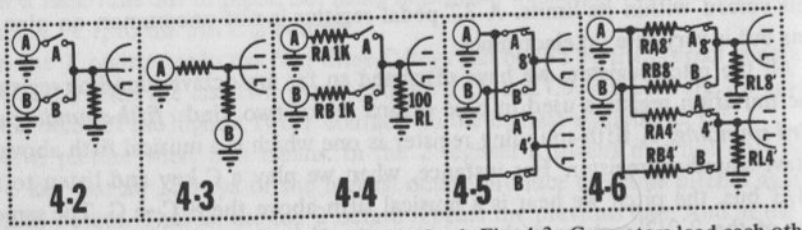


Fig. 4-2–Two generators keyed into a common load. Fig. 4-3–Generators load each other. Fig. 4-4–Load resistor RL added. Fig. 4-5–Two pitch registers keyed without isolation. Fig. 4-6–Series resistors and loads provide isolation.

have roughly equal audio output levels. You may also assume that the grid-ground resistor is much higher than any of the generator impedances. There may or may not be a d.c. blocking capacitor in series with each generator, depending on whether or not they are needed.

If only switch A is closed, tone from generator A appears at the grid. Or if only switch B is closed, tone from generator B appears at the grid. So far so good.

If *both* switches are closed, both tones appear at the grid. However, each generator has now been loaded with the impedance of the other and thus appears at the grid with reduced amplitude. This is illustrated in Fig. 4-3 by redrawing the two generators and including with each a series resistance which represents its internal impedance. We now obviously have a voltage divider, in which the signal from each generator passes through its own internal resistance as the series leg and through that of the other generator as the shunt element. The voltage output of each generator is halved at the junction of the two and the tube grid.

Worse yet, the level at the grid due to any generator depends entirely on how many generators are being keyed into this bus at one time in playing a chord or chord plus melody. Considering, for example, the level of the tone from generator A reaching the grid, it gets smaller and smaller as more and more other tones are keyed at the same time and appear between grid and ground, across generator B and its resistance, making the shunt element of the voltage divider seen by tone A smaller and smaller.

As we have said and you can see, it makes no difference what the internal impedance of each generator is; and adding an external resistor in series with each generator does not help either.

There is a simple solution, shown in Fig. 4-4. That solution is to add resistor RL, a bus load resistor, which is very much lower in value than the impedance of any generator–not more than a tenth at most. We have assigned values to make things clearer.

Now when we key generator A, its level at the grid will be only 1/11 of that at the generator output before keying, due to the voltage division. This is the price we pay. With switch A remaining closed, we now close switch B. As far as generator A tone is concerned, the 100-ohm shunt resistor of its voltage divider is now paralleled by 1,000 ohms, making the resistance from grid to ground about 91 ohms instead of 100. This is such a small change that the generator-A level at the grid is hardly affected at all.

You can see that as more and more generators are switched in, the net resistance across RL gets smaller and smaller, but it does so very slowly. Switching in a total of 11 generators–generator A plus 10 more including B–would reduce RL only to 50 ohms as seen by the generator-A signal, causing a drop in the level of generator A of only 6 db. This is still, of course, too much and RL should actually be much lower than 100 ohms in this case (or the generator impedances much higher than 1,000), but there is quite an improvement. In this case, having a total of 11 generators switched in reduces the level of each to only 1/2; while if RL did not exist the level of any generator with a total of 11 switched in would be only 1/11 of that obtained from that generator by itself.

It follows, of course, that the lower the value of RL with respect to the generator impedances, the greater will be the immunity of the system to level changes caused by playing different numbers of keys at one time.

A simple system like that discussed, without any external resistors except RL, would work provided (a) the actual operation of a generator were not affected by shunting it with low impedance or connecting other signals directly to it, (b) there were only one pitch register and one keying bus in the organ, and (c) generator impedances were low enough so the designer would not hesitate to run the signals in a cable from generators to key switches.

In actual practice these conditions are rarely met, and external resistors are placed in series with each generator. For a 1-manual, 1-bus organ, the resistor could theoretically be placed right at each generator. It would still be better to place them at the key switches, as this would give lower-impedance lines between generators and key switches in most cases. With external resistors, the theory of interaction is still the same, and all the resistors in Figs. 4-2, 4-3, and 4-4 may be regarded as external ones without any change in theory.

The values of the resistors are not chosen at random. As we have explained, the value of the bus termination resistor RL (which may simply be the input impedance of a transistor stage rather than an actual resistor) is related to the

resistances of the generators (or, now, the external series resistors), the smaller the better by comparison, for better freedom from level changes. However, RL will not be made any smaller than necessary (or the series resistors any larger) because this reduces level and requires more eventual amplification, in addition to increasing noise pickup problems. If the engineer were a mindless computer (which a few, unfortunately, are) he would determine the maximum level change possible without detection—perhaps 1 db—and assume that as many keys as exist on the keyboard might be pressed simultaneously as the worst condition. This would indeed be an alarming condition, as it would indicate a creature from another planet was playing the organ. Humans rarely have more than 10 fingers, so we can assume that 10 keys is about the maximum to be played at once and proportion the resistances accordingly. It is true enough that 1 db is about the minimum change that will probably be undetected. but since it is rare that as many as 10 keys would be pressed on a manual at a time, we can fudge a little if desirable to avoid depressing signal level beyond a certain point, and allow as much as 2 or even 3 db level change for 10 keys. Fudge factors operate as much in organ design as elsewhere, possibly more so since music and the human ear can tolerate some things that a meter would throw right back in your face. (And the reverse is true, sometimes, too—so watch out!)

So much for the relationship between RL and the generator series resistors. The values for the individual generator resistors are determined by two factors. The first is simply the minimum load a generator will tolerate. Sawtooth generators ending in capacitive voltage dividers, for example, undergo a change in waveshape when load resistance gets too small. Blocking oscillators may actually be thrown out of sync. Flipflops may stop flipflopping.

The second factor is the desirability in some designs of compensating in advance for the effects of voicing filters. Flute filters are essentially low-pass filters, and they attenuate fundamentals as well as harmonics. This means that the higher the fundamental of a tone—the higher its position on the keyboard—the lower its fundamental level will be after passing through a low-pass filter. In the final tone of the organ, this will produce a *scaling* deficiency whenever low-pass filters, which are included in the majority of voices, are used. As you play higher on a keyboard the volume decreases.

To offset this, the resistors in series with the generators can be made smaller as the frequency increases, thus preemphasizing the higher pitches. With judicious selection, the final result can be a very nice, even scaling for most of the voices. As a practical matter, the preemphasis should be less than would be required for even flute scaling; the psychoacoustic fact that higher pitches penetrate more and *seem* louder will give good aural results, while those filters that have high-pass action will not be treble-emphasized to the screeching point.

So far we have spoken of only one manual and one pitch register. An actual organ usually consists of several pitch registers and usually two or more keyboards, and this makes for an absolute necessity for external keying resistors.

In Fig. 4-5 we show generators A and B back in original condition, with no series resistors. We have not bothered to show their internal resistances, as they

do not matter. As in Fig. 4-2, we have shown them both with switches feeding a bus and the grid of a tube. We assume that this tube plate leads to the 8′ stops and the bus at its grid is the 8′ keying bus.

We also show a 4′ bus and amplifier tube, fed by the same generators, as is the usual scheme. We have assumed that at this moment the keys being pressed call on generator A to produce a tone for both 8′ and 4′ busses, while generator B is producing tone only for the 4′ bus. (Note which switches are closed.)

At the 4′ bus we will find both tones as we should. At the 8′ bus, however, we wish to find only tone A, as it is only the generator-A switch which is closed. In fact, though, we can trace the output of generator B, through its closed 4′ switch, through the 4′ bus, through the closed generator-A 4′ switch, back to generator A and through its closed 8′ switch to the 8′ bus. The result of a system like this, without external resistors, is that all tones keyed into any one bus will appear on *all* busses, an entirely unsatisfactory condition. Remember that the keys which call upon a particular generator to feed tone into the 4′ bus are physically an octave higher than those which feed tone from the same generator into the 8′ bus. Thus this situation is entirely possible—common, in fact.

Matters would be no better even if each bus had a load resistor which would obviate level changes. Obviously, it is simply not possible to connect generator outputs directly to key switches. As additional busses (pitch registers) and manuals appear, the situation gets even worse. Everything appears everywhere!

Fig. 4-6 shows the correction, which is the inclusion of a resistor in series with each switch. Let us again trace the generator-B signal to see whether it can get into the 8′ bus.

From generator B it goes through RB4′, the keying resistor in series with the 4′ generator-B key switch, through the switch to the 4′ bus. Its level is greatly reduced at the 4′ bus because of the voltage divider RB4′ and RL4′.

From the 4′ bus it goes through the closed 4′ generator-A switch and keying resistor RA4′ back to generator A. When it reaches generator A it undergoes further loss because usually the keying resistors are very much greater than the generators' internal impedances. This RA4′ is the series leg and the A generator the shunt leg of a second voltage divider. The signal from generator B thus appears at generator A after having been severely attenuated twice. Its level is now extremely low with respect to that of generator A, in practice at least 50 db lower, and usually even less. It is masked completely for all practical purposes and cannot be heard after passing through RA8′ and the associated switch to the 8′ bus.

And thus we arrive, after considerable palaver, at all the reasons for the resistors shown in series with all the switch contacts in Fig. 4-1!

Other Audio Switching Schemes

The simple system we have discussed—a resistor in series with each key switch, graded for scaling, plus a load resistor on each bus—is not the only possibility for achieving the desired results. One of the other ways of getting these re-

sults, and, in fact, an additional result, is shown in Fig. 4-7. It has been used by Baldwin and in Schober's percussion.

The diagram shows the skeleton of the switching system for one pitch register of one 5-octave manual. Full detail is shown only for the fourth octave of keys. Notice that each generator output for this octave of keys—the exact generator outputs depending on which pitch register is involved—goes to a switch contact through a 5,000-ohm series resistor. These resistors perform exactly the functions we have described for the simpler system, though only for one octave. The 12 switches contact a 1-octave collector bus when keys are pressed.

The 1-octave collector bus shown for octave 4 is duplicated for the remaining four octaves. Thus each of 60 switches (plus the 61st key for the top C) is connected by a resistor to the appropriate generator, and each group of 12 switches contacts a 1-octave collector bus.

Each octave collector bus is connected to a point in the resistor network shown. Since output from the system for one pitch register is taken from the right end, it is obvious that the tones keyed by the highest octave of keys have the highest amplitude and those at the lower end have the smallest. Thus, the purpose of the network is to give the graduated scaling necessary to offset the effect of the low-pass voicing filters. The network also, if you care to figure it out, places the same load on each of the octave collector busses.

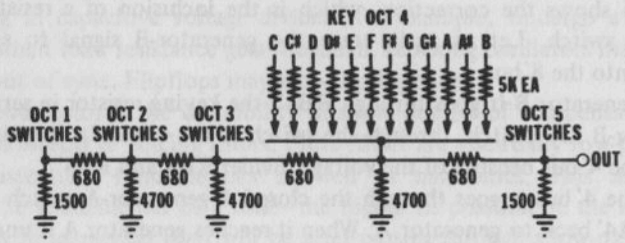


Fig. 4-7—Different isolation-resistor scheme provides preemphasis of the bass or the treble.

This system has both an advantage and a disadvantage. The disadvantage is that tone output level is changed only once per octave. Thus, to achieve the total change desired between top and bottom of the keyboard, about 15 db, in only five steps, the difference between a B and the C just above it is more audible than if smaller changes were made more often. This disadvantage can be overcome simply by dividing the system into half-octaves.

The advantage seems to me important. It is that output can be taken from the left end of the network in Fig. 4-7 as well as from the right. Output taken from the right end, as shown, has emphasized higher notes to offset the rolloff in low-pass voicing filters. Some filters, however, notably the reeds and some strings, require the opposite kind of emphasis from the keying system—uptilting of the lower notes. This can be obtained by taking output from the left. Ordinarily, an 8′ reed like a trumpet or oboe stop, which may be fine in the middle of the keyboard, becomes strident at the high end and loses all of its body at the low end.

Feeding such stops from the low end of the network offsets that. Unfortunately, the high-pass action in reed filters, which are tuned rather than rolloff types, is not so straightforward as the low-pass action in flute-type filters, and the resultant scaling can rarely be as good. Nonetheless, with careful compromise, the bass end of a reed stop can be improved by deriving its tones from a system having gradual emphasis in amplitude as tones get lower.

Though I have never seen it, a much more delightful system could easily be devised by using subcollectors, as in the scheme of Fig. 4-7, but without the resistor network shown below. First, the subcollectors should be limited to six notes. Next, each should be terminated to ground so that at each subcollector all tones on that subcollector would be at equal amplitude. Then the organ voices should be divided up into groups, all stops in each group operating ideally from a particular keyboard emphasis curve. For each group of stops, tones should be taken from all subcollectors through a mixing system which would take output from each subcollector in the proper proportion to achieve the desired emphasis curve, and the mixing system used to feed that group of stops.

Each subcollector would thus feed several mixing systems, one for each group of stops. Each group would achieve much better scaling than is now common in organs of this type, since it would have its keyed tones provided in an amplitude curve that would closely offset the scaling effects of its filters. A system of this kind would be more expensive than the simple one and would complicate the coupling system. However, organs are becoming more and more elaborate, with much attention being paid to the niceties, and it would seem such an idea could be acceptable.

Electronic Keying

For some purposes, a greater measure of control can be exercised over keying if the audio tones are not switched directly by key-operated contacts but are switched instead by tubes, transistors, or diodes which, in turn, are turned on and off by key switches. Historically, the first need for electronic switching arose when sustain circuits were introduced. When sustain is in use, you will recall, pressing a key causes tone to come on in the usual way, while releasing the key makes it die away gradually. Electronic keying can provide this effect.

Basically, electronic keying involves connecting generator outputs permanently so that they pass to the following circuits through tube, diode, or transistor gates instead of key switches. Some essential control or operating voltage of the active device is supplied or killed by the key switches either to block the active device or cause it to transmit.

There are two obvious advantages. The first is that since a d.c. potential does the controlling, the action need not be a simple on-off. Causing the d.c. control voltage to build up or decay gradually when a key is pressed or released requires only some sort of time-constant circuit. Thus any sort of attack and decay envelopes may be obtained.

The sustain idea has already been mentioned, and that is one type of special keying envelope. More basic is the desire to imitate the speech of pipes and, in

fact, most orchestral instruments and the human voice. None of these tones go on and off instantly. All of them have fairly gradual attacks and decays. The smoothness of these tone onsets and recessions helps keep the effect musical, while instant attacks and decays tend to sound somewhat artificial, at least in an acoustically dead environment which has no natural or artificial reverberation to soften the effect. Fortunately, the effect of direct instant keying is far from fatal (assuming there are no clicks and pops), for it is rather expensive to provide an active device and associated components for every note of every register.

Keying Transients

While the gradual-onset effect of pipes is a less common reason for using electronic keying, a much more potent reason in some organs is the elimination of *key clicks.* This term refers to the transient electrical noise caused by introducing an audio signal into an electronic circuit by simple switching, in which the

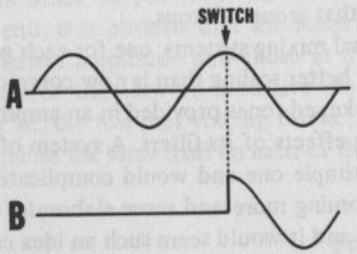


Fig. 4-8–Transient is heard when keying causes steep wavefront not inherent in waveshape of the tone.

signal appears in the loudspeaker system instantly. Several different kinds of noises can occur, depending on various factors, such as clicks, ticks, pops, thumps, and so on. The correct term, therefore, rather than "key click," is *keying transients.* An organ should produce almost* no keying transients which are audible during the playing of music, though for psychoacoustic reasons certain small transients which are audible during note-by-note, ear-to-the-speaker testing can be tolerated because they are simply not obtrusive or apparent during real playing.

The experiment to show what causes a keying transient can easily be made with any audio generator, amplifier, and switch. To avoid being misled, use a good but physically silent switch.

In A of Fig. 4-8 we assume that the output of a tone generator is the sine wave shown and that it is fed into a tube. The horizontal line is the baseline of the wave and represents the steady-state plate current.

Suppose we interpose a switch between generator and amplifier tube. At the instant we happen to close the switch, the sine wave has reached the point indicated by the arrow at A of the figure. B shows the plate current of the tube. Up

*The "almost" refers to at least one type of transient which is sometimes desirable and which one or two manufacturers spend a good deal of money to obtain deliberately–the "chiff," a fifth- or octave-sounding component of the tone which is heard in unnicked pipes at the instant of tone onset only, and which gives an interesting accent to each tone.

to the time of switching it is at its steady-state value. At the instant of switching it rises (or falls, depending on phase) instantly to the value represented by the sine wave at that instant.

Something bad has happened. While the tube may faithfully transmit the sine wave *after* it is switched in, it has created a vertical pulse of plate current at the instant of switching. This pulse is no part of the input signal; it is added by the tube as the result of closing the switch and allowing the signal suddenly to begin controlling the plate current. The pulse is just as bad as it can be; it is perfectly vertical and therefore contains an infinity of harmonics. If no spectrum modifying circuits appear from this amplifier to the speaker system, all the high-frequency components the system will pass appear at the speaker and at the listener's ear, and the result is a sharp click. The effect is the same whether tubes, transistors, or any other instantly responsive devices are used in the signal path. Even if no electronics were involved and there were a way of instantly starting some physical tone, the effect would occur in the ear, as the eardrum would be displaced from its resting position with high velocity.

With a sine wave or any other waveform this effect varies in intensity according to which part of the cycle is keyed. If the switch closed just at the instant the sine wave passed through the zero axis there would be no transient. At the 90-degree point shown, with the wave at maximum amplitude (or 180 degrees later), the transient is maximum. And for all intermediate points the transient has intermediate amplitudes. The effects are therefore unpredictable and just possibly this may be even more annoying than if they were all maximally bad!

There are three well known ways of killing or minimizing keying transients. The first is filtering. Obviously, a low-pass filter following the keying at some point will cut down on the high-frequency harmonics of the transient. Depending on other things, this may bring transients down to an acceptable level or may simply substitute a slightly less annoying thump. The method is rarely used because it also filters out desirable high-frequency harmonics of the tone and makes effective string and reed stops almost impossible. Only an organ which actually does key sine waves, such as the Hammond, could use filtering without impairing the tones themselves, since a sine wave is not supposed to contain any harmonics anyway. Hammond does indeed filter every note–but *before* the key switching, which does not help the transient. So Hammond organs are famous for (among other things) key thumps. Never add an amplifier and speaker system having good treble response above 6,000 cps to a Hammond. The racket can be astounding.

The second escape hatch is the use of electronic keying, with just enough gradualness in the application of the d.c. control potential to turn on the keying devices so slowly as to make the attacks inaudible.

The third attack on the problem, a rather sneaky one, is to use a generator waveform which, in effect, has a key click in every cycle to begin with–a sawtooth or square wave. Each of these contains at least one very vertical wavefront per cycle. If the sides of the square wave are very vertical or the sawtooth flyback is very short, the vertical wavefronts caused by keying are only a trifle

richer in harmonics than those inherent in the keyed tones. Since both sawtooth and square waves are highly useful in the production of organ voices anyway, and incidentally are very easily produced, at least in frequency-divider systems, we have one of nature's happier coincidences and all points fit together to produce virtually clickless keying and useful waveforms.

Strictly speaking, even sawtooth and square-wave keying does produce transients—frequency discontinuities. The first one is an abrupt transition between zero frequency and a finite one. The second and more important stems from the

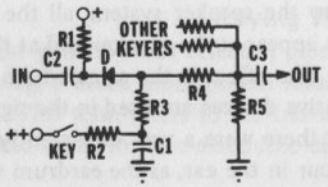


Fig. 4-9—A basic diode keyer of the series type.

fact that either of these waves, keyed at something other than the point where a vertical wavefront already exists, produces one shorter-than-normal cycle at the onset of keying. Since the period of this cycle is short, it appears to the ear as a higher frequency, which after one cycle abruptly changes to a lower one. By itself this transient is inaudible, since it is small and is masked by the main tone and its high harmonic content. However, it can sometimes be detected in flute tones which are obtained by filtering sawtooth or square waves, since the low-pass filter can cut down harmonics but cannot affect repetition rate. In such cases, the higher harmonics of the tone have been filtered out, and along with them much of the sound that masks the frequency discontinuity. Fortunately, when the sawtooth or square waves have really good, vertical sides, even this transient is so small it has to be discovered by careful testing.

Typical Electronic Keyers

In very modern organs, those of the expensive, elaborate breed which has grown up among several manufacturers whose main business has always been smaller instruments, electronic keying is actually used to avoid complications. Where you have several sets of tone generators and a good many pitch registers, a physical keying system with contacts and wiring to handle every tone individually would be a 25th century monster. A playing key can be asked to do only so much before its mechanism, and even the pressure on it, become too great. In elaborate systems there must, of course, be an elaborate set of electronic keyers —but all keyers which are supposed to be turned on when one key is pressed can be handled with a single key contact which transmits the d.c. control voltage simultaneously to all of them. The same keyers can be operated also by other keys when that is desirable, or even grouped differently for the purposes of other keys, by running more control leads from the additional keys (still with the one contact) and placing a cheap diode in each lead so that voltage applied from one source cannot back up in the interconnections and cause chaos. The

chapters on individual organs show some of these fancy schemes. In this chapter we will show only enough actual circuitry to demonstrate the principle. The variety of keyer circuits is great, and you will find different kinds described along with the organs they go into.

A simple type of series diode keyer is shown in Fig. 4-9. Input from the tone generator is at the IN point. A positive potential is connected through R1 to the cathode of the diode, which may be germanium or silicon, and this blocks it so that it does not conduct.

When the key is closed, a higher positive voltage (designated by the two plus signs) goes through R2 and charges C1. R2-C1 is a time-constant circuit deliberately inserted to prevent instant application of on bias to the diode and thus provide a keying delay. The (relatively) slowly rising positive d.c. is applied through R3 to the anode of the diode. This overcomes the positive off bias and causes the diode to conduct, so that audio tone passes through it and isolating resistor R4 to load resistor R5. C2 and C3 are just d.c. blocking capacitors.

The OUT point may be likened to one of the pitch register busses in Fig. 4-1. An additional diode for each key exists to handle the remaining tones for that pitch register, and, as shown, they all join signals at the right end of their R4's. R5 has a relatively small value with respect to R4, so that positive keying voltage is very small by the time it gets there and will not turn on the keyers for any other pitches by backing up through their R4's.

If the organ has several pitch registers for the manual concerned, several diode keyers may be connected to one R2-C1 combination; all the R3's, one for each pitch register, connected to the junction of the single R2-C1 provided for the key.

This is about as simple as a series diode keyer can be. It is not very effective when the organ output must contain the harmonic range needed to achieve good reed and string tones. There is leakage through the diode even when it is blocked —mostly higher harmonics. If these are not filtered out, either by electronic components or by failure to provide a speaker that will reproduce highs, before reaching the listener's ear, there will be a constant high-pitched whine. The circuit is usable only for flute tones where there is little harmonic development or for inexpensive instruments without much tonal variety.

A shunt diode keyer is diagrammed in Fig. 4-10. C1 and C2 are simply d.c. blocking capacitors. The signal goes through R4 and R5 to the output. The diode at the junction of R4R5 is normally biased on by negative voltage applied through R1, R2, and R3. It conducts, therefore, and shorts the signal to ground.

When the key is closed, the positive d.c. control potential is applied to the R1-R2 junction. Being higher than the negative source, the net charge on C3 changes (gradually because of R2, through which its charging current must flow) to positive. This positive is applied to the cathode of the diode, turning it off and removing the ground shunt from the signal path. When the key is opened again, the negative voltage once again takes over and the diode again conducts and shorts out the signal.

This system can also be operated on the basic scheme of Fig. 4-1. Additional

keyer outputs for the same pitch register can be connected to the OUT point through their own R5's. Additional keyers for the different pitch registers of the same key can be operated by connecting their R3's to the same R1-R2-C3 combination.

This shunt diode system is potentially somewhat more effective in killing tone with the switch open than the series diode scheme, but because there must

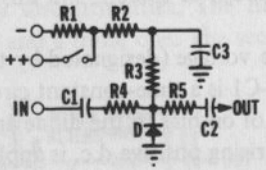


Fig. 4-10–A diode keyer of the shunt variety.

always be resistors high enough to avoid damaging signal transmission, there is difficulty in getting enough current through the diode to make it conduct heavily enough in the signal-off condition. To reduce leakage, two or three stages can be used, as you will find in the Gulbransen chapter.

A basic problem with diode keyers is that the diode does not in any sense amplify. To make it conduct, you must pass a reasonable amount of current through it. When isolating resistors must be used in series with the d.c. voltage, passing enough current through the diode is difficult.

Using a transistor as a keyer is easier because, while no element of the transistor has the practically infinite impedance of a tube grid, impedances are many times that of a conducting diode. It is thus much easier to make desired turn-on and turn-off voltages appear at transistor elements without using supplies of astronomical values.

Fig. 4-11 shows a simple transistor keyer which is basically an emitter follower. With the key up, the base is at ground potential and the transistor is almost at cutoff, which is normal. To cut it off all the way, a negative bias is applied to the emitter through the emitter resistor R4. This is the same as applying positive bias to the base, but eliminates a double power supply. C1 and C2 are just d.c. blocking capacitors.

When the key is closed, negative voltage (much greater than the negative applied to the emitter, as indicated by two minus signs) is applied through R1 to C3, which gradually charges. The negative charge is transmitted through R2 to the transistor base and it is large enough to overcome the emitter bias and turn the transistor on.

The beauty of using a transistor, even in a simple circuit like this, is twofold. First, when the transistor is cut off, leakage through the internal base-emitter diode is much less than with a simple diode, since the diode impedance is greater and R4 can be relatively small. Second, the current required to bias the base on is very small and large resistors can be used for R1, R2, and R3 without in any way sacrificing results. In practice, the correct voltage between base and emitter of an emitter follower is about half the supply voltage. R1, R2, and R3, therefore, are simply proportioned to get this value at the base when the switch is

closed, and the same B– used to energize all transistors in the organ can be used as keying voltage.

To consider the OUT point as a pitch register bus, all emitters for that pitch register can be connected to it, each through an isolating resistor at least large enough to prevent audio shunting effects on the emitter follower. For notes of different pitch registers operated by the same key, other transistors can be connected, the R2 of each going to the one R1-C3 combination for the key.

The two best electronic keyer circuits I know both use transistors, and I call them the best because when shut off they do not leak and the keying envelope is easy to control. The first is an emitter-follower circuit like Fig. 4-11, but with the bottom of R4 connected to ground and the small negative off bias applied through a diode with its anode on the emitter (assuming a PNP transistor). This circuit is used by Schober; see Figs. 10-9 and 10-20. In the absence of base bias provided by the keying circuit, the fixed bias applied to the diode causes the diode to conduct so that off bias is applied to the emitter. At the same time the conducting diode constitutes a very low resistance between the emitter, which is the audio output point, and ground (through the power supply), thus shorting out any signal that leaks from base to emitter. A value of this keyer is that the audio input has a relatively high impedance–if that matters.

The other good keyer is Conn's (Chapter 19), diagrammed in outline in Fig. 4-12. It is essentially a grounded-base amplifier stage (the dual of a tube-type grounded-grid stage), with audio input to the emitter, and output taken from the collector. The stage has gain, just as the more usual configuration, the only disadvantage being that the input impedance at the emitter is always around 600 ohms, which may be rather low for some purposes. However, since there is gain, a series resistor can be added between input terminal and emitter to present a higher total impedance to the source and still, in most cases, provide enough output. This keyer, too, leaks virtually not at all when off. Its advantage is that the base becomes a single-purpose element which need handle only the keying

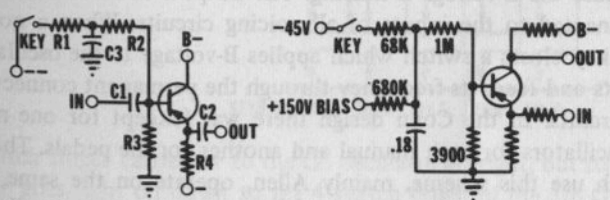


Fig. 4-11 (left) – Basic emitter-follower transistor keyer. Fig. 4-12 (right) – Grounded-base amplifier makes a good keyer.

voltages and need not be complicated by signal circuits. Here a permanent high positive off bias is applied through large resistances, and this is overcome by the –45-volt keying potential when the stage is to be turned on, both circuits acting with the .18-μf capacitor to give a gradual buildup and decay.

A normal common-emitter stage does not make a satisfactory keyer. It leaks like a sieve.

One peculiarity of all diode and transistor keyers is that if they incorporate any delay going on or off or both, the only kind of waveform they can handle without distortion is a square wave. If a diode or transistor is biased on and the bias is slowly reduced, the amplitude of the transmitted signal is reduced by a clipping action, not in the manner of an increasing interposed resistance. And obviously, the only kind of waveform which can be clipped and still retain its original shape is a square wave. In actual practice, some organs do use these keyers with sawtooth waveforms, with results that are sometimes detectable but not usually objectionable. A good sawtooth contains so many high harmonics that altering them for a short time during a buildup or decay does not seem to cause anyone any heartburn. What actually happens, as you can see by drawing a sawtooth and clipping it with pencil lines, is that the waveform becomes more pulse-like. This means that the lower harmonics are actually attenuated during the clipping. However, as the effect gets worse the amplitude is constantly decreasing, which mitigates matters. I would not really want to do all this with a waveform not having pretty high harmonic content; and of course you cannot have more than one tone at a time passing through such a gate, as intermodulation would make life quite impossible (unless the attack and decay times were so short you did not have time to hear the IM).

Oscillator Keying

Instead of having generators which run all the time, the outputs of which are switched to the desired points by the keys, some organs have generators which are normally turned off and are turned on by a key switch only when wanted. One of the major manufacturers, Conn, used to use this system but has discontinued it because of its rather strong disadvantages in anything but very large instruments having many generator ranks.

The scheme is easy to understand. There is a rank of generators which includes an oscillator for each desired pitch. All of them are normally inoperative, usually because the B-voltage is missing. The outputs of all oscillators are permanently connected to the inputs of all voicing circuits. When a note is to be played, the key closes a switch which applies B-voltage to the oscillator, whereupon it starts and feeds its frequency through the permanent connections to the following circuits. In the Conn design there was (except for one model year) a rank of oscillators for each manual and another for the pedals. The remaining organs which use this scheme, mainly Allen, operate on the same basis as we shall describe but the voicing circuits are largely built into each oscillator.

The major disadvantage of the idea is that it is not possible to have each stop speak in its own pitch register as long as there is only one rank of oscillators per manual. Different pitch registers may be obtained, since one can cause the switch of a different key to turn on a given oscillator. But since all oscillators of a given rank are permanently connected to a single group of voicing circuits, any oscillators of the group which are turned on, no matter by what keys, will be heard with all voices that have been selected.

Fig. 4-13 shows the basic philosophy, using only the G generators and keys of

one manual. The generators G1 through G7 are shown at the top. We have assumed only three pitch registers, though in the usual case more are used. For each register there is a transverse bus and a contact for each key. The lettering at the bottom shows which key the three switches above are for. All three switches, of course, make contact to their respective busses simultaneously when a key is pressed.

At the right of each bus we show a switch leading to the B+ (we are assuming the generators are tubes). These switches are called couplers, though they do not quite conform to the scheme of pipe-organ couplers. They might better be named "pitch register selectors," since they are the only means of determining pitch registers. The lines coming out of the tops of the generators are the audio leads, which mix and go to voicing circuits.

If we want to hear the 8′ pitch register we close the 8′ switch at the right and then play the keys. With this switch closed, only the contacts of the 8′ bus are effective in energizing oscillators. With the connections as shown, pressing the third G of the 5-octave manual energizes oscillator G4, which, from Table 1-1, has a frequency of 392 cps, middle G.

If, instead of the 8′ switch we close the 4′ switch, then the lowest bus carries B+. When we press the third G key oscillator G5 is energized and we hear tone an octave higher. And so on.

While Fig. 4-13 introduces the idea, it contains a fallacy. Suppose we want only 16′ and 8′ pitches, so we close only the 16′ and 8′ switches, which apply B+ to those two busses. Now we press, say, the 4th G key. The voltage from its mid-

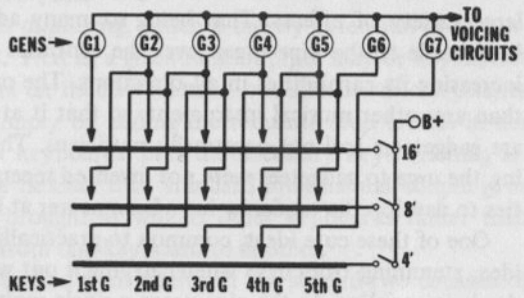


Fig. 4-13—Basic way of obtaining pitch registers when keying oscillators.

dle (8′) switch finger carries B+ to the G5 oscillator all right, but from the oscillator the B+ goes right on down to the 4′ switch finger of the 3rd G key. If now that 3rd G key is also pressed, B+ is placed on the 4′ bus, just as though the 4′ switch at the right had been closed, a most confusing and undesirable situation.

One solution is the most common one. B+ is applied to all the busses permanently, but half the circumference of each bus is covered with an insulating shell, and the busses can be rotated axially. To turn a pitch register off, the bus is rotated so that the switch fingers hit the insulating material, not the metal. Nothing can then cause voltage from one finger to pass to another.

The second solution is to operate exactly as in Fig. 4-13 but to add a diode in series between each switch finger and the oscillator to which it is connected, the

anode connected to the switch finger. Positive potential will then be applied from any energized bus to an oscillator, but trying to get back down through another finger to energize a different bus, it is blocked by the diode.

As you can see, all tones are heard through the same voicing circuits, even though the tones sent to those voicing circuits may be in any pitch registers selected by the switches. The point is that *all* selected voices sound in *all* selected pitch registers. This is a major reduction in versatility of the organ because it means that you cannot, for instance, mix a 4′ flute and an 8′ oboe. If you want both stops and both pitches, both stops will sound at both pitches, which is quite a different matter.

Where an organ is large and expensive enough to have several ranks of generators, the disadvantage of oscillator keying is mitigated or disappears entirely. There can be, for example, an 8′ rank which is keyed only by the 8′ bus; the stops it feeds will then sound only at 8′ pitch—and so on for other registers. All sorts of possibilities exist for even the clumsiest scheme when you can invest enough money in multiplicity of facilities. The apogee is reached, of course, when you have a rank of generators for every voice, which is just what the pipe organ has. As far as I am concerned, when you get to that point of extravagance, you have eliminated the advantages of electronics and you can afford a pipe organ which, if well designed and executed and properly maintained, is still the artistic superior of any electronic.

Coupling

One of the chief values of an organ is that a single player can obtain from it a large variety of effects. That being so, many additions and modifications have been made to the pipe organ over the centuries by people with bright ideas for increasing its capabilities in all directions. The organ is much more of a machine than any other musical instrument, so that it attracts the interest of those who are gadgeteers and not necessarily musicians. The chromosomes, if any, governing the urge to gadgeteer were not invented recently, even though the opportunities to develop the tendency have been nearer at hand in this century.

One of these cute ideas, common to practically all pipe organs—and a very old idea, stemming from days when carrying it out was mechanically very clumsy indeed—is coupling. In the pipe organ a single rank of pipes can give only one tone color, which depends on how the pipes are made. Each of the several ranks provided for the purpose of making additional tone colors available sounds on just one keyboard when a particular stop tablet is turned on, since if it always played from all keyboards there would be no point in having more than one keyboard. The basic tonal equipment of an organ is, therefore, its ranks of pipes, and those are what cost all the money. The console equipment, wind chest, etc., cost money, too, but when a designer wants to add to the organ's "specification"—make it bigger and (hopefully) better—it is the added pipe ranks that eat up the shekels and not the extra stop tablets and console parts.

Coupling is a way of squeezing more out of a given number of pipes without incurring substantial cost. *Intermanual* coupling makes pipes normally assigned

to one keyboard or "division" also available on the other(s) so that a larger number of different ensembles of stops can be made up than the stops of one keyboard alone would allow. *Intramanual* coupling is a way of keying a rank of pipes which normally sounds in one pitch register in two or more registers. See Chapter 2 for the details.

Two other terms are used in connection with certain kinds of coupling. *Unification* is the use of a single rank of pipes of each tone color, with regular stop tablets—not couplers—arranged to key each rank at different pitches. One difference between unification and intramanual coupling is that in unification the additional stop tablets usually purport to be different stops if you look at the engraving on them. Of two tablets which key a given flute rank, for example, one may read OPEN FLUTE 8′ and another PICCOLO 2′; but they both key the same pipes in different pitch registers. It is easy enough to figure out how the key switching can be arranged to carry that out. Or another pair keying a rank of reed pipes may read TRUMPET 8′ and TROMBONE 16′. In the first case the pipe rank is called a "unit flute," and in the second a "unit trumpet" or some such name.

A unit system like this is more flexible than intramanual couplers—though one should keep in mind that it *is* effectively a coupler system and not a satisfactory substitute for differently voiced ranks for (using an example above) a trumpet and a trombone. With standard couplers, pressing a Swell To Swell 16′ tablet causes *all* ranks of the manual to sound an octave lower than normal (in addition to normal pitch). With the unit plan, certain ranks may be keyed as 8′ stops while others are keyed as 16′, etc.

The second additional term is *duplexing,* which is merely extension of the unification idea between manuals. That is, a given rank of pipes may be keyed, not only at different pitch registers on its own manual, but also at various pitches from other keyboards, again simply by adding the requisite stop tablets in the groups assigned to those other keyboards, plus the necessary key-switching arrangements. Again, this is more flexible than standard intermanual coupling, as single selected ranks may be brought at will to other keyboards rather than wholesale transfer of *all* voices from one keyboard to another.

Unification and duplexing reached their zenith in the second two decades of this century in theatre organs built for the great and not so great silent movie palaces. These instruments were required to have the most enormous versatility adequately to enhance the moods and action of silent pictures. But neither funds nor space were available in great quantities; these organs were, after all, bought by commercial enterprises and did not have the backing which has enabled organized religion to acquire the much grander church instruments. Cinema organs, therefore, had to give the most results for the least money. Thirty ranks was almost never exceeded in even the most grandiose theatre, and this is a peanut whistle compared to some of the gigantic church organs. But the last ounce of performance was squeezed out of those 30 ranks (usually far fewer) by unifying and duplexing every rank within an inch of its life, so that the great horseshoe consoles bristled with stop tablets. Just one rank made available at the

eight pitch registers shown in Fig. 4-1 (and this is not the limit) would account for eight tabs for just one keyboard. Assuming three manuals, this makes 24, plus maybe two or three registers in the pedal, and you can easily have a total of perhaps 27 tabs for just one rank of pipes. Multiply this by anything from 8 to 30, and you can see why the curve of the console was necessary to enable the organist to reach all the tabs without growing arms twice as long as usual.

An organ which does not include unification and duplexing, but only normal coupling, is known as a *straight* organ. Most church organs are built this way, though in some larger ones there are one or more "floating" ranks keyable from two or more keyboards.

Many people concerned with the church organ look down their noses at unification and duplexing and abhor anything but straight design. I can agree with them if unification and duplexing are used as an alternative to furnishing a sufficient number of pipe ranks (or, in electronic organs of certain designs, enough generator ranks), as was usually the case in the cinema theatres. But as a matter of ideology I find the objection silly. Given a good organ, with good voices or ranks and enough of them, nothing but good can come of making them available at more than one pitch and on more than one keyboard, though disguising the additional tablets with different names seems rather underhanded. In most electronic organs, of course, additional voices do not cost additional ranks of generators. In those using formant filters it is very nearly as cheap (and musically more effective) to provide extra filters and thus extra voices as it would be to unify and duplex by switching the tones through the same filters at different pitches and from different keyboards to get the same effect as unification and duplexing.

Coupling Circuitry

Coupling in an organ whose oscillators are normally silent like that for which a keying system is shown in Fig. 4-13 is easy to understand and straightforward, as the diagram already indicates. You simply provide a keying bus for each coupler, whether intermanual or intramanual. When that bus is energized its contacts turn on the oscillators, either those of that manual or those of another keyboard to which the contacts are connected, in whatever pitch register you have wired the contacts for.

Coupling in an organ having electronic keyers can be exactly the same, since again the keying system merely supplies turn-on voltages to selected keyers.

When audio is being keyed directly things become a bit trickier. Referring to the scheme shown in Fig. 4-1, the output of each bus goes to one set of stop filters of one pitch register. To couple, we must send the outputs of other keying busses to the same stop filters, without allowing tones from the first bus to back up and sound through the stops of the second. One-way audio coupling is required, in other words. A complication is that the stops must speak with the same volume when the coupled tones are passed through them as when their own keyboard tones appear, so losses, if any, must be the same for both.

There are so many ways of accomplishing this that it would be futile to try to

be thorough in this chapter. Let us just show one simple but effective method so you will get the point.

Assume we wish to have a Swell To Great 8′ coupler. This means "Swell stops to Great manual." With this coupler on, whenever we play the Great manual keys, we will hear any Great stops we have pulled,* and in addition we will hear any Swell stop which is turned on. The 8′ means that all Swell stops selected will sound from the Great keys in the same pitch registers engraved on the stop tablets.

In the electronic organ what is actually happening is that the tones switched to the Great keying busses are going to their own stop filters as usual and *also* to the same stop filters which are normally fed tone by the *Swell* keying busses. Don't worry if this whole thing plants some wrinkles on your brow. Even the oldest hands have to concentrate and think twice to keep this coupling business straight. (At least I'd like to think so because I'm in that boat myself.)

Let us assume that the only stop pulled on the entire organ is the Swell Trumpet 8′. With our coupler on we should expect to hear this trumpet (always at 8′ pitch) when we play the Swell keys, as usual, and also when we play similar

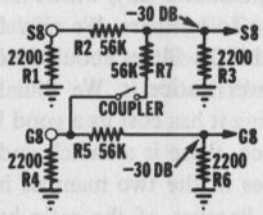


Fig. 4-14—Diagram shows basic philosophy of coupling.

Great keys. The sounds from both keyboards should be identical in tone color, pitch register, and volume. How do we get tone from the Great 8′ keying bus into the Swell 8′ stop filters—without changing or interfering with any of the normal feeds?

One way is diagrammed in Fig. 4-14. Please ignore R7 and the coupler switch for the moment. The circles designated S8 and G8 are the keying busses for the Swell 8′ and Great 8′ tones, as per Fig. 4-1. Each is terminated by a 2200-ohm resistor and fed through a 56,000-ohm resistor to a 2200-ohm load, which may actually be the input impedance of a transistor bus amplifier. Due to the voltage-divider effect of R2-R3 or R5-R6, there is a loss of roughly 30 db between each keying bus and the input to its filter system.

Our coupler calls for us to feed tone from the G8 bus into the filters normally supplied tone by the S8 bus. However, reverse coupling must not occur. That is, tone from S8 must not feed into the G8 filters. Let us now open our eyes to the coupler switch and R7.

When the switch closes G8 tone indeed goes to the S8 filters. It goes there through exactly the same kind of network involved in transmission of tone from

*"Pulling" stops is the common term for turning them on, derived from the old drawknob type of stop control, which actually does have to be pulled out.

each keying bus to its own filters—a 56,000-ohm resistor, R7 in this case. And switching in the extra 56,000-ohm resistor does not affect anything in the S8 circuits, for, as seen by the S8 tone it is just another resistance to ground across R3, which is too small to be affected by it. It does not affect anything in the G8 circuit, since G8 tone sees it as a resistor across R4, but again one so large as to have no effect. And since R7 has the same value as the regular transmission resistors R2 and R5, the G8 tone transmitted through it to the S8 filters gets there at the same volume level as the regular S8 tones.

But what about back-coupling of S8 tones into the G8 filters? Notice that we have marked each filter input "–30 db," meaning that the tone at that point in each case is 30 db lower than at the keying bus from which it comes. Tracing the back-coupling path, we note that at the point in the S8 network to which R7 is connected, the S8 tone is at –30 db. That S8 tone passes through a voltage divider consisting of R7 and R4, which knocks another 30 db off it as measured at the G8 keying bus. To reach the G8 filters it must then pass through voltage divider R5-R6, along with the G8 tones, which knocks it down another 30 db. S8 tone therefore reaches the G8 filters at a level of –90 db (referred to the level at each keying bus as 0 db), which is 60 db lower than the G8 tone which is and has the right to be there. We may therefore say that the back-coupled S8 tone appears at the G8 filters about 60 db down, which is enough to insure that nobody will ever notice it. We thus have thoroughly correct coupler action, although getting it has cost us a good bit of attenuation.

In practice, there is a switch and an R7 for each of the corresponding pair of keying busses of the two manuals involved in the coupling, and all the switches are ganged. Because of the ratio between the transmission resistor R7 and the load resistor R3, quite a number of couplers can be used even when several of them require transmission of tones into the same R3 or R6. The limit only begins to be reached when the net parallel value of all the R7's feeding a particular R3 or R6 starts to be low enough to reduce the net resistance across the R3 or R6 to the point where volume level from the circuit's own keying-bus source drops audibly. With the values shown, a minimum of three couplings could be made to any R3 or R6. If more were desired, all the R3's or R6's could be made smaller or the R2's, R5's, and R7's larger.

Intramanual couplers are, of course, handled in the same way. Instead of introducing tone from a keying bus of another keyboard into the filters, tones from another bus of the *same* keyboard are passed through a switch and an R7, so that the filters are fed tones from the same keyboard but in another pitch register.

A Unison Off coupler is different from any other intramanual coupler only in minor detail. (Refer back to page 25 if you don't recall what it is musically.) In the organ of which Fig. 4-14 is a small part there might be a Swell Unison Off. In that case, there would simply be a switch between the S8 keying bus and the left end of R2—and the switch would be normally closed instead of normally open as is true of other couplers. Turning the tablet to the on position would *open* the switch, depriving the S8 filters of S8 tone. There would be a similar

switch, of course, between each of the Swell keying busses and the 56,000-ohm resistor leading to the stop filters of corresponding pitch, and all switches would be ganged.

As I have said, keeping coupling straight in one's mind is not easy. Like other mnemonic chores of certain kinds, they are better understood in the first place so you can reconstruct the process when necessary in future rather than trying to recall something once memorized. To help you understand thus thoroughly we shall end this chapter with a dual-purpose list of couplers that can exist, the first purpose being to show how wide the variety can be. The second reason for the list is to give you a sort of exercise in which, for each coupler listed, you determine for yourself exactly (a) what the playing result is in terms of what stops sound from what keyboard and what keys, and (b) what goes on inside the organ in terms of which keying busses feed tones to which filter-system inputs. You may actually never see some of these couplers—but then again, who knows?

Great To Swell 8′	Swell To Great 2′	Great To Pedal 2′
Great To Swell 16′	Swell To Pedal 8′	Great To Great 16′
Great To Swell 4′	Swell To Pedal 16′	Great To Great 4′
Great To Swell 2′	Swell To Pedal 4′	Great Unison Off
Swell To Great 8′	Great To Pedal 8′	Swell To Swell 4′
Swell To Great 16′	Great To Pedal 16′	Swell To Swell 16′
Swell To Great 4′	Great To Pedal 4′	Swell Unison Off

Chapter 5

Tone Coloring

The most important characteristic of any musical instrument is its timbre or tone color. Differences in tone colors are really the most important reason for the existence of the wide variety of instruments available to the composer and performer, although, especially for ensemble purposes, differences in pitch registers run a close second. Music is, for the most part, a series of intrinsically meaningless aural impressions, of which the most vital aspects are variety and contrast. Just as music becomes monotonous and uninteresting when a single rhythmic pattern is endlessly repeated, or everything is played with the same loudness, or a melodic or harmonic device is used *ad nauseam,* one would eventually become bored with music if all of it were heard with the same tone color.

It is, of course, true that one often does hear music in large segments played on a single instrument such as the piano. But even the piano does have several capabilities for variety in loudness of each individual note and it actually changes tone color as a tone is prolonged. Instruments like the piano and harpsichord came into being largely because of convenience. Several other people are not always available when one wants to make music, so a single instrument which can produce melody and harmony under the hands of a single individual is very useful, and these instruments are really compromises between musical desirability and convenience. Their lack of tone-color variety severely limits their range of musical expression. While many people derive immense enjoyment from listening to a whole evening of piano music or from playing, themselves, for several hours a day (and I am one of them), or from participating in long sessions of other music with relatively invariant tone colors, like solo instrument recitals and string quartets (count me in here, too), a person who would be satisfied to go through life hearing music expressed through only one instrument or small ensemble is very rare. Even the music of Bach, which in my opinion is so purely self contained that it is practically independent of the instruments which express it, was written for a wide variety of instruments and groups and is more enjoyable, on the whole, than when it is heard in, say, piano reductions.

The pipe organ is unique among musical instruments in that, while it is a keyboard medium like others, with the convenience of playability by a single musician, it is capable of having a great many different tone colors. In a sense it is an amalgam of the symphony orchestra, with its great variety of sound qualities, and the piano or harpsichord with its amenity to a single musical will. All of this makes it intrinsically the most musically interesting solo instrument.

Unfortunately, this great capability is at the same time its musical downfall.

To achieve its effects, the organ becomes a large machine and the machinery tends to come between the musician and his audience, so that the musician at the organ loses personal, tactile intimacy with his sound production and sacrifices much of the capability of the ordinary instrumentalist to express subtle nuances. It is for that reason (among others) that the organ has never really taken a significant place alongside the more conventional instruments and ensembles in popularity among the listening public.

It is nonetheless still true that the organ, pipe and electronic, has its reason for being in its variety of voices or tone colors, and that this variety alone (in addition to the fact that it makes a big noise in church) justifies its existence. Tonal variety becomes all the more important in today's world where many times as many organs are sold for the amusement of householders as for use in church. By and large, the home organist is not a master of music, and the wider and pleasanter the array of voices on his organ, the bigger kick he will get from it. While the accomplished organist is not so dependent on voicing variety as the amateur, his experience tells him more of what to expect in voicing, so, for a slightly different reason, he, too, is robbed when the organ he plays has insufficient tone color variety.

All this leads back to a paraphrase of the first sentence of this chapter: The most important characteristic of an organ is its voicing—the number of different tone colors available, the differences between voices, and how they combine. Unfortunately, those new to the organ do not always realize this simple fact, and are often led to purchase organs in which all the design money has been spent for monkey tricks, the straight voicing being skimpy and monotonous. I cannot lay too much stress on the necessity for varied voicing in the regular organ stops. It is not really essential that the voices resemble those of the pipe organ because today's home organ is rarely used for pipe organ music and thus has become a sort of *different* instrument. But, stated in simple but correct terms, an organ with a good number of voices, each different from the others, the selection varying from smooth to blaring, biting sounds with as many stops as possible in between, is probably a good organ. One with only a few voices or whose voices sound pretty much the same is nearly worthless, regardless of how many glides, slides, repercussions, and cowbells it may have and that it may shoot out an American flag when you play "The Star-Spangled Banner." Organ manufacturers know all this as well as I do, but some (as in every field) design their products to make money rather than music. It is quite possible to do both.

Harmonic Synthesis

The simplest possible view of the mechanism of tone coloring is that the timbre of a tone is determined by its harmonic structure—that is, how many harmonics it contains and their relative amplitudes. This is true—for a single tone of specified pitch. Indeed, a waveform can be broken down by Fourrier analysis, or physically by a harmonic analyzer, into its component harmonics; and if a new waveform is then constructed with the same harmonic components in the

same proportions, no matter what the nature of the generator(s) supplying them, the sound of the tone will be the same.

It can also be assumed (incorrectly) that a given musical instrument or organ stop has a fixed harmonic structure which is the same for all notes of the scale, and that if all the notes are produced with that fixed harmonic structure or waveshape the aural effect will be that of the complete pitch range of the tone color in question. This assumption is the basis of the Hammond organ. The organ contains almost 100 sine-wave tone generators corresponding to over seven octaves of normal musical pitches. The controls are so arranged that they can be set to give any one of a large possible number of fixed harmonic structures to all notes. For a particular setting, as an example, if relative fundamental amplitude is 8 on a scale of 8, second harmonic might be 7, third harmonic 4, fourth harmonic 2, fifth harmonic 6, and so on. This structure will then hold true for all notes played (except perhaps for the highest ones, for which the generating system runs out of generators for some of the upper harmonics). The "harmonics" are all taken from generators which are part of the tempered scale, so that only the second, fourth, and eighth are really harmonics, the rest being approximate—like borrowed mutation pitches in a more conventional organ (page 12). However, inaccuracy makes no real difference in our discussion and it is not responsible for the "Hammond sound."

That this system lacks something is apparent to anyone who has ever heard a Hammond organ, for quite obviously none of the music sounds like music played on any other instrument. It is not necessarily unpleasant—this depends on whether one is or is not a Hammond *aficionado*—but it gives a very strong sense of artificiality. It is also, to a degree, monotonous, in part because there has to be a finite number of generators to furnish the harmonic components and economy dictates that there be no generators yielding tones over about 6,000 cps, so that all the sound is relatively bland.

The Hammond system is known as *harmonic synthesis* because it synthesizes a tone from harmonic components, just as a complex tone can be synthesized graphically and mathematically by building it from harmonics in a Fourrier series.

Spectrum Shaping

The fallacy of the harmonic synthesis system, if one is trying to eliminate the feeling of artificiality or even to imitate the musical effects of pipe organs or other instruments, is that no "normal" musical instrument (and no other soundmaker covering a range of pitches) has an identical harmonic structure for all pitches. The very simple reason is that there is no known physical method by which any such result could be obtained, assuming anyone wanted to do it. It is only the enormous flexibility of electrical and electronic technology that makes this and other "unnatural" things possible. Fortunately, electronics also makes it possible to synthesize "natural" phenomena.

The characteristic tone color of any conventional instrument is actually determined by a spectral response curve showing, in effect, the frequency response

for tonal transmission through the instrument. In a trumpet, for example, the mouthpiece and the player's lips constitute a tone generator, and indeed a trumpeter can play tunes just on the mouthpiece, without having it attached to a trumpet.

The mechanics of playing with the lips is simple in theory, though the performance of it is not to be learned in a day. The player holds his lips together with muscular pressure and then blows. The pressure of air opens the lips slightly and as the air pressure is continued they open more and more. After the lips have opened a certain amount the velocity of the air stream is high but its pressure is not great enough to keep the lips open, so the muscular effort causes them to snap shut, after which the cycle begins again. Stated another way, the high velocity of the air stream through the open lips causes a reduction in pressure according to Bernoulli's theorem, just as air flowing over the convex upper surface of an aircraft wing lowers the pressure on the upper surface, causing the maintained pressure on the undersurface to lift the wing. In this case the pressure of air which would be needed to keep the lips open is reduced below the outside pressure tending to close them, so that the latter, in conjunction with the muscular control of the player, snaps the lips shut.

This gradual opening and snapping shut causes an air stream which gradually increases, then suddenly shuts off, giving an acoustical sawtooth wave. By skillful control of muscular effort and air expulsion, the player can cause this wave to repeat at audio rates over a fairly wide range of frequencies, so that he can actually play tunes.

This is exactly what happens when the mouthpiece is connected to the trumpet. However, the trumpet itself is a resonator, a length of tubing, folded up for convenience, which, like any pipe, has a certain fundamental resonant frequency. When the mouthpiece is coupled to the instrument, only those "lip tones" corresponding to the instrument resonant frequency—and its harmonics, to a lesser degree—are transmitted through the piping and the end bell to the outside air. (To give the instrument the capability of playing the scale, three supplementary lengths of tubing can be inserted by pressing valves, and by combining these in different ways, eight different resonant frequencies are available.) It is thus the total length of tubing in use which determines, along with the selection of lip-tone repetition rates, the fundamental pitches of trumpet tones.

However, the body of the instrument, along with the shape of the bore and the shape and size of the end bell (which is the coupling element between the instrument and the outer air), has a function over and above determining pitch. These items constitute an acoustical filter, the response of which can be plotted like that of an amplifier or speaker system. A characteristic of the spectrum response of a trumpet is that frequencies at and around 1,000 cps are transmitted more efficiently than others to the outside air. This means that, for example, C-1047 (Table 1-1) has an emphasized fundamental, C-523 has an emphasized second harmonic, C-262 has an emphasized fourth harmonic, and so on. None of this spectrum shaping has anything to do with the player's lips, but only with the construction of the instrument. If we replaced the trumpeter by holding a

small loudspeaker against the mouthpiece, energizing it with sawtooth signals at the right pitches, we would get the same results and they would still be trumpet sounds.

The action in a trombone is qualitatively the same, except that the variation in resonance is achieved with a sliding section of tubing rather than with valve "switching." The trombone tubing adds up to about 9 feet, compared to the trumpet's approximately 6, so, as you would expect, it has a lower fundamental frequency range. In addition, however, its tubing is of larger diameter and the bell is bigger, and the part of the spectrum over which its response is best centers on about 500 cps.

Formants

In the same way, each of the brass instruments has its own particular range of frequencies over which response is at maximum. This range is called the *formant* of the instrument, and it is the main determinant of the over-all tone color of the instrument. Fig. 5-1 shows the formants of a few common orchestral instruments. Each of these instruments generates its tone in roughly the same way, either by using the lips as we have described or by using reeds which operate in the

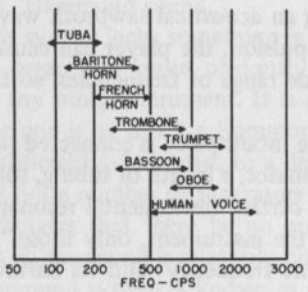


Fig. 5-1–Formant ranges of typical musical instruments.

same way to produce sawtooth waves at the mouthpiece. The instrument bodies differ almost wholly in that their formants are at different frequencies, and that the formant bandwidths differ somewhat. For example, the center frequency of the oboe formant is about the same as that of the trumpet, but the emphasis band is narrower and more pronounced in the oboe, as though there were an electrical filter with a higher-Q inductor. This is sufficient to make a marked difference in sound between the two instruments. As you know to be the fact, and as you would expect, the oboe has a rather tight, constricted sound.

Fig. 5-2 shows the total effect of a formant. You will recall from Chapter 3 that a sawtooth wave contains all harmonics, odd and even, with the relative amplitude of each harmonic inversely proportional to its ordinal number. Bar diagram A in Fig. 5-2 shows the relative strengths of the harmonics of any sawtooth tone generated by the lips or a reed.

At B is the spectrum or frequency response of a musical instrument which has a formant at 1,000 cps. At C we see the amplitudes of the harmonics of a 125-cps tone as the tone emerges from the instrument. It is, of course, the eighth

harmonic which has maximum amplitude. At D is output resulting from a 500-cps input sawtooth, and it has the emphasized second harmonic. At E input is 1,000 cps, and the fundamental has the highest amplitude.

The last diagram illustrates an obvious but important point. Since it is usually the formant which distinguishes the tone color of one instrument from another, any notes having fundamental frequencies higher than the formant would lose

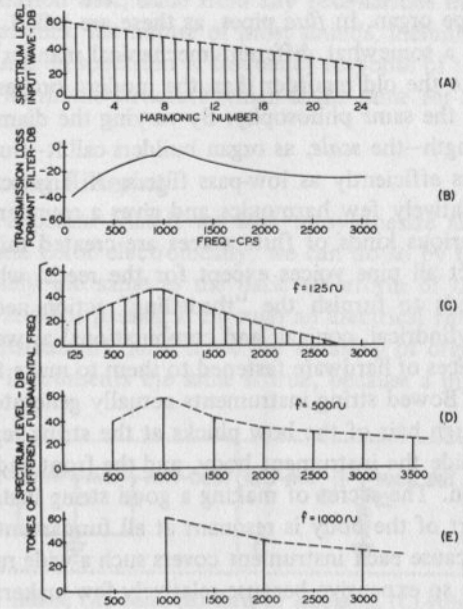


Fig. 5-2–How formant range affects spectrum of an instrument.

their character. And in fact the trumpet, with a formant around 1,000 cps, cannot normally produce any notes higher than the A# at 932 cps. We will come back to this point later.

Other Instrument Characteristics

All the orchestral brass and reed instruments work much as we have described –a sawtooth tone generator followed by an instrument body which in addition to furnishing resonance for the production of fundamental tones acts as an acoustical filter with the relatively small range of frequencies over which it is maximally efficient and produces a formant. These instruments may also have other spectrum filtering characteristics such as high-pass or low-pass action combined with the formant.

The flute operates differently. The player blows a narrow stream of air across the blowhole or embouchure. As he begins to blow, the air goes into the body of the instrument. When sufficient pressure has built up inside the flute, no more air can enter, and the player's breath is deflected *past* the hole. The excess pressure in the flute then forces the air out of it, after which the player's breath is able to enter and a new cycle begins. The fundamental frequency is varied by

opening or stopping up holes along the cylinder of which the flute body is composed, thus varying the effective length of the air column and its resonance. The basic flute tone contains the fundamental and all harmonics, but since the relative diameter of the bore is fairly large, the lower harmonics are emphasized and the upper ones are rather weak. The instrument body is basically, therefore, a low-pass filter. It has no emphasis region or formant.

The flute, in its embodiment as pipes of varied lengths, is the backbone of the pipe organ. In *flue* pipes, as these are called, air is flipped in and out of the pipe in a somewhat different mechanical manner than in the orchestral flute (more like the old recorder than the modern orchestral *flauto traverso*), but according to the same philosophy. By varying the diameter of the pipe with respect to its length—the *scale,* as organ builders call it—flue pipes can be made to act more or less efficiently as low-pass filters. A large-scale (large-diameter) pipe produces relatively few harmonics and gives a rounder, more bland tone, and *vice versa.* Various kinds of flute voices are created this way, as are the diapasons and in fact all pipe voices except for the reeds, which have pipes containing beating reeds to furnish the "throttling" action needed. Pipes vary in shape of bore (cylindrical, conical, and combinations) as well as in having chimneys and other pieces of hardware fastened to them to make for the desired sounds.

Bowed string instruments actually generate tone in the form of pulses as the rough hair of the bow plucks at the string repeatedly. The pulses excite the air inside the instrument body, and the front and back plates are also set into vibration. The secret of making a good string instrument is to build it so that some part of the body is resonant at all fundamental frequencies, and this is not easy because each instrument covers such a wide range. Fine violins, violas, and cellos are so expensive because relatively few makers in history really succeeded in getting all the needed resonances and making them overlap satisfactorily so as to give a fairly even and responsive instrument. There is no real low-pass filtering action, so that the buzz of the string pulses always comes through to some extent—which is desirable. How much body and "mellowness"exists depends on the quality of the fundamental resonances.

As an aside, it is interesting to notice that human speech is based on formants. The vocal chords produce tone which contains all harmonics. The tone passes through the oral cavity. To make a particular vowel sound, we shape the cavity and the opening (the mouth) so that its resonance frequency corresponds to one of several available formants. The tone, with the formant emphasis added to the spectrum response of its path from larynx to outer air, takes on a particular timbre recognizable as a vowel. While the pitch of the fundamental may vary with males and females, from time to time, and with different individuals, the frequency of the formant which causes a particular vowel is about the same for everyone, which is why we can understand each other. If you will say "ee," you can get the impression that the formant is higher than when you say "oo."

If you have ever played a 33-rpm speech record at 45 or 78 you have noticed that the speech is hard or impossible to understand. Obviously, that is not because the fundamental pitches have risen; it is because the formant frequencies

are much higher than they should be and the vowels cannot be recognized.

Another interesting point. If you have ever noticed that the words of a high soprano often cannot be understood, no matter how good her diction, it is because the fundamental pitches she happens to be singing are higher than the formants required for certain vowels; so no matter how hard she tries she cannot pronounce them.

This is merely another demonstration that, aside from any peculiarities that may be inherent in fundamental pitches, the nature of most sounds, including those of musical instruments, depends on spectrum or frequency response of the transmission path, not on a fixed harmonic structure which is the same for all pitches.

Formant Filtering

That being the case, it seems apparent that if we wish to synthesize any steady-state musical instrument tone color electronically, we can do so by (a) generating a waveform approximately the same as the basic waveform of the model we are trying to duplicate, and (b) passing it through an electrical filter having the same response as the instrument. And if we are speaking of organ pipes rather than single orchestral instruments the same is true, because a pipe

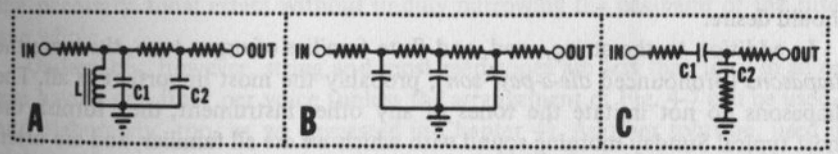


Fig. 5-3—Three basic types of formant filters, (A) reed, (B) flute or diapason, (C) string.

designer shapes his pipes for a given voice so that he achieves about the same effect over the range of pitches as though they were a single instrument.

A tone coloring system of this kind, which is the most common in electronic organs today, is known as a *formant system.* Strictly speaking, a formant is an emphasis band created by a bandpass filter; but the term is commonly used for systems employing all sorts of spectrum shaping, including high-pass, low-pass, bandpass, and combinations.

Pipe tone is divided into four tonal families, and all of these are represented on any good organ, each in several varieties. A distinctive type of filter may be employed to produce the tone of each family, with a good deal of variation within the type to make changes. Representative filters appear in Fig. 5-3, and many more will be found in the chapters on organs.

At A is a typical *reed* filter. The organ reed family includes all voices which are more or less imitative of the orchestral reeds and brasses. The pipes contain reeds, and the electronic filters have tuned circuits to furnish the bandpass action which produces the formant. L and C1 make up a tuned circuit whose values can be set as needed for the frequency of the formant. The Q of L affects the width of the formant range. To produce tone colors having less formant

emphasis and wider passbands, a resistor can be inserted in series with L or C1. This particular filter also provides some low-pass action with C2, a typical example of giving a filter more than one characteristic. This sort of filter is normally fed sawtooth waves.

At B is a *flute* filter, also meant to be used with an organ which generates sawtooth waves. The 3-section low-pass filter attenuates upper harmonics, and a very satisfactory flute tone can result. It will not be a sine wave, but a real flute, orchestral or organ, is not a sine wave either.

A *string* filter appears at C of Fig. 5-3. A sawtooth is easily differentiated to produce pulses and that is the purpose of the high-pass action of which C1 is a part. The proportioning of R and C in the high-pass filtering determines how much fundamental and low-harmonic content is passed, and thus how much body the string tone has. C2 is part of a low-pass section. It is only designed to kill some of the extreme buzz of the pulse in this case, so the resistor in series with it minimizes its effect. Obviously, string tones with more and less body and buzz can be created by selecting values. It is almost impossible in an organ to create the subjective effects of a real bowed string instrument because so much of that effect depends on bowing transients and vibrato variations, as well as other instant to instant things the player does. However, imitation of pipe "strings," created by flue pipes of small diameter, is easy and is probably all one should desire.

In addition to the string, reed, and flute families of organ tone, there are the *diapasons* (pronounced *die-a-pay'-son*), probably the most important of all. The diapasons do not imitate the tones of any other instrument; they furnish the most typical Sunday-morning sound with which we are all familiar, and are often referred to as the backbone of the organ. Diapasons are flue pipes of medium scale, which are flutelike but have more harmonic development than most flutes, the second harmonic usually being especially strong, with gradual rolloff after that. This is enough to give them a kind of life and vibrancy flutes do not have, so that they furnish basic tone with good body and a firm feet-on-the-ground feeling, without being dull or mushy. They are usually voiced fairly loudly and are available all over the organ in most pitch registers. In the 8′ register they are usually called diapasons or principals; in other registers they may be named merely to indicate pitch, like "octave" at 4′, "superoctave" at 2′, etc.

A diapason filter usually looks like a flute filter, but it has less severe low-pass action. Electronic diapason voices can be further distinguished from flutes by giving them a small infusion of second harmonic from the next higher pitch register. An 8′ diapason, for instance, may have a little 4′ tone fed into the filter through a larger resistor than that through which the 8′ input comes.

As usually designed, a formant organ has just one filter for each voice and this gives two problems. First, the tone color must sound as much the same as possible from one end of the keyboard to the other. Where there are pronounced formants and fundamentals must go higher than the formant this is not really possible, but the transition is usually smooth enough so the result is satisfactory. In pipe organs the higher notes of a reed are often produced by flue pipes.

Harder to achieve is equal apparent volume, called *scaling* (see Chapter 4), over the entire keyboard for a given voice.

The scaling problem arises because a filter, in addition to operating on harmonics, also operates on fundamentals, since it cannot tell the difference. A low-pass filter for a flute stop, which is to attenuate harmonics of lower tones, also attenuates fundamentals of upper ones. This can be taken care of by preemphasizing the keyed tones, by scaling the series key-switch resistors or by a network like that of Fig. 4-7, but any such prescaling can really only compensate precisely for the characteristics of one particular voicing filter. In practice, however, it can take care of a good many. First, the ear is not that sensitive a volume indicator, and discrepancies of a few db are not noticeable as long as they do not take place suddenly between one note and the next. Second, less preemphasis of the high fundamentals can be used than measurements would seem to require, because as fundamental frequency rises tones appear to cut through more, due, I suppose, to enhanced sensitivity of the brain to higher pitches. A practical organ, therefore, contains the minimum of preemphasis necessary to produce *apparent* satisfactory scaling for most flute filters. Since this is much less preemphasis than a meter would indicate to be needed, minimum damage is done when the tones are used for high-pass and bandpass filters. Much of the result from the latter depends on clever and judicious design of the string and reed filters to give the necessary tonal effect without unduly narrowing the passband of the filter as a whole.

Undeniably, however, string and most reed tones are less than ideal in an organ having one filter per voice (unless the arrangement of Fig. 4-7 can be used—and elaborately, too, as we suggested in Chapter 4). Another solution, used in elaborate organs sometimes, is to have several filters for each voice, with only a certain range of tones from the key switches fed to each. Each filter can, of course, have the correct spectrum characteristic (all about the same) but the resistive attenuation can be altered to give over-all lower or higher transmission level as necessary and thus preserve scaling. As a matter of practicality, reeds and strings are not often essential at lower pitches unless they are designed that way (16′ reeds, say), but it is certainly very pleasant to play down the scale and have the reeds and strings follow right down without petering out. In pipe organs this is not a problem, of course, since each pipe is individual and the voicer can adjust its sound level as necessary. The same would be true of an organ containing a filter for each generated note, but they are rare. In sum, the extent to which you go to achieve the best results depends on the money you have available.

Stopped Tones

One class of voices which we have not mentioned is produced in organs and certain orchestral instruments such as the clarinet. Its characteristic is that all or almost all even harmonics are suppressed, leaving only the odds. The waveform is symmetrical, and that gives to the tone a sort of hollow or woody quality which is recognized at once. This kind of tone is produced by a quarter-wave pipe or resonator which is closed or *stopped* at one end. The action is the same

acoustically in these instruments and pipes as for a shorted resonant r.f. transmission line of one-quarter wavelength. As with the transmission lines, it is a characteristic of these pipes and instruments that they require only half the length of an open one (which is a half-wave long) for the same fundamental frequency. The clarinet is effectively a stopped "pipe;" it has a cylindrical bore and the acoustical impedance at the mouthpiece is high.

To synthesize stopped tone colors in an organ—certain flutes, for example, including the theatre-organ tibia, the formal-organ stopped flute, stopped diapasons, clarinet, and others—one of the types of formant filters shown in Fig. 5-3 may be used, but a symmetrical waveform is fed to it instead of a sawtooth. In the usual frequency-divider organ, square waves are available at the divider outputs and they can be used directly. The Baldwin organ described in Chapter 9 does this, and then mixes square waves of different pitch registers to obtain assymmetrical tones for other voices by staircasing (Chapter 3).

In frequency-divider organs where the staircasing or sawtooth waveshaping is done at the generators, a dual keying system would be needed if both square and sawtooth tones were to be made available to the voicing filters. To avoid this, an *outphasing* system can be used *after* the keying system to transform sawtooth waves to symmetrical ones. Outphasing with what Schober calls "woodwind circuits" is explained in detail in Chapter 10.

Precisely the opposite problem exists in organs which derive pedal tones from a "pedal generator," which is actually simply a monophonic frequency divider of tones switched to it from the main generators. The resulting 16′ tones are square

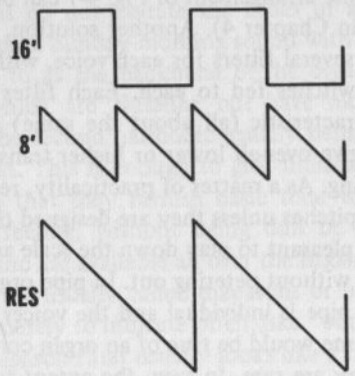


Fig. 5-4—Transforming a 16-foot square waveform to sawtooth.

waves from the pedal-generator flipflop, and they are not suitable for most of the pedal voicing filters (though designers of some organs ignore this fact sometimes). In Fig. 5-4 the 16′ waveform of which two cycles are shown is the output of the pedal frequency divider. The 8′ tone is the sawtooth (or staircased) output of the main generator system selected by a pedal switch and fed to the pedal generator, from which the 16′ tone is obtained by frequency division. If the two are combined with equal amplitudes the resultant is, as shown, a sawtooth at the 16′ pitch.

Analyzing the effect numerically, assume that the lower (square-wave) frequency is *f*. It generates harmonics 3*f*, 5*f*, 7*f*, etc. The fundamental of the tone (sawtooth) an octave higher is 2*f*. It generates harmonics of 2 X 2 or 4*f*, 3 X 2 or 6*f*, 4 X 2 or 8*f*, etc. As you can see, this fills in all the missing harmonics of the lower tone.

Other Voicing Systems

Harmonic synthesis and formant filters constitute the two clearcut voicing systems, of which the latter is by far the most often used today. There are, however, some other systems.

There are some organs, invariably those which generate tone with individual oscillators rather than frequency dividers, which provide flute tones by using simple sine waves from the oscillators. I personally believe this practice makes an organ tonally less interesting because real organ (or orchestral) flutes are not completely lacking in harmonics, and the lack of any, inherent in a sine wave, makes the flute voices uniformly too bland. It also pretty well removes the possibility of providing any variety of flute stops.

Allen and Rodgers organs create diapasons by half-wave rectifying sine waves, and then adding some low-pass filtering. This was a highly original idea to me, but with the usual 20-20 hindsight it makes some sense. The rectification creates a strong second harmonic—which we often do in formant filtering for diapasons by leaking in some of the next higher pitch register—plus, of course, other harmonics, which can be removed as desired by filtering.

Lowrey produces bland flutes from square waves by chopping up the keying system and feeding small groups of notes to cutoff filters, which are also used in a slightly different way to obtain the other voices. Thomas feeds sections of the key output busses to a single multistage L-C low-pass filter to get rid of harmonics and make flute tones. Allen generates sine waves, then distorts and filters them to some extent at each oscillator. This, by the way, is the only way to get 100 percent perfect scaling, because you have the same freedom as the pipe voicer to adjust the output of each tone for each voice. It is, of course, either highly expensive or greatly restrictive in terms of voicing variety, whichever you choose. Conn uses formant filters for all voices other than the flutes, but besides a sawtooth waveform Conn uses a pulse of special shape with, of course, filters designed to utilize that pulse.

Advice To Designers

Because the voicing of an organ is what eventually makes or breaks it (musically if not always from the sales department's viewpoint), it is perhaps the most important aspect of design. It is also technically one of the easier aspects, for the choice of a voicing system and eventual detailed selection of components does not demand genius. Unfortunately, voicing is not really a technical function at all, but a musical one. There is simply no way in which an engineer can voice an organ successfully by depending on engineering knowledge and training. Only a team composed of an engineer *and a musician* can do a suitable job, though when a company is fortunate both may be combined in one envelope.

Certain things can be said which I hope both members of the team will take to heart. First, plan to have as many voices as costs will allow; and generally voicing is cheaper than almost anything else, especially in a formant system. Second, throw away the harmonic analyzers and the slide rules. The only way to design voicing circuits is experimentally, judging the results with an educated ear and actually playing real music at every step in the process. (The latter is why I specify a musician rather than merely someone who has listened to a lot of music.) Third, give as wide a range of *different* tone colors as possible, which means inclusion of some good reed and string voices despite the two little old ladies who complained to the local dealer only last week about the organ's harshness. They just didn't know that every tab has an off position.

Selecting the voices for the specification is one of the toughest of jobs. It can only be done by someone highly familiar with organs—or by looking hard at the competition and copying a specification straight from one of his models you like. You not only must have a sufficient number of voices which make good solos, but all or almost all of your voices must be usable in good combinations. To make sure this is so, each time you arrive at a tentative selection of components for a voice, the organ must be played in every possible way, using the new voice and also all the ones previously arrived at. You must also be willing to go back and change a previous voice at any time up to the point where the design is completely frozen, if it turns out to be out of balance with later ones when used in combination.

Whether or not you aim to make your voices sound like pipes is a matter of choice. It is not always necessary, though pipe sounds are sufficiently varied to afford about anything desired, and they do give a point of reference. Do not worry excessively about making voices which are exact imitations of orchestral instruments, though some of the more obvious ones are useful in both music and sales—oboe, trumpet, clarinet, etc.

Chapter 6

Special Effects And Equipment

The functions we have discussed in the last three chapters—tone generation, keying, and tone coloring—are the essential ones. The remaining elements of an electronic organ are sound production, which is largely a conventional audio function, plus any special effects a particular instrument may have. This chapter looks briefly at what special effects are available, and also gives some nuts and bolts information about special kinds of equipment needed for organs, which cannot be found through the usual electronic manufacturing channels.

Vibrato

I do not know of any electronic organ which does not include *vibrato,* and indeed there are very few musical instruments of any kind (with the notable exception of most keyboard types) which cannot be "vibrated." Vibrato is a cyclic variation in the pitch of a tone which, under most circumstances, is most pleasing at the rate of about 7 per second. A second effect, *tremolo,* also at the same rate, is a variation of amplitude rather than pitch. The pipe organ incorporates it, along with a bit of vibrato, but few other instruments do, and then only as a matter of necessity, for it is much less pleasing than vibrato.

The need for vibrato arises because the brain finds a single tone of constant precise pitch and volume uninteresting, regardless of its tone color. In fact, human beings find practically everything less and less interesting as it approaches "perfection," even though near perfection may sometimes be highly desirable for practical purposes other than simply being interesting and attractive in an esthetic sense. The thought, however, is more profound than you may think at first reading. In addition, it is one that should be kept firmly in mind by electronic organ designers.

To begin at the top, Sir Thomas Moore in 1516 invented the Commonwealth of Utopia, a society in which everything would function and everyone would live perfectly. (The word was made up from the Greek *οὐ–not–*plus *τόπος–place–*meaning *Nowhere.**) While at first glance many of us in this troubled world might sigh for such a life, it is apparent that unwavering perfection would

*This is just meant to display my erudition—courtesy of the Encyclopædia Britannica.

be a monumental bore. It is said, in fact, that in a country like Sweden, where what many people call socialism has solved most material problems for most people, the high suicide rate is at least partly due to this very boredom, caused by the lack of necessity to cope with and surmount many of the problems which in other societies would mar the smoothness of life.

In the visual arts a painting is more highly prized and produces greater emotional response than a photograph as a rule. One of the large reasons is that most photographs are merely perfect mechanical representations of what already exists. A painting, on the other hand, is technically quite imperfect, which lends it interest. There are, of course, some "photographic" painters, who faithfully reproduce every line and shading. To the extent of this perfection, such painters are ineffective unless, as, for example, in the case of Salvador Dali, the literalness of reproduction is combined with some other kind of variation like rearrangements of parts or startling combinations.

Variation and imperfection add interest even to people. The Miss Rheingold "beauty" of perfectly regular features may attract an appreciative glance from nearby men, but the great beauties of the world have been those who look unusual, generally because of some "defect," like a large mouth, a slightly crooked tooth, thick lips, or some other variant. (Even the black "beauty spot" fashionable at one time indicates by its name that it was an imperfection deliberately added to increase beauty!) And most women of mature judgement will readily admit that the most attractive man is not the collar-ad face but one with character made visual by imperfections. Even nonphysical aspects of people conform to the rule. The person of unvarying pleasantness and unfailing good humor becomes a crashing bore after a while. (I do not suffer from this problem.)

A musical instrument cannot, therefore, be artistically satisfying if it is a total product of a scientific outlook which regards perfection as the aim of the game. Engineers are, of course, trained to try for perfection. That is a fine thing when you are designing a computer or a high-fidelity amplifier. But in designing a musical instrument, some way must be found to mask and diffuse the inherent perfection of electronic tone production and give tones the warmth and interest they need.

A primary method of enhancement is *chorus effect,* the simultaneous sounding of several tones at the same or octavely related pitches, each tone sounding at a slightly different frequency. This point was discussed in Chapter 3 and the effect can be obtained only in organs having several ranks of generators. A second method is *reverberation*—real or artificial acoustic enhancement, discussed in Chapter 7. The third method is vibrato, the most easily available of all in electronic instruments.

Vibrato is produced in electronic organs by three different methods. The most common is variation of the frequency of the self controlled L-C oscillators by an a.c. signal of around 7 cps, and the method is used in both frequency-divider and other instruments. The chapters on commercial organs show several ways of doing this, as well as several vibrato-frequency oscillators.

Another method is necessary for organs whose tone generators are inherently

incapable of being varied in frequency, like spinning wheels or reeds. The tones can be passed through special circuits or devices which shift phase at a vibrato rate. While the ear is normally insensitive to the phase of a signal as such, a constantly *shifting* phase produces a Doppler effect which alters apparent frequency. Phase-shift vibratos are described in Chapters 10 and 16.

A third method also involves dynamic phase shift, but acoustical instead of electronic. It is produced by a rotating speaker or baffle, usually one model or another of the Leslie Organ Speaker, manufactured by Electro Music, Inc., Pasadena, Calif. Leslie speakers are mentioned in Chapter 7.

The tone-generator frequency-swing method has the disadvantage that if a single rank of generators provides all tone for the instrument, vibrato, when turned on, affects all tones. This can be a nuisance if, for example, there is a percussion circuit with an effect that sounds unnatural with vibrato. The system also inevitably puts vibrato on the pedal tones. While this is not fatal, low pedal tones sound rather sick with vibrato, probably because the vibrato frequency is such a large percentage of the audio frequency. The final disadvantage is that it is not easy to make oscillators for the generators which are maximally stable and at the same time will vary in pitch at the command of an injected signal.

The one advantage of applying vibrato directly to the generators is that the system is the simplest of all. Unlike most phase-shift systems, it also results in the same percentage frequency deviation for all harmonics and, in divider systems, for all fundamentals as well. Depending on the effect one wants, this may be either good or bad. Personally, I don't much like it; I prefer the characteristics of systems which concentrate maximum vibrato effect over a given central band of the spectrum, with less violent vibrato for lower fundamentals and upper harmonics. This is purely personal and you may disagree.

In any organ meant for popular, standard, and theatrical music, a good vibrato is essential, meaning a frequency deviation on the order of ±2½ or even 3%, at least as a maximum available over the middle three 8′ octaves of fundamentals. More than that can sound rather sickening; less is simply inadequate, to give the desirable juicy effect to some popular music. There should always be some method available to the organist for greatly decreasing this maximum, since by no means all music ought to have it, or even any vibrato at all. Methods of having vibrato on one manual and not the other are also very useful.

Other Special Effects

The most common special effects (aside from vibrato) to be found on present electronic organs are *sustain* and *percussion.* Both were defined in Chapter 2 and require some sort of electronic keying system rather than simple physical switching. A sustain is arranged so that the on potential is applied to a keyer almost immediately when a key is pressed, while the release of a key allows it to decay slowly. R-C time-constant circuits are always employed. Percussion systems are more elaborate, since the on potential must begin to decay as soon as it appears even if the key is held. Systems of both types are discussed in following chapters.

The unfortunate aspect of both sustain and percussion is that they do require

an electronic keying system, meaning at least one diode or transistor keyer for each note. In an organ which already has electronic keying for the purpose of more closely approximating the relatively gradual speech of pipes, adding the necessary R-C networks for sustain or percussion is not usually a major expense factor. In organs which are basically fairly expensive, therefore, the purchaser does not pay disproportionately for sustain or percussion.

In less expensive organs, however, it has become common to include electronic keying for no other reason than to give sustain or percussions. In these organs, even though the electronic keyers may also be used wholly or partly for normal keying as well, straight keying could have been used and the instrument's price contains too many dollars which could have gone into more keys, more generators, and/or more voices, any of which would have yielded more solid, long-term benefits for the owner. On this subject I am either a voice crying in the wilderness or a nut; you will have to take your choice.

Almost every manufacturer also offers other special effects and devices, many of which are shown in remaining chapters.

Chord Organs

The easy-to-play feature has been a strong factor in electronic organ promotion since the advent of models primarily intended for the home and for the inexperienced player—which is where the largest dollar market lies. The chord organ was an effort to make the organ even more easily playable—or at least to make it seem so—and for a relatively short time chord organs were popular. Today very few electronic chord organs are sold, compared to the rest of the market, and so far as I know, only Hammond—who originated them in the first place—still makes a chord organ.

The Hammond chord organ has a 37-note keyboard on which both melody notes and harmonies can be played with the right hand. At its left is a bank of 96 pushbuttons, representing eight chords for each of the 12 keys in which music can be played. The idea is that instead of having to learn chords as played in the normal way, on black and white keys, the player has only to push a marked button for any chord he wants, plus a bass note.

The fallacy is that it is actually easier (after the first extremely simple selections are learned) to locate the notes of a chord than it is to find a series of buttons. As I pointed out in an earlier chapter, the keyboard, with its groups of two and three black notes, makes finding notes extremely easy, so much so that after a relatively short learning period it becomes a series of reflexes. The use of buttons, too, each always with its invariant automatic selection of the same three notes for a given chord, severely restricts playing style. A few performers did achieve quite satisfactory results on the chord organ. I suspect they treated it just as an accordionist does his instrument, which was the more possible because the buttons were laid out like accordion "basses"—the buttons which the left hand uses to select chords and bass notes. Accordionists learn these by touch. If memory serves, the aural results of good chord organ players were always very like those of accordionists—a distinctive, rather choppy style of playing.

At any rate, the chord organ did not hold the public's fancy very long, and rightly so. The handwriting on the wall was obvious even before the instrument's relegation to the background; an enormous percentage of owners traded them in for conventional organs within a few months. Of course, even this was of value to the maker. The chord organ undoubtedly persuaded a great many people to get their feet wet in the first place, after which they "traded up."

Two or three other makers also went briefly into chord organs. The only interesting one was Electro-Voice, in a very inexpensive instrument otherwise unremarkable. It had two conventional (short) keyboards. However, at the turn of a switch, each of the lowest 17 keys of the lower manual, instead of playing individual notes, played a complete chord and a bass note. A system like this offered the best of both worlds—a chord organ when that particular kind of simplification was desired (and with recognizable keys to produce the chords instead of anonymous buttons) or a conventional organ at other times.

If you happen to be interested in details of the Hammond chord organ, it is fully described in the first two editions of this book, available in most public libraries. (I hope, of course, that you have bought your own copy of the present edition, as I make more money that way.)

Organ Equipment

While the guts of electronic organs are composed of the same items as you will find in any other electronic apparatus, certain physical equipment is special and is unfamiliar to people who have not worked with organs. Although some organ manufacturers brew their own home versions of these items, most of them are now available for purchase from specialty manufacturers for companies (and sometimes individuals) who prefer not to put in the necessary design time or tooling money.

Keyboards are, of course, basic. Ordinary wooden pipe-organ manuals (as well as piano keyboards) are principally available from two manufacturers in the U. S., Pratt, Read & Co., Inc., and Wood & Brooks. As furnished by these makers, they consist of white keys covered, usually, in plastic, blacks which are plastic caps on wood sticks, frame, pivot pins, and front guides, all assembled. There is, however, no upstop system or downstop felt, and no springing, all of which must be added by the organ manufacturer. Of course, no key switching is furnished.

The majority of organ makers employ all-plastic (cored) keys, usually fastened to a metal rail by flat springs which serve both to keep the keys in place and to furnish tension. However, an excellent steel and plastic keyboard is available to the industry from Pratt, Read. Used by several manufacturers, it is available with and without a very good switching system. It is described briefly in Chapter 10. The Pratt, Read metal-plastic keyboard is, in my opinion, better than most others, but may be more expensive than some (assuming the high tooling costs a company must undergo to mold its own have been amortized). There is nothing to recommend a wooden keyboard other than perhaps low cost, except that usually their pivot pins are farther back from the key fronts, so that a pressed key is not slanting downward at its front so much. In the Pratt, Read metal-plastic

keyboard the pivot is farther back than in most others and the difference in playing feel between it and wood keyboards is negligible. Plastic keyboards are also available from Italian accordion makers, but not, so far as I know, with an

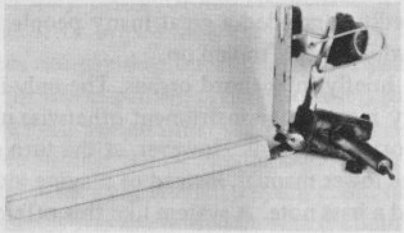


Fig. 6-1–Stop tablet of the tongue type.

integral switching system. I have never seen one and can therefore say nothing of their quality.

There is no such thing as a standard key-switching system; each manufacturer designs his own. Some remarks about switches are in Chapter 4.

Stop tablets are of two types (ignoring the drawknobs so seldom used in electronics). Fig. 6-1 shows one simple and very good tongue tablet available from Klann, Inc., Waynesboro, Va. The base is sheet steel with its lower end formed into a pair of loops with a pin passing through them. The twisted steel back piece is pivoted to the pin and a plastic tablet is fastened to its bent bottom. As the tablet is pushed up and down, the back piece moves and the black plastic molded around it contains an actuator with a pin which goes up and down and will actuate a switch which may be a flat-blade type. The pin is in an arm which is inserted into a hole in the black plastic and can be adjusted as to height by pivoting, then held in adjustment by a setscrew. These tablets come in strips–a single front plate and a series of tablet mechanisms fastened to it.

Fig. 6-2 shows a different kind of tongue tablet. Here a bent plate holding the tablet is pivoted to the front plate of a steel frame by a pin. The frame contains a pair of electromagnets which can be energized as part of a combination action.

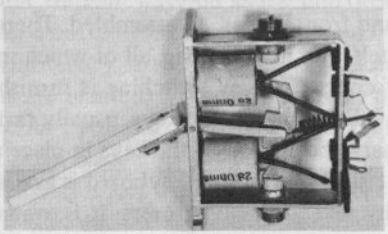


Fig. 6-2–Electromagnetic stop tablet for combination actions.

flipping the tab up and down electrically. Both these tablets, as well as most others, have a toggle action, but since the actuating force has to be light to suit the organist and operation must be silent, the toggle action is not positive as in a conventional toggle switch.

There are some different styles in tongue tablets, but the usual one is made so it will extend roughly 2 inches from the slot in the wood stop board (assuming 3/4-inch wood). It is 3/4 to 7/8 inch wide. Width of the mechanism is usually such that the tablets can be spaced on 7/8-inch centers.

Rocker tablets are customary on smaller organs. A few are merely flat pieces of plastic (notably Baldwin's), but most are of box shape with open bottoms. Actions vary; there is usually some toggle action. Dimensions are usually on the order of 3/4 X 1 3/4 inches. The rocker tablets in Fig. 6-3 represent, to my knowledge, the only ones made for the electronic industry by an outside manufacturer, again Pratt, Read (who also offer tongue tablets).

Swell shoes are easy enough for an individual manufacturer to design and make, especially since he can, if he likes, buy the major part of it, the molded pedal, from Harry Davies Molding Co., Inc. Whatever shoe is used is pivoted to a wood or metal base, with suitable padded upstop and downstop. Swell shoes

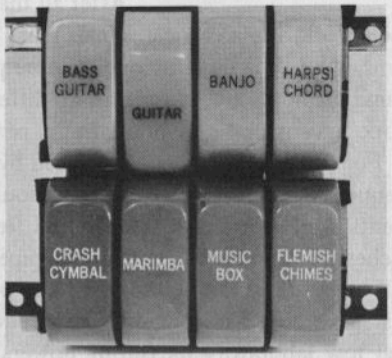


Fig. 6-3–Group of rocker-type tablets.

pivot at the center and the pivot has friction so that the movement presents a little resistance to help the foot control it, and so it will stay where it is put. There is no spring return as on an automobile accelerator. If a potentiometer is to be operated by the shoe, a rack and pinion arrangement is usual. A plastic gear for this purpose is available from Klann, as is a complete shoe assembly. The range of movement of a swell shoe is around 30 degrees. Be careful to fix things so that the range is not too great or the full on position is not so nearly flat that the organist's ankle is overworked. Other remarks about swell shoes appear in Chapter 2.

Large pedal claviers of the standard-organ type can be purchased from Klann, The W. H. Riesner Mfg. Co., Inc., Hagerstown, Md., and Organ Supply Corp. and Durst, Inc., both of Erie, Pa., but most organ makers manufacture their own. Specifications appear in the Appendix. Toe-pedal assemblies may be available from Pratt, Read, but again most are home grown. The black sharp caps can be purchased for standard claviers from Organ Supply and for toe pedals (a smaller size) from Harry Davies.

Consoles are available from Klann and Riesner. Though most makers make or contract for their own, these supply company items are useful for companies planning to make custom organs, with too few of any one specification to justify a local effort. The supply houses will equip a console with any ancillary equipment desired–stop tabs, keys, pedals, combination actions, lights, etc.

Chapter 7

Sound Production And Reverberation

After all the organ circuits and mechanisms have done their work, the electrical signals must be converted to sound. As in all audio systems, the job is done by one or more power amplifiers and loudspeaker systems. These elements are, in general, no different from those used for all sorts of audio work, except that certain minimum performance levels are necessary.

For one thing, the bass produced by an organ usually extends farther down in frequency—to 32.7 cps—than most sound sources, since the only common instruments which produce fundamental frequencies of Octave 1 (Table 1-1) are the orchestral double-bass and organ pipes. Obviously, the best production of the full organ (and orchestral) range is possible only with amplifiers and speaker systems which have not only flat frequency response down to 32 cps at the normal low laboratory measuring powers but also flat power response. The power content of music in the high regions above one or two thousand cycles tapers off sharply, so that appreciable power capability for the high fundamentals and harmonics is not needed. The greatest power requirement is in the bass, and this imposes stringent requirements on the equipment. In every case, the amplifier, at least, must be able to take care of real power right down to the bottom of the range. If it cannot, it will be overdriven and severe intermodulation will destroy the total effect.

A loudspeaker system with insufficient bass capabilities is a horse from a somewhat different garage. If it is built sturdily it will not be damaged by having to soak up power at frequencies below those it adequately reproduces, so the only effect is that bass fundamentals will not be reproduced. And this is not fatal, even in an organ, because of a phenomenon known as *virtual bass.* If, let us say, low C at 32.7 cps is fed to a speaker system which cuts off at about 60 cps, the 32.7-cps fundamental is not heard. The harmonics are, however. The frequency difference between each harmonic and the next is equal to the fundamental, and the ear automatically senses this frequency difference between harmonics and deduces from that the fundamental, which it then believes it is hearing. Thus, even on a system that cuts off at 60 cps, the listener can tell the difference between the low 32-cps C and the 65-cps C an octave higher. This is only true, of course, when the pedal stops are voiced to have some harmonic development. The phenomenon can be used, in fact, to give an organ a pedal 32′ register

(which should go down to 16 cps) even though it is playing through a speaker system which only goes down to 32 cps. The trick is to voice the 32′ stop as a string or reed which is not supposed to have a strong fundamental but has lots of harmonics. A stop like a bourdon, which is almost all fundamental, does not do well in any of this hokus pocus.

There is still a big difference, however, between being able to detect the frequency of the fundamental and actually hearing and feeling it. The woods are full of stereo system owners today who have never really heard the solid bass imparted to a full orchestra passage by the bass fiddles (unless they have gone to a concert hall), but are quite satisfied all the same with the reproduction of their systems—until one day they visit a friend and hear what real bass is, with its enveloping, stomach-pushing effects.

Good bass reproduction is no easier or cheaper to achieve in an electronic organ than it is in a hi-fi system. It requires a speaker system with enough radiating surface to push an adequate volume of air, which means either an efficient speaker of large diameter or an enclosure precisely designed to complement the driver in radiating adequate bass, such as a horn or one of the varieties of bass reflex. In smaller and less expensive organs, therefore, good fundamental bass is not common. The speakers usually are chosen for sturdiness rather than low resonance, and, more important, an organ console is usually designed to house parts and look nice, not as the ideal acoustic enclosure. In most cases, the back of the organ is not even sealed off by a panel adequately rigid and airtight to serve any acoustical purpose—and sometimes there is no back at all. Unfortunately, when a speaker system (system meaning drivers + enclosure, each equally important) decides to cut off, it does so in no uncertain terms, and no amount of preemphasis in the amplifier or prior electronics can offset the bass lack.

Good fundamental bass very definitely enhances the musical effect of an organ. It is worthwhile for the designer to do everything possible to achieve it. Where speakers must be in the organ console certain things help—drivers with as large cone radiating surfaces as possible and with large magnets, and whatever acoustic design can be applied to the console, including solid, well fitting backs or even good acoustic subenclosures built inside the console.

Treble production is not nearly so great a problem. Even rather inexpensive cone speakers can do it well enough, if quality is at least high enough to eliminate severe response peaks and dips (which would be equivalent to putting permanent formants in the organ) and to extend response out to around 12,000 cps, where the organ can stand it. Some organs are not meant to have extended treble reproduction and the speaker system is deliberately designed as a part of the over-all spectrum shaping to eliminate stridencies or treble keying transients. Hammond is a case like this; since no audio above 6,000 cps is generated, full treble response would add nothing to the music but key clicks.

In any organ having separate drivers for bass and treble (and there really should be no organ without them, however cheap), there should be an electrical crossover network which not only keeps bass out of the tweeter to avoid destroying it, but also keeps treble out of the woofer to reduce intermodulation.

A low crossover frequency gives the best results—perhaps 250 to 300 cps.

It is impossible to make a rule about the audio power which ought to be available in organs. Probably 20 *steady-state sine-wave* watts is the barest minimum for a small room. The requirement is tied in with two problems. The first is that peaks build in organ music, especially in frequency-divider organs when octaves are played and voices added, all from the same phase-locked sources. Any such peak that sticks its head above water and is clipped causes bad distortion. Audible results are often more specific, too, than simple mushy or slightly broken up sound. Playing two adjacent or nearby notes (depending on the octave involved) creates an IM product which may be quite audible as a low-frequency note having no musical relationship to the others being played. The result is most unpleasant. This point also dictates the lowest possible nonlinearity in all amplifier stages within the instrument, well beyond the highest peak signal level that will be present. Intermodulation problems are especially bad in an organ, too, because of the treble and bass extremities available. A low pedal note can easily modulate a higher manual note, with most unfortunate results.

A second aspect is that organ stops, in addition to differing in pitch register and tone color, ought to vary quite widely in volume. This is an essential requirement for flexibility in making up ensembles of stops—not just to be able to play loudly or softly over-all, something which the swell shoe can control. If the loudnesses apportioned to the various stops resemble those of a pipe organ, and if the amplifier gain control is adjusted so that with combinations of only soft stops the volume is adequate, loud combinations may cause overload of an inadequate amplifier. Or conversely, if the organ is adjusted so that the amplifier will not overload with the loudest combinations the soft stops may be too slight for good results. All this means that more rather than less power should be the rule.

One answer to the last mentioned problem, of course, is to limit the differences possible in the organ from the loudest to softest stops. This can easily be done (and has been) by arranging the stop switching so that as more stops are switched on, a bus is more heavily loaded and stops already on become softer. This is a poor solution (but better than none), because a cardinal rule of a good organ is that no stop shall affect another; all of them should simply add as though they were separate instruments. This solution degrades the basic organ.

Merely making all stops of about the same loudness not only kills one of the good points of an organ but does not do the job if you can still build up sound level by adding more stops to the ensemble.

A solution of my own has been a volume compressor which can be adjusted by the user for his own installation and circumstances. As the organ puts out higher sound level, the gain of the system is reduced so that variations of level fed to the amplifier are less than those supplied by the instrument. The compressor is adjustable. An owner with a small amplifier, or one whose family and neighbors severely limit the maximum loudness of his playing, adjusts for greater compression, so that his soft combinations are not enormously softer than his loud ones and thus can be heard satisfactorily. But in an installation with a powerful sound system and no restrictions on sound power, the user can adjust

the unit for the widest differences he likes. The virtue of the adjustable compressor is that it allows each installation the maximum dynamic range it can tolerate and does not limit the basic design of the organ to take care of the worst cases. To my knowledge, no other manufacturer uses a compressor, but our results have been gratifying, as reported by customers.

It is unfortunate that most organs must have speakers built into the console, largely because (a) it is cheaper than a separate cabinet and (b) most people (especially women) balk at the idea of an extra piece of furniture in today's fairly small rooms. Even if good sound reproduction is built into a console—and it can be—the sound source is too close to the audience, which is usually the organist himself. This is again the plague of perfection discussed in Chapter 6; the sound is too apparent, too cleanly and precisely discernible, especially when the speakers are in the usual place, the toeboard, spraying sound right up into the organist's face. The effect can be ameliorated to a degree by having the speakers on the back of the console, which is then placed an inch or two from a hard wall which diffuses the middle and high frequencies.

By far superior sound production is furnished by what the trade calls a *tone cabinet,* which is simply a loudspeaker system complete in its own enclosure, usually with an amplifier inside. Most organs have tone cabinet connectors in the console and most manufacturers have tone cabinets available. However, except with large organs that simply do not have built-in speakers, no one seems to have had much success in persuading the public to buy tone cabinets. A tone cabinet is usually capable of far better sound than any built-in system; and it can be placed on the other side of the room so that the organist is not, in effect, sitting on a volcano.

Multichannel Organs

Just as stereo systems for record and radio production have taken over in recent years, a great many organs produce sound through more than one audio channel. One manual may sound through one speaker and the other through another; or flutes may go through one channel and all other voices through a second; or division may be effected in other ways—and through as many as six channels in expensive instruments.

There is a strong question in my mind as to whether this sort of elaboration is worthwhile in terms of more satisfying audio effects. Certainly the effect is not "stereo," as some advertising claims. Stereophonic sound *spreads out* the apparent sound source over an *area,* while multichannel organs merely produce all of one channel from one side and all of the other channel from the other side, with nothing in between. And where both speakers are in a small console, not even much of *that* kind of division is discernible unless you put your head on the floor about where the swell shoe is.

It is sometimes claimed that the division simulates the effect of many pipe organs which have pipes in two or more chambers some distance apart. That may be so, but it has never seemed very pleasant to me. I remember so well my first hearing of a pipe organ of this kind, which had a chamber above each side

of the altar, perhaps 30 feet apart. As tones would come from one chamber and the other, I had something of the sensation of a spectator at a tennis match—eye, head, and attention following the ball from court to court. Highly distracting.

My own preference is to have all sound coming through one channel. For enhancement I like to have the composite signal driving two or more speaker systems placed around the room, so that I, as the listener (and organist), feel surrounded by sound, none of which is easily identifiable as to source.

I happen to have had an excellent opportunity to see what people will do when given their choice on this question. I designed dual preamplifiers into the Schober Recital Organ; with the change of a single connection, the owner can either have the entire organ sounding through one audio system or each manual sounding through a separate one. The organ is large and not cheap, and a large percentage of owners, whose roster includes a lot of technical people and those of means, add quite expensive optional equipment. But only a very small percentage of them have chosen to utilize the dual-channel possibility. I will readily admit, of course, that organ kit builders are not typical of the vast organ buying public. They are undoubtedly more discriminating because (a) they include more technical people, (b) they must have a certain amount of enterprise to decide to embark on such an elaborate kit project, and (c) they end the project knowing much more about organs than ordinary plug-it-in-and-play buyers. These same kit builders, by the way, also differ from the mob in that by far the great majority of them use tone cabinets even though quite good built-in speakers are available for each organ model. I think this shows the results both of our brainwashing them on this subject and of their own greater concern for the best possible sound quality.

The Leslie Organ Speaker

One special kind of loudspeaker system is in wide use in the industry, available with or incorporated in organs of almost every manufacturer. Made by Electro Music, Inc., and invented by Don Leslie, it is the Leslie Organ Speaker. It is made in several forms and models, but the major purpose of them all is to impart both phase and amplitude modulation to organ tones with a rotating system which directs the output of a loudspeaker continuously around a 360-degree circle at one of two selectable rotational rates, either approximately 7 times per second (tremolo speed) or about once every 1.3 seconds (for the so called "chorale" effect).

Fig. 7-1 shows in essence one form the Leslie system takes. The speaker points down into a rotor. Made of a very light substance, the rotor core is shaped so as to direct the sound to an opening in the rim so that the greatest sound amplitude radiates in the direction of the opening. When the rotor turns, the sound amplitude varies, to a stationary listener, according to the direction of radiation at each instant, as well as to the nature and distance of reflecting surfaces in the room at each point surrounding the system. The cabinet has openings, of course, on all four sides. This produces amplitude modulation.

As the system rotates, the sound heard by a listener also traverses paths of

different lengths, depending again on the position of the rotor and on the distances of reflecting surfaces. The cyclically changing transmission-path differences cause an acoustical Doppler phase shift, which is heard as a cyclic frequency shift. When the rotation rate is fast the aural result is combined tremolo and vibrato. When it is switched to the slow rate the tones acquire a sort of movement which tends to take away the coldness of electronic perfection without introducing vibrato.

Depending on the room, the effects of the Leslie can be extremely complex (which is all to the good), perhaps more so than would be economically feasible with a purely electronic device. The vibrato-tremolo effect is very strong and it tends to help increase the theatrical quality of an organ. The slow "chorale" speed is very definitely cyclic, a drawback, but in most cases it is better than nothing.

Some Leslie systems contain "straight" as well as rotating speakers. The speaker associated with the rotating element is deliberately made so as not to reproduce higher overtones; the maker intends it only for use with flutes, on the premise that the rotation would cause movement of upper harmonics of more

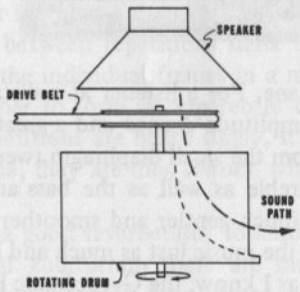


Fig. 7-1—"Schematic" representation of Leslie action. As drum rotates, sound is sprayed around a full circle.

complex tones which would conflict with the beats caused by tempered tuning and produce what one organ engineer has called a *chigachigawahwah.* For this reason, many organs incorporate separate audio channels for the flute stops, so that they can be fed independently to a Leslie. I can state from experience that this is not really necessary. I can also say that if an organ has a good internal vibrato the benefit of a Leslie is less apparent, though it still adds something if you like its rather violent effect.

Originally Leslie speakers were used only with Hammond organs, for which, in fact, they were developed (independently) as an effort to add some excitement to the inherently rather insipid Hammond tone. In recent years, however, Leslies are found built into the consoles of even the cheapest home organs of all brands, to the point where the public calls for them by name without, often, even knowing what a "Leslie" is. The company, in fact, enjoys the position of being the only outside manufacturer whose products are used by almost all organ makers and are demanded by large numbers of customers. CBS bought up the company not long ago, and not, I should think, for tax losses.

Gyrophonic Projector

One other type of rotating speaker has become well known, Allen's Gyrophonic Projector, an inside view of which (an old model) appears in Fig. 7-2. The Leslie rotation is in the horizontal plane and directs the sound alternately toward and away from the listener. The Gyrophonic Projector speakers are mounted on a circular front-panel cutout with an axle, so that they rotate in the vertical plane. For a listener at a distance, the effect is more phase modulation and less amplitude change and a great deal less violence in both respects. As you can see from the small diaphragm tweeters in the photo, Allen is not afraid to rotate the treble as well as the bass and middle tone sources. The sound of this system is much gentler and smoother than that of the Leslie. To my own ears it warms up the music just as much and I like it better.

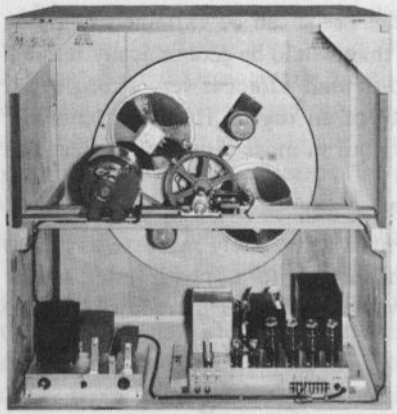


Fig. 7-2–Inside view of a Gyrophonic cabinet shows how speakers are rotated. This is an old model.

So far as I know, the Gyrophonic Projector is not (officially) available for use with organs other than Allen, but I know people who have rolled their own.

Having kept for some years a complete file of U. S. patents on everything connected with electronic music, I know that there are a great many "inventions" dealing with rotating elements in speaker systems. I also know by experiment that a great many of them are practical, and that practically anything that moves enhances the sound in some way. It is all the more interesting, then, to note that there are no systems with any reputations or popularity other than the two I have mentioned. This is so true that organ makers fully capable of cooking up their own buy Leslie instead because it's the name the customers know. Is this just good merchandising, the luck of getting there firstest with the mostest, or what? I'd like to know how to duplicate the feat with Reverbatape!

Reverberation

There is almost no musical instrument or ensemble which does not sound better in a hall having some reverberation than in an acoustically dead room. This is especially true of an organ–pipe or electronic–since the organ is the most mechanical of all instruments.

Reverberation is so important, in fact, that without it even the most realistically voiced electronic organ cannot give the musical impression one expects from an organ; while with it even many organs with inferior tone sound unexpectedly pipelike and attractive. Since most organs today are placed in homes, where rooms are relatively small and acoustically dead, reverberation can only be supplied artificially.

Reverberation is the "sound of a large auditorium" which is brought about by the bounces of the original sound against the walls of large, live halls, with consequent audible repetitions of each sound at each bounce. The total effect caused by the reverberatory bounces is an apparent slow decay of the sound after each note has ended, rather than the abrupt, cold harshness obtained in a small, padded room where everything is heard with razor-sharp precision and immediacy. Electronic organs, with their scientific precision of sound, are too precise and impersonal to begin with. The defocusing and softening effects of reverberation are really a necessity to give both player and audience the emotional satisfaction and sense of bigness so attractive in a church or theatre organ.

Though not easy to achieve in practice, the requirements for an artificial reverberation system are simply stated. It is only necessary to pass the sound electronically through a device which will repeat each sound continuously at short intervals, each repetition somewhat lower in volume than the last, until finally nothing more can be heard. The period between repetitions must be short enough so that, as happens visually with the individual frames in a motion picture, the discrete repetitions blend together to form an apparently continuous sound which gradually decays. If the repetitions are heard singly, it is because they are not close enough together in time; they are then *echoes,* which are unpleasant.

Certain characteristics are desirable in a good reverberator to lend the effect realism and aural satisfaction. In a real auditorium there are many sound bounces from different surfaces at different distances from the source, so that the repetitions heard by any individual come irregularly and at fairly short separations, usually. Nonetheless, the brain seems to be able to separate out the longest repetition periods, which are caused by reflections from the most distant surfaces, and it subconsciously forms an impression from them of the size of the hall, almost regardless of whether the walls are highly reflective and thus cause long total reverberation due to many bounces or are absorptive and cause a fast over-all sound decay because of high energy loss on each bounce. To simulate a large hall, therefore, repetitions must not come too frequently; if they do, the result sounds like a small room with excessively hard walls, like a bathroom.

The reverberation should affect tones and harmonics of all frequencies, but not evenly. In general, tones around the middle of the spectrum naturally have longer *reverberation times* (the time required for a tone to decay by 60 db) than bass and high treble. The rate of decay of any tone should be approximately linear in decibel terms. And ideally, the reverberation time should be adjustable, as though absorbing draperies could be placed selectively over the hard auditorium walls for different kinds of music.

There are five kinds of artificial reverberators in present use. Normal organ sustain, with a rather short decay time, is one of them, and a control producing this is sometimes marked "Reverb." While the effect is better than nothing in many cases, the ear knows that this is not reverberation, because it is a smooth decay and not repetitions, and usually because the curve of amplitude *vs* time is not correct. A second is a large steel plate to which two transducers are attached. One sends acoustic waves through the plate; they are reflected and repeatedly rereflected at the edges, closely approximating the effects of real reverberation in all its complexity. A second transducer picks up the sound from the sheet and passes it on to the amplifier. These plates are not used in organs because of their large size and cost, but some of them are used by recording companies to add reverberation to a performance done in a dead studio, and the result is very realistic.

Hammond Springs

Of the remaining three types which are used in organs, the Hammond springs are the earliest and now the most widespread. Fig. 7-3 shows the better Hammond unit, used on larger Hammond organs. It supports three hanging springs of

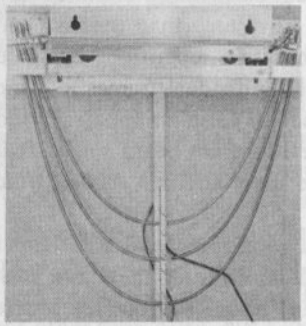


Fig. 7-3—Reverberator used in the larger modern Hammond organs.

different lengths. At one end of each spring is a transducer which produces a twisting force in the spring in response to audio excitation. At the other end of each spring is a second transducer of the same general type which translates spring twists into audio. The whole thing is about 14 inches high, including the hanging springs, 13 inches wide, and 2 inches thick.

When signal from an amplifier is applied to all three driver transducers, the acoustic signal travels along each spring to its other end. The transmission velocity is very greatly less than the transmission of sound in air. Transmission times are $1/22$ second for the shortest spring, $1/17$ for the middle one, and $1/15$ second for

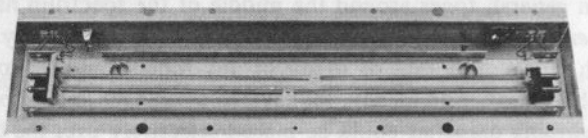


Fig. 7-4—One of the small Hammond spring units sold to organ and equipment makers.

the longest. The pickup transducers are not very compliant and the mechanical impedance is therefore high, so most of the signal is reflected and returns to the sending end. Here, too, the transducer compliance is low, so the sound reflects again. In this way, sound travels repeatedly from end to end of each spring, until it finally loses enough energy to be below audibility. Once each trip it gives up some of its energy to the receiving transducer, which is connected to an amplifier so that the repetitions are heard. Some mechanical damping is applied to the springs so that their energy loss is equalized and limited; otherwise reverberation time would be longer than desired and it would be unequal for the three springs. Actual reverberation time is 2 seconds. The reason for the three springs is to furnish repetitions at complex, rather than regular rates, adding an element of uncertainty which enhances the effect, and reducing the comb-filter effect caused by standing waves.

The organ signal is sent through two parallel channels, both involving voltage amplifiers and one including the spring system. A volume control in the reverberation channel allows setting the total reverberation effect.

A second Hammond spring reverberator is smaller and less elaborate. It is used inside some Hammond models and it is also sold to manufacturers of audio equipment and of other organs. Shown in Fig. 7-4, it has two springs, and they are taut rather than hanging. The metal container measures $16\frac{3}{4} \times 4\frac{3}{8}$ inches over-all. Electronically, it is used in the same way as the other spring set. Each equipment or organ manufacturer using it designs and furnishes his own electronics. The two springs have different delay times, nominally .037 and .029 second, compared to about .045 second for the shortest spring in the more elaborate device. This shorter delay gives the impression of a much smaller room, and the total effect is less desirable for that and other reasons. Three major versions of this unit are available, differing in delay times and frequency response.

Ultrasonic Reverberator

Basic spring reverberators of the Hammond type suffer from important problems. As anyone who has ever twanged a doorspring knows, a spring has its own strong resonances. And as an audio transmission line it has practical upper and lower frequency limits outside of which transmission is poor or nil. Judicious damping can reduce these limitations, but the more damping exists the poorer the over-all transmission, until eventually the spring is useless for reverberation because too much energy is absorbed by the damping. In the Hammond springs a medium has been struck between excessive damping and excessive spectrum-response distortion and ringing, but in the process the upper and lower frequencies are pretty well eliminated. This is even more true in the smaller spring unit.

All of this is hardly surprising when one recalls that the audio band is enormous in terms of ratio between upper and lower frequencies. Even for a band of 30 to 9,000 cps, for example, the ratio is 300 to 1. If the required ratio could be drastically reduced, a spring would be much more likely to function well over the narrower band.

Baldwin has put this thought to work in its Panoramic Tone device by using

ultrasonics. The audio signal is used to modulate an ultrasonic oscillator. The modulated ultrasonic signal is passed through a spring which reflects its signals back and forth until the energy is lost, and each time the signal hits the receiving transducer some of it is passed to a demodulator which removes the ultrasonic carrier and passes the resulting audio repetitions to the system output. Since the carrier is at 20 kc, the maximum band over which the spring has to work extends in our example from 20,000–9,000 or 11,000 to 20,000 + 9,000 or 29,000, the sideband frequencies for the highest (9,000-cps) audio signal. This is a ratio of less than 3 to 1.

Fig. 7-5 is a schematic diagram of the single-channel Panoramic Adapter. Audio input signal at the IN terminal, suitably attenuated, goes to the grid of triode V1, thence to the output. This is the direct signal, which is always present with reverberated signal in these systems.

V2 is the 20-kc oscillator, its inductor the primary of transformer L4. V3 is an ultrasonic amplifier feeding broadly tuned coil L1, from which (through a .01-μf capacitor) ultrasonic signal goes to the input crystal transducer X1 of the

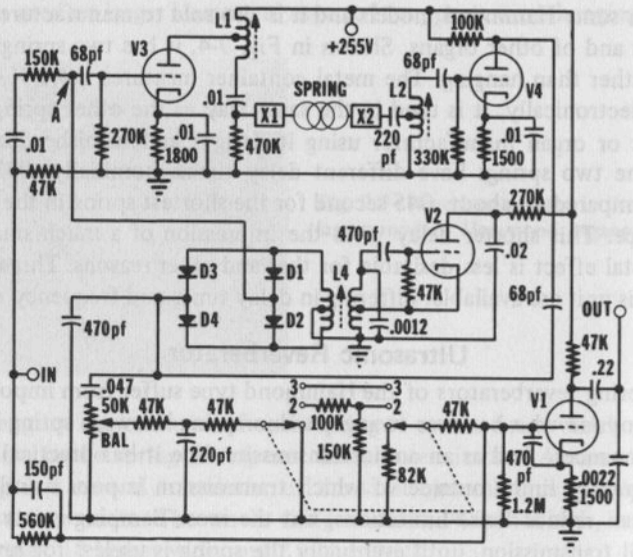


Fig. 7-5–Schematic of a single-channel Baldwin Panoramic Sound system using ultrasonics.

spring delay line. Since ground is at the center of the L4 secondary and both ends are symmetrically above ground, the load which receives the 20-kc generated signal can be assumed to lie between ground and the junction of D1-D2, which have finite and equal resistance even when conducting due to the 20-kc signal.

In the absence of any audio signal at the IN terminal, the two halves of the secondary of L4 and D1-D2 form a bridge circuit, the same potential always being at the junction of D1-D2 and ground, which is the coil centertap. No 20-kc

carrier is therefore placed on the V3 grid, so that lacking an audio signal the carrier is suppressed. When audio is applied, D1 conducts on the negative alternations and D2 on the positive, the conduction being proportional to the instantaneous audio amplitudes. At point A, therefore, there appears both the audio and clipped negative and positive cycles of 20-kc signal corresponding in polarity and amplitude to the audio wave. These 20-kc alternations are clipped, of course, due to the diode action. The 20-kc signals pass through the 68-pf capacitor to the V3 grid and V3's plate passes them to L1, which removes the upper harmonics added by clipping and passes the signal to transducer X1 as a double-sideband suppressed-carrier signal.

X2 is the receiving transducer, connected to another broadly tuned circuit L1 and amplifier V4. From the V4 plate the signal goes through a blocking capacitor to diodes D3-D4, where the carrier is reinserted and the signal demodulated. The signal, which now consists of the audio plus the 20-kc carrier, goes to a balance control, which is a preset adjustment of maximum reverberated-signal level compared to direct signal. From here it passes to a 3-position, 2-circuit switch controlled by the organist. The switch in the position shown kills reverberation. In the other two positions it transmits differing amounts of reverberated signal to output tube V1. Between the junction of D3-D4 and the tube grid there are capacitors to ground which filter out the 20-kc carrier signal.

A 2-channel Panoramic unit is also available. It has a single oscillator and L4 which feeds two sets of the remaining circuits, including two springs.

The Schober Reverbatape Unit

A spring reverberator has the great advantage of simplicity and resulting reliability, but most people do not seem to feel that it gives a particularly realistic effect. Springs have their own resonances and sometimes even twang, especially when shocked by application of a sudden tone. At least the existing systems do not reverberate the lower and higher frequencies, and the existing methods of controlling apparent reverberation time are not parallels of the real thing. The worst offender is the small Hammond unit, at least for organs and quality music systems; it is often used today in guitar amplifiers, where its rather odd effect may be good since the popular guitar combos seem to look for out of the way sounds, including deliberate distortion. But any of the three units described is better than nothing if used in judicious moderation, so great is the enhancement afforded by any suggestion of reverberation. The larger Hammond unit and the Baldwin panoramic are quite beneficial to organ sound.

The Schober Reverbatape Unit is a development of my own, begun several years ago in an attempt to make a reverberator with a definitely realistic effect. Thousands of them are out in the world now, on both Schober and other organs, and I think I can unblushingly report that no one has ever seen fit to deny that its effects are more realistic and satisfying than anything else available to organ owners.

The Reverbatape Unit is a small, special-purpose tape recorder-reproducer. The use of tape provides the complete controllability of repetition rate, decay

rates, and reverberation time *vs* frequency which is nearly impossible with springs. The idea is far from new, as tape reverberators have been used for years in professional recording and broadcasting, but the units have all been highly expensive and rather large. The problem was to make a device at a relatively low price but with the kind of reliability needed for use by laymen. If the following description tends to sound rather enthusiastic, remember that Daddy is talking about his baby. Quite frankly, over the five years or so the Unit has been on the market there have been some problems with mechanical reliability. In the latest model, however, cleaned up by Robert C. Avedon, a far better engineer than me, these problems have been thoroughly licked, as more than a year and a half of use in a great many homes has shown.

A Bird's Eye View

There is a continuous loop of tape which constantly passes through the machine, running in turn over a recording head, three playback heads, and an erase head. Program material to which reverberation is to be added is fed to the input of the Unit.

The output of the Unit, which is fed to the usual power amplifier and speaker system (or, depending on the application, to recording or broadcast circuits) contains first the original program material merely passed through the Unit without change of any kind. Second and more important, it contains program material which has been recorded on the tape, then picked up by each of the three playback heads in turn. Since the tape takes time to get from the record head to each playback head (about .11 second from each head to the next), each playback-head signal is a delayed repetition of the original material.

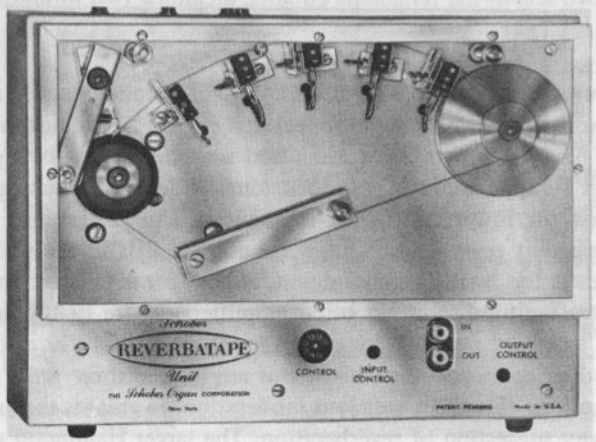


Fig. 7-6—Schober Reverbatape Unit is a small box of heavy steel, tape mechanism on face.

The third playback head, in addition to feeding the third delayed repetition to the output, also feeds a signal back to the record head, so that the process is repeated for another tape pass across the heads. The result is a series of equally

time-spaced repetitions, each repetition decreasing in amplitude by a calibrated amount from the previous one until finally the signal is too small to be heard. The effect is that of an auditorium where repetitions are obtained by bounces from the walls, the sound losing some energy at each bounce, until nothing audible is left.

The user controls the effective length of the decay, and thus the apparent characteristics of the "auditorium," with a single-knob Reverbatape Control which varies reverberation time (time for a 60-db decay) from zero—no reverberation—to over 6 seconds.

The Reverbatape Unit consists principally of a tape-handling mechanism built on the front face of a heavy steel box, shown in Fig. 7-6. The tape itself is a very special kind. The oxide is dispersed in an unusual binder which has enormous holding power, so that continuous passing over magnetic heads causes very little deterioration. The small amount of oxide rubbed off comes away as a dry powder, which simply falls, rather than as "gunk" which would remain on the head gaps and foul them as with the usual tape.

The tape is in the form of a single continuous 19-inch loop, guaranteed for 500 hours of operation. In laboratory tests single loops have been run for over 1,000 hours without audible degradation. A standard splice, made with pressure-sensitive splicing tape, is unsatisfactory because the adhesive cold-flows and the splice slowly pulls apart. The tape is formed into loops by a welding process.

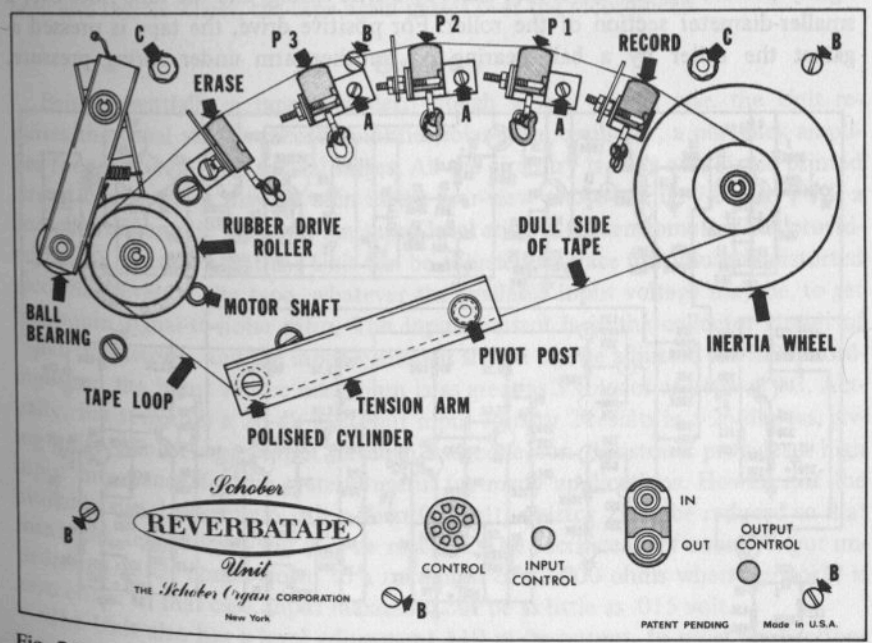


Fig. 7-7—Details of the Reverbatape mechanism. B: Screws holding printed board inside chassis. C: Screws for external head cover. P1, P2, P3: Playback heads. A: Head azimuth adjustments. Entire tension arm swings freely from pivot post to supply tape tension.

Tape Drive

The tape is driven by a small 4-pole motor within the box, whose steel shaft emerges as shown in Fig. 7-7. The rim of a rubber drive roller on an arm is very

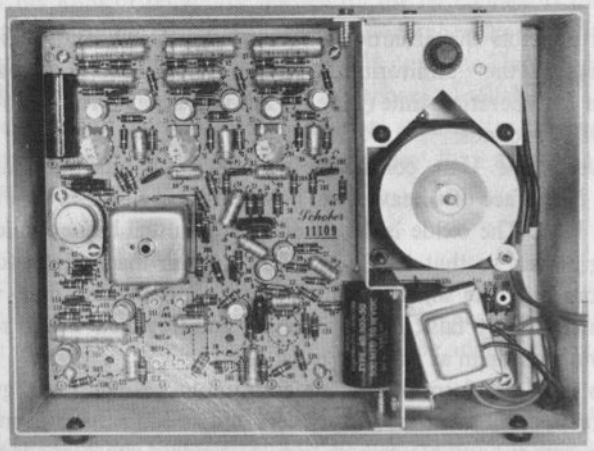


Fig. 7-8–An etched circuit board inside the Unit holds all electronic components.

lightly pressed against the motor shaft by a wire spring, and the tape rides over a smaller-diameter section of the roller. For positive drive, the tape is pressed against the roller by a ball bearing on another arm under spring pressure.

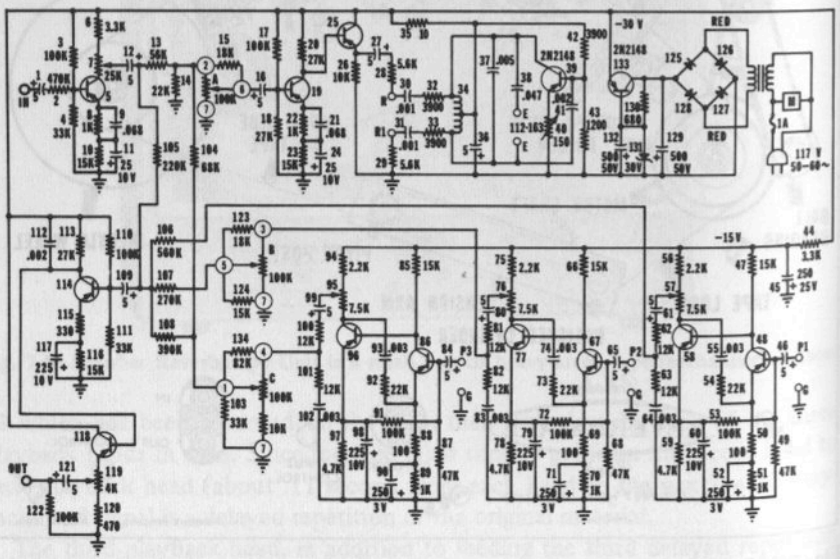


Fig. 7-9–Full schematic of the Schober Reverbatape Unit. The latest bias-erase oscillator circuit is slightly different from the one shown.

When the machine is not running, the pressure of the rubber roller rim against the motor shaft is light enough so that no permanent indentation is made in the rubber by the shaft. However, when the machine starts, the fact that the roller is pulling the tape against some resistance tends to snub the roller against the motor shaft. The roller, in effect, pulls *itself* up tight against the shaft. Also, as the motor shaft drives the rubber roller, a force is created which tends to swing the arm on which the roller is mounted *toward* the motor shaft, further pressing the roller and shaft in firm contact. This is a "self-energizing" system, unique in tape mechanisms to our knowledge, which does away with any need for mechanical disengagement of the two when the Reverbatape Unit is shut off, and considerably simplifies it.

A tension arm near the center of the mechanism is pivoted at one end and has a highly polished cylindrical surface at the free end. The weight of the arm pressing down on the tape provides just the right tension to hold the tape in firm contact with the magnetic heads without pressure pads. After the tape passes beneath the tension arm, it rides over the inertia wheel at upper right. This precisely machined brass wheel runs freely on its shaft and smooths out any variations in speed. It is a flutter filter.

The tape runs over the record head, three playback heads, and an erase head, in that order, at a speed of about 12 inches per second. The separation between heads is such that a given point on the tape takes about 110 milliseconds to go from one head to the next. After passing over the rubber drive roller, tension arm, and inertia wheel, the tape again passes over the record head.

Circuitry

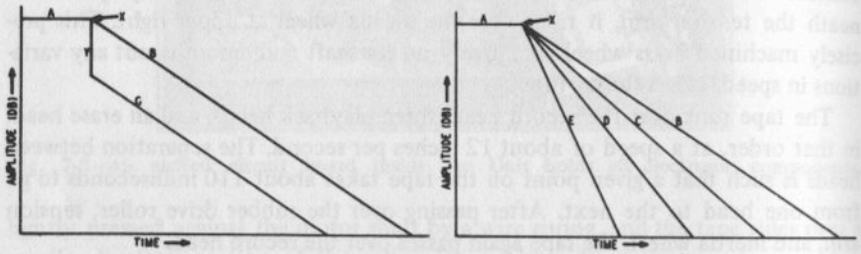
Being essentially a tape recorder, though a very special one, the Unit requires the usual stages–record amplifier, bias-erase oscillator, a playback amplifier for each head–plus special stages. All the circuitry is built on a single printed circuit board which may be seen in the rear-view photo of Fig. 7-8. Fig. 7-9 is a complete schematic diagram. An input level control (potentiometer 7) is provided on the board so that the Unit can be adjusted to place maximum undistorted recording level on the tape, whatever the available input voltage may be, to get maximum signal-to-noise ratio. The input control is in the collector circuit of input transistor 5, and the input system as shown can be adjusted for full recording when the input source maximum is as great as 3 volts or as little as 0.3. Actually, the stage has a 26-db gain, but input resistor 2 results in a 26-db loss, giving unity gain between input terminal and collector. Resistor 2 provides a high input impedance for the system, useful for many applications. However, if the available signal maximum is less than 0.3 volt, resistor 2 can be reduced so that maximum record level will still be reached. The sacrifice is, of course, input impedance, which comes down to a minimum of 22,000 ohms when resistor 2 is zero ohms. In that case, input maximum can be as little as .015 volt.

The Unit also has a level adjustment 119 at its output. In most applications this is set to that output is the same as input. However, since each control has a 20-db range, the Unit can usually be set to give either a loss or gain if wanted.

The input stage has most of the emitter resistance bypassed (11) sufficiently to keep the response flat to well below the audible range. The 1,000-ohm part (8) of the emitter resistance is bypassed by 9, which is small enough to result in a response at control 7 which begins to rise at 2 kc and reaches +12 db at 10 kc. This preemphasis (which is cancelled in a later stage by a complementary deemphasis) is instrumental in reducing the hiss at the output of the Unit to a very low level.

The signal leaving the input stage is applied simultaneously to the output stages 114-118 and to the recording system. The signal passed to the output stages eventually emerges at the output unaltered in any way. This is called the *direct signal* and is the original sound of the program source.

The signal going to the recording channel is first passed through a voltage divider (13 and 14) before going through control A (one section of the external 3-gang Reverbatape Control) and finally reaching the base of transistor 19, the recording preamplifier. The voltage divider adjusts the signal level to that required at the base of transistor 19 and provides the proper source impedance for control A.



Figs. 7-10 (left) and 7-11 (right)—Graphs show realistic and unrealistic sound decays.

The recording preamplifier stage 19 is virtually identical to the input stage. Capacitor 21 provides preemphasis as in the input stage, but for a different reason—to compensate for the high-frequency loss of the playback heads.

The output of transistor 19 is direct-coupled to the base of transistor 25, the recording amplifier. This stage operates as an emitter follower with the record head (terminals R and R1) connected from emitter to ground through suitable series resistance 28 and 29. The series resistance is sufficient to cause constant audio current at all frequencies up to about 10 kc, dropping off to −3 db at 20 kc.

Each playback head feeds signal to a playback amplifier, transistors 48-58, 67-77, and 86-96. The amplifiers are identical, each having d.c. negative feedback which stabilizes them thoroughly and raises the input impedance so that with the 90-mh heads used (Nortronics) there is no treble loss below 20 kc and the 5-μf input capacitors allow flat response down to 20 cps. Output from each amplifier is taken from a miniature potentiometer (57, 76, 95), provided to allow factory calibration compensating for head production tolerances. (Mechanisms of kit Units are calibrated in a special jig and the preset, marked pots go

into the kit.) Resistor 63 and capacitor 64 and their duplicates in the other two channels provide a slow rolloff amounting to about 6 db at 10 kc, to simulate the reduction in reverberation time in auditoriums, where highs are always absorbed more rapidly by the reflecting surfaces than lows.

The output signal of the first playback amplifier is passed directly to the mixer, transistor 114, through 106. The outputs of the second and third playback amplifiers are passed through reverberation control sections B and C and resistors 107 and 108 to the mixer stage. The direct signal is also mixed into this stage through 105.

The mixer stage is basically the same as the input stage with the exception that high-frequency *deemphasis* takes place here to offset the preemphasis of the input stage. Deemphasis is caused by 112, which shunts the collector resistor at high frequencies.

The output stage of the Unit is emitter follower 118. This stage is directly coupled from the mixer, its stability being controlled by that of the well stabilized mixer. The output impedance of the Unit depends on the setting of the output control 119, varying from about 50 to 1,000 ohms.

Control System

The control methods usually used in artificial reverberation systems—especially the spring systems—are unrealistic. Usually there is a direct channel, through which sound goes without added effect, and a parallel reverberation channel, in which reverberative repetitions are added to the original. These are combined at their outputs. To reduce reverberation time, a control simply reduces the gain of the reverberation channel, so that relatively more direct sound is heard.

Fig. 7-10 shows the result. With full reverberation setting, the direct sound is represented by line A. The sound source ceases at time X. Then the reverberation channel causes a slow decay along line B.

When the reverberation channel is turned down, the direct sound A stops at time X. Then there is an abrupt drop in level (Y), after which the decay occurs at the same rate (C) as before. No real auditorium does this. Instead, the decay always begins smoothly just when the sound source is removed. Fig. 7-11 shows this effect at various total reverberation times. Each decay curve B, C, D, E, is a smooth one beginning exactly at point X, without any abrupt drop after cessation of the source. The decay slopes are, of course, different. To accomplish this necessary bit of realism in the Reverbatape Unit a new kind of attenuation system had to be devised.



Fig. 7-12—Reverbatape Control is a 3-gang potentiometer, can be mounted on wood or metal panels or in available plastic box. Decal is provided for markings.

The Unit is so calibrated that when the external control, Fig. 7-12, is at MAX, the output of the Unit due to the first playback head is about 1 db less than the output due to the direct signal. Output due to the second head is 1 db less than that; and signal due to the third head is still another decibel lower. Output of the third head is also fed back to the record amplifier through resistor 104, Fig. 7-9, and rerecorded, so that this cycle continues—each succeeding repetition 1 db less than the last.

With this control setting, and with the repetitions spaced .11 second apart, it takes 60 repetitions or 6.6 seconds for an attenuation of 60 db, and this is the maximum reverberation time of the device.

Faster music or music requiring greater clarity requires shorter reverberation times. These are obtained when the control is turned counterclockwise, by increasing the attenuation between played-back repetitions. For example, when the setting gives 2 db between repetitions, only 30 repetitions or 3.3 seconds is needed to reduce level by 60 db. The control is continuous from maximum down to zero. The following explanation of how it works requires careful attention to avoid confusion.

Maximum reverberation time is obtained with the sliders of controls A, B, C all the way up. (These controls are ganged on one shaft and available to the organist.) Control A then sends full signal to the recording amplifier 25. Signals from the three playback amplifiers join at the base of transistor 114. With the Control all the way up, the calibration potentiometers 57, 76, and 95 have been so set at the factory that the signal reaching 114 due to head 2 is 1 db less than that due to head 1; and the signal at 114 due to head 3 is 1 db less than that due to head 2. In other words, the outputs due to the three playback heads are 1 db apart.

Let us specify certain terminology. Let us always refer arbitrarily to the level of signal appearing at the Unit's output terminals *due to passage through the direct channel and not from the recording system* as 0 db. With the Control at maximum, the three calibration potentiometers have been so set that output signal due to head 1 is at −1 db, that due to head 2 is at −2, and that from head 3 is at −3 db.

Signal from playback head 3 not only goes through resistor 108 to join the other signals at the base of 114; it also goes through resistor 104 to control A for rerecording or feedback. This signal appears at the control along with the signal from the Unit's input. However, the signal appearing at the control from head 3 is 3 db lower in level than the signal due to the Unit input. This 3-db-less relationship, which is determined by choice of the components, is always present. Since the signal obtained at the base of 5, recorded and played back through head 1, resulted in a system output of −1 db, the *fed-back* signal produces a system output due to head 1 which is 3 db less than that, or −4. This is 1 db less than the output due to the original head 3 signal, and it continues the scheme of continual 1-db-apart repetitions.

We may also consider that, due to the setting of calibration potentiometer 76, the output of head 2 has a permanent 2-db attenuation, and that of head 3, due to the setting of 95, has a permanent 3-db attenuation. Adding 3 db to each of these, for the second round of signals produced as the result of the feedback gives 3 db less than the original −2 for head 2, or −5, and 3 db less than the original −3 for head 3, or −6. Thus the action continues, repetitions at the output always 1 db apart.

So far the three ganged control potentiometers have always been at maximum. Before moving them, we must explain that they have special tapers, so that in any position the attenuation given by control C is always twice that (in decibels) of controls A and B, which are identical. Thus, if the Control is set so that A attenuates its signal 3 db, then control B has 3 db of attenuation and control C attenuates by 6 db.

Let us now reduce the Control setting so that A and B have 1 db of attenuation each—and C has 2 db.

The input signal passes through control A and suffers 1 db of attenuation. The output of head 1 is thus 1 db less than with the Control wide open. Instead of appearing at the Unit output at −1, it appears at −2.

The output of head 2 is also 1 db less than before due to control A. It also suffers another 1 db of attenuation in control B, making a total of 2 db. Therefore, instead of being at −2 as originally, it is now at −4.

The output of head 3 is also 1 db less than before due to control A. It also is attenuated 2 db more by control C, making a total of 3. So instead of appearing at the output at −3, it appears at −6.

You will note, therefore, that at this Control setting, the system outputs due to the three heads are, respectively, −2, −4, and −6 with respect to our standard 0-db direct channel. The reverberation time is reduced, but the decay curve, composed of equal decrements in repetitions, is still linear.

Now let us consider the fed-back signal with this Control setting. Since the signal to the recording head is attenuated 1 db by control A and control C attenuates the head-3 signal by 2 db more, the fed-back signal arrives at control A 3 db lower than it was before. In going to the recording amplifier it passes through control A, for 1 more decibel of extra attenuation. It therefore comes to the record amplifier attenuated by a total of 4 db more than it did when the Control was wide open. When the Control was wide open, the fed-back signal produced a Unit output of −4 due to head 1. Subtracting 4 db from that, the fed-back signal with the new Control setting, due to head 1, is −8 db, which continues the cycle of 2-db decrements as advertised. For a little mental exercise, you can postulate other Control settings and additional feedback cycles; you will find that decrements between repetitions are always equal.

Control rotation does not produce linear increments of reverberation time; there is a special over-all time *vs* rotation taper. The numbers on the Control dial do not indicate anything; they are just reference numbers. As the Control is advanced from 0, reverberation time increases rapidly up to a setting of about 1½ or 2 on the dial. As the Control is advanced beyond about 2, reverberation time increases much more slowly. The No. 2 setting on the dial gives about the minimum reverberation necessary for any effect at all. Thus, virtually unusable re-

verberation times are relegated to a small portion of the total rotation, while those which are the most useful are spread out over the Control range.

Reverbatape Installation

The Reverbatape Unit is essentially a voltage-amplifier stage, usually set for unity gain, with high-impedance input and low-impedance output. It can therefore be inserted in any audio system where these conditions are appropriate, and where the available signal-voltage maximum is something between 0.3 and 3 volts. (As indicated earlier, it can accommodate input signal maximums as low as .015 volt, but for this input the output voltage is still 0.3, and external attenuation may be needed.) A simple installation system, usually involving connectors and socket adapters has been worked out to fit each individual make and model of organ. The Unit is also useful for recording and home music systems. One version, called the RV-3C Reverbatape Tone Cabinet, sold through dealers, includes a power amplifier and speaker system with the Unit in the cabinet. It has six inputs so that as many channels can be reverberated; and the direct channel is removed from the Unit so that direct sound continues to come from the organ's individual channel speakers. A recent development is a 2-channel adapter for the basic Unit. This small printed board, optionally mounted inside the steel box, conveys direct signals from two channels to their individual power amplifiers without mixing them, and injects reverberations of both channels into both audio systems.

After any installation is connected, the Unit's input and output controls, available through holes in the front panel, must be adjusted. The input control is turned up so that the loudest possible signal just fails to overload the record head; and the output control is then adjusted for the desired volume. These adjustments are made by ear.

The Reverbatape Unit is available directly from Schober in kit form, with the mechanism assembled and the three pots precalibrated. It can also be sold by any organ dealer, completely assembled, with installation. The kit produces just the Unit itself, as it appears in Figs. 7-6 and 7-8 (plus the Control at the end of a 6-foot cable). The two dealer versions are the RV-3B—the Unit in a walnut box, the Control on the front, placed on top of the owner's organ—and the RV-3C, the Tone Cabinet setup.

Chapter 8

Tuning And Servicing

By and large, electronic organs are rather hardy animals, and the amount of repair service needed is satisfyingly small, considering the complexity of their design. This happy state has prevailed all along, and now that most modern organs lack those old hot vacuum tubes, with their inherent life limitation, potentially destructive heat, and high operating voltages trying to break down any breakdownable component, service trouble is even less prevalent. Regardless of whether one likes the purely musical performance of particular organs, almost every one of them has been designed with components of highly adequate quality and engineered, both mechanically and electrically, with skill. It would be a fine thing indeed if television receivers performed with anything approaching organ reliability.

Since perfection, however, like infinity, is an unattainable quantity, service technicians must and do exist. Ordinarily, organ service is handled by the dealer's service department; but more and more an independent breed of organ servicemen is arising. They are not numerous, but the work can pay well and can be interesting. Of interest to many of these men is the very fact that it is specialized, requiring knowledge on a subject not often written of in books or technical periodicals. Most radio-TV servicemen will not touch an organ—and usually they should not be allowed to do so!

Tuning

Most organs (spinning-wheel Hammonds are the notable exception) have self controlled electronic oscillators, either as control elements in frequency-divider chains or throughout the instrument. The self controlled oscillator which permanently remains tuned to its nominal frequency with enough accuracy to satisfy good musical standards has not been invented, though organ oscillator stability in general conforms to the highest standards of technical excellency. Inevitably, even though it may take six months to a year, every such organ needs a tuning. While a professional tuner (a piano tuner, for instance) can do the job if properly instructed on what screw to turn,* it is usually a technical man—the organ service technician—who is called on for the job.

The unfortunate part of this basic service job is that it is rarely done until the owner calls for it, and that is not nearly often enough. The human brain has a fantastic ability to tolerate a mistuned organ or piano which has gone out of

*I once had my own very expert piano tuner tune a frequency-divider organ. It took him about 10 minutes, and he expressed the wish that he could get jobs like that every day—at the usual piano tuning fee!

tune gradually over a period of time; many musicians share in this unwelcome ability. The mistuning is not recognized as such, but the instrument seems after a while subtly to lack life and some of the interest it once had. A small, gnawing feeling of puzzled dissatisfaction descends on the owner, but he can't quite put his finger on it.

When the tuner arrives and puts the pitches back where they should be, the owner typically gets the feeling that the instrument is brand new again! Only then does he realize that little by little the chords were getting sourer and sourer. The tuning of organs (and really good pianos) rarely jumps out enough in a reasonable time to be really obvious, so the tuner is rarely called until things are out of hand entirely—or he simply arrives on a set, but inadequate, schedule.

It is certainly in the interest of both organ owners and dealers that each instrument give the greatest possible pleasure, which it will not do unless it is in good tune. And contrary to old wives' tales, the most untalented and uneducated owners are affected by mistuning. They are the worst off, in fact, because they recognize the *cause* of the apparent deterioration least but experience it just the same.

I therefore strongly suggest that both dealers and independent servicemen who have an existing relationship with a customer make periodic tuning visits without waiting to be called. The tuning itself can take as little as 10 minutes if one of the devices I shall describe in a moment is used, and the fee can be very small or even nonexistent if things are managed cleverly to call on several customers in one vicinity during a couple of hours. The good will engendered by both the technician's obvious desire to keep the customer happy and, many times, the obvious improvement in sound, can be valuable. Tuning is actually so easy that it does not even require a technician. A salesman can do it—and perhaps gently introduce the subject of a larger instrument at the same time!

There are certain organ salesmen who state that the organs they are selling "never need tuning." The grain of truth in this statement (and again we are not talking about instruments which are obviously fixed in tuning) is that almost any organ goes out of tune much less quickly than even the best piano. But other than this, the statement is a bald lie. It is, moreover, an unnecessary one and it is sure to be found out sooner or later. Everyone knows that other common and highly respected instruments must be tuned occasionally, including many people's most prized possession, the automobile. So where's the sting? The correct statement is that the organ should be touched up every eight to 12 months or so, depending on weather and use, and that it takes practically no time and costs very little.

Pitch Accuracy

The requirements for pitch accuracy in any keyboard instrument are very high compared to almost any other frequency-control application. The actual tolerance depends on the individual and the situation. Speaking of single tones, Carl Seashore, in his classic book, "Psychology of Music" (McGraw-Hill), concludes from a long series of tests that the average person can detect a difference

of about 3 cps at 435 cps, which is about .69% or around $1/17$ of a whole tone. More sensitive people can detect an interval of 0.5 cps at the same frequency, about .115%. And in a test made on a group of excellent musicians the best score was 0.1 cps or .023%.

These figures refer to the differences between pairs of single tones. When no such comparison is being made, tolerances are much wider, for few people have *absolute pitch,* the ability to identify a pitch exactly without any reference. In practice, however, two or more tones almost always sound simultaneously, so the test is effectively made. If the tones are nominally identical or octaves of each other, or if they are part of a harmony which is built, as are most, on harmonic ratios (adjusted for the tempered scale), relatively slight errors are very obvious.

Because tolerances build and it is rare that just one note would be in error, frequency tolerance in any instrument should be kept down to .25%. A figure worthwhile shooting for is 0.1%.

Tuning Devices

There are two instruments in common use, the Conn Strobotuner and the Schober Autotuner. Both make it quite unnecessary for a tuner to have any "ear," any training, or any knowledge of music. Both are stroboscopic, very fast, and have the accuracy of secondary frequency standards, depending entirely on the 60-cps power-line frequency, which in almost every part of the U. S. is so nearly on the nose as to make no difference. Both are equally accurate, but the Strobotuner is somewhat faster to use with organs having individual oscillators for every note. It also has the advantage that the power line need not necessarily be used as the standard. If for some reason the organ is to be tuned to a standard other than A-440, any tone known to be of the exact pitch of any one note can be used to calibrate the instrument, after which all its indications will be correct as referred to the desired standard. In return for these benefits the Strobotuner is a great deal more expensive and considerably larger and heavier than the Autotuner.

Three more devices are available but are less commonly used. One is Conn's Stroboconn, a very large, heavy (though still portable), and expensive stroboscopic machine, which was described in the first two editions of this book. It is more suitable for laboratory research than for field use. A second is the Peterson Tuner, a switchable oscillator. The third is any modern digital counter. Counters are never used in the organ industry for organ tuning; it would be like using an elephant gun on a gnat. (However, I know of several organ *amateurs* that use them—builders of Schober Organs who happen to be engineers working in labs where counters are available and can be borrowed overnight.)

It is not possible to use a variable-frequency audio generator of even the highest quality for tuning. The scale markings and resettability are simply not accurate enough. It is just barely possible to use Lissajous patterns on an oscilloscope, with a starting tone from a tuning fork, through a complicated series of frequency ratio comparisons. This is not worthwhile and we shall ignore it.

The Conn Strobotuner, pictured in Fig. 8-1, is housed in a gray metal case about 10 inches long, 7½ deep, and 8 high, with a removable front cover (not shown). It weighs about 14 pounds. List price is around $180. A motor rotates the stroboscope disc visible through the center window at 12 selectable rates of speed. Two neon lamps behind the translucent disc blink at the frequency of any



Fig. 8-1—Conn Strobotuner gives individual indications for seven octaves of tones.

audio signal fed into the instrument. When the repetition rate of the lamp blinks equals the rate at which strobe disc divisions pass a given point, the disc markings appear to stand still. As the disc speed is known, the frequency of the signal can be recognized.

Fig. 8-2 is a reproduction of the markings on the strobe disc. There are seven concentric bands of equally spaced radial marks—2 in the innermost band, 4 in the next, and 8, 16, 32, 64, and 128 in the remaining ones. If the disc rotated at 1 revolution per second, 128 marks in the outer band would pass any point per second. If the neon lamps blinked briefly once per cycle in response to a 128-cps input signal, the outer band would appear to stand still, since each time the lamps blinked a new radial mark would have arrived at exactly the same position occupied $1/128$ second earlier by the previous one. This is, of course, the same standard stroboscopic principle involved in many devices, most familiarly the ordinary phonograph turntable speed checking disc.

The value of stroboscopic systems like this is twofold. First, the eye easily can detect perfect agreement between neon blink rate and rate of passing of the stroboscopic markings, and even the slightest disagreement. Second, the ingredients are cheap and reliable. The motor must, of course, rotate the disc at a perfectly steady speed and be rigorously controlled by the frequency of its driving source. This is easily achieved with an inexpensive synchronous motor. And the

disc markings must be extremely accurately spaced for easy and unmistakable reading. That is the one element of design that requires special techniques, as we discussed for rotary tone generators in Chapter 3. Hand layout is very difficult; engine dividing is most practical. Certain companies specialize in this and can produce a master with all the accuracy required for almost any purpose involving perfectly reentrant patterns. The cost may be a couple of hundred dollars for the master, but pieces can then be produced in quantity for next to nothing by printing, screening, and other common processes.

As with rotary tone generators, it is not possible to take care of all 12 notes in the chromatic scale without having discs running at 12 different speeds. The large Stroboconn indeed does have 12 discs intricately geared together. In the Strobotuner, however, the knob-operated rotary switch below the window controls a special oscillator which provides 12 selectable frequencies. The oscillator signal is amplified and drives the synchronous motor at any of the 12 speeds.

If, for example, the knob is switched to A, the oscillator frequency is such that the motor rotates at 27.5 rps (1650 rpm). Since there are 2 marks in the inner ring of the disc, the lamp frequency at which this band appears to stand still is 2 X 27.5, or 55 cps, the frequency of A in Octave 1 (Table 1-1). The next band has 4 marks, so its indication is 4 X 27.5, or 110 cps, the A an octave higher. In this way, the disc can indicate any A from Octave 1 through Octave 7, at this one motor speed. The same applies to the speed for each of the remaining eleven notes.

It is interesting to note that while a neon lamp does not have a great range of brightness change, it has enough so that harmonic content causes the light to

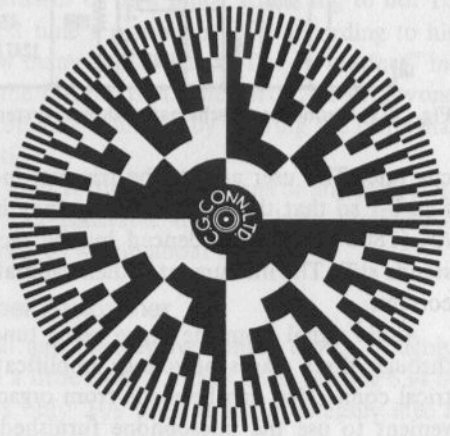


Fig. 8-2—Seven-band Strobotuner disc.

give indications of harmonics as well as fundamentals. On a complex signal fed to the lamps, therefore, the instrument will indicate the fundamental in the correct band, and the higher bands (those nearer the disc perimeter) also seem to stand still. In most cases the harmonics are weaker than the fundamental. The visual indication of this is that the lamps do not go completely off at these fre-

quencies, or do not vary so widely in brightness, so the apparent contrast between the clear and opaque black radials is less. It is not possible to evaluate relative harmonic amplitudes, however.

Fig. 8-3 is the schematic of the Strobotuner. The heart of the oscillator is an inductor with 13 taps going to points on the rotary note selector switch. These taps are very precisely placed in manufacture to obtain the correct ratios between the required 12 frequencies. Twelve of the taps are for the 12 notes. The next to lowest one is for calibration. When the operate-calibrate switch is in the calibrate position, this tap is used for tuning the oscillator and some voltage from the power transformer is fed to the neon lamps so that they blink at a 60-

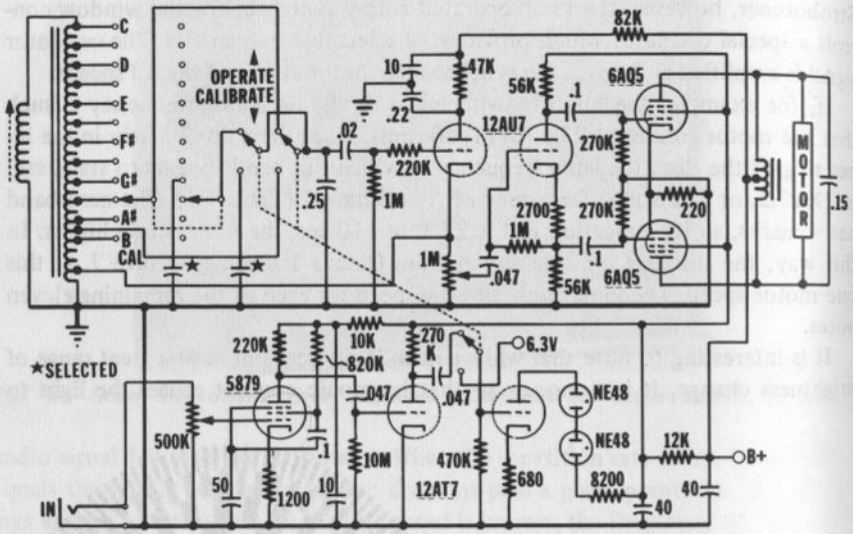


Fig. 8-3—Strobotuner schematic shows system of taps on oscillator tuning inductor.

cps rate. The user adjusts the trimmer inductor with the knob at upper left in Fig. 8-1 so that the oscillator frequency is such as to make the disc rotate at 30 rps (1800 rpm), as evidenced by the fact that the inner disc band of 2 marks stands still. The instrument is then calibrated and all the note frequencies will be correct.

Audio signal from the organ being tuned is fed into the input jack and goes through three stages of voltage amplification to the neon lamps. A direct electrical connection can be made from organ to input, but it is usually more convenient to use the microphone furnished. The amplifier is sensitive enough to make for good indications even with low sound levels and often in the presence of ambient noise.

The knob which adjusts the trimmer inductor has a pointer which moves independently of the shaft and is held on the shaft by friction. After the instrument is calibrated, the pointer can be moved with the end of a finger until it points to zero. Thereafter the knob can be turned, taking the pointer with it, to

indicate deviations from nominal frequency in *cents* as marked on the panel. A cent is 1/100 of a semitone. Even though the markings are not highly accurate, the cent as a unit is such a small percentage of actual frequency (in the order of .06%) that for almost any frequency measuring purpose the instrument is more accurate than any usual laboratory device but a digital counter. Where fairly precise measurement of random frequencies is desired, a book of tables can be obtained from Conn. Actually printed for the more elaborate Stroboconn, these tables show actual frequencies *vs* deviations in cents from standard tempered scale frequencies.

It makes sense that Conn should have produced the Strobotuner, because Conn organs are of the individual-oscillator type and contain large numbers of oscillators, each of which must be individually tuned. This is time consuming at best. It can be done with tuning forks or by any other method which first tunes one octave, after which remaining octaves can be tuned by zero beating octaves of corresponding notes. Obviously, the job can be done much faster by this instrument, which indicates exact tuning for each individual note over a 7-octave span.

The Strobotuner can also be used for piano tuning, though it has never achieved popularity for this purpose. (Neither has any other tuning device.) The problem in piano tuning is that it is common (and necessary) practice to tune the higher octaves progressively sharp of the middle octave and lower octaves progressively flat. This is justified partially by the fact that the overtones of a struck string are sharp of perfect harmonics, but the *stretching,* as it is called, usually exceeds the amount of sharpness of harmonics and is done simply because pianos sound more brilliant and lively that way. Each tuner (and many professional pianists) has his own standards of how much stretching to do. To use the Strobotuner a tuner must first tune a piano perfectly according to his usual methods. Then he discovers how many cents sharp or flat of "perfect" intonation each octave is and records the information. Thereafter, he or anyone else can tune a piano to this same stretching standard by altering the cents dial for each octave per the noted information.

The instrument can, of course, be used to tune pipe organs or anything else. When very carefully calibrated after a considerable time has been allowed for warmup, the Strobotuner can be accurate to within about 1 cent.

The Schober Autotuner

The Schober Autotuner is a small and inexpensive tuning device, roughly 5 X 7 inches in width and height and a little over 2 inches deep, costing $34 in kit form or $48 assembled. It weighs only 2½ pounds and goes easily into a service kit or a good-sized overcoat pocket.

Although its principle is stroboscopic, it manages to tune all 12 notes of one octave with high accuracy with just one strobe disc rotating at just one constant speed. It does this by measuring, not the absolute frequencies of the notes, but the *differences* in frequency between adjacent notes.

The Autotuner consists functionally of two parts. One is a strobe disc, repro-

duced in Fig. 8-4 (in reduced size). Behind the translucent disc are four NE-2-type neon lamps, and behind them the small Synchron clock motor which

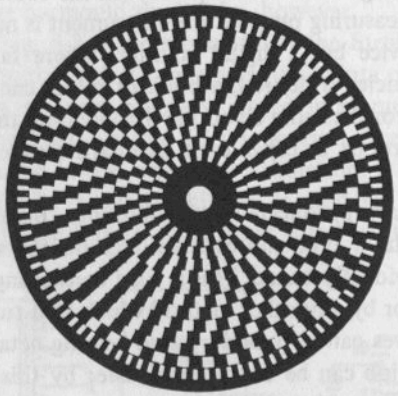


Fig. 8-4—Autotuner disc has 12 bands.

drives the disc. There is also an amplifier which supplies signal to the neon lamps. Fig. 8-5 shows the aluminum-plate face of the unit. A segment of the disc shows through the window. The 12 concentric bands of strobe marks are in-



Fig. 8-5—Face of Autotuner is aluminum plate with window for viewing strobe disc.

dicated by letters representing the 12 notes of the chromatic scale, engraved on the aluminum and also screened in black on the thin Plexiglas window directly over the viewing area.

Fig. 8-6 is a schematic diagram. The two duo-triode tubes amplify the incoming audio signal from the organ and feed it to the four neon lamps which, in parallel and each with a series resistor, form the plate load of the final stage. There is so much gain that the amplifier is easily overloaded, deliberately designed to produce high intermodulation. If a single tone is fed in, the neon lamps blink at the fundamental rate of the tone. But if two tones are fed in at the same time, the high IM causes production of high-amplitude additive and subtractive beat frequencies. Capacitors C1 and C2 severely attenuate the original tones, which in the beat-measuring mode of operation are all above 390 cps, and the additive beats, which are, of course, higher, while leaving the subtractive beats, the maximum frequency of which is 42 cps, at full amplitude to drive the lamps. The motor is simply connected across the 60-cps line and a simple half-wave power supply with a silicon rectifier provides high voltage (about 180 volts). A micro-

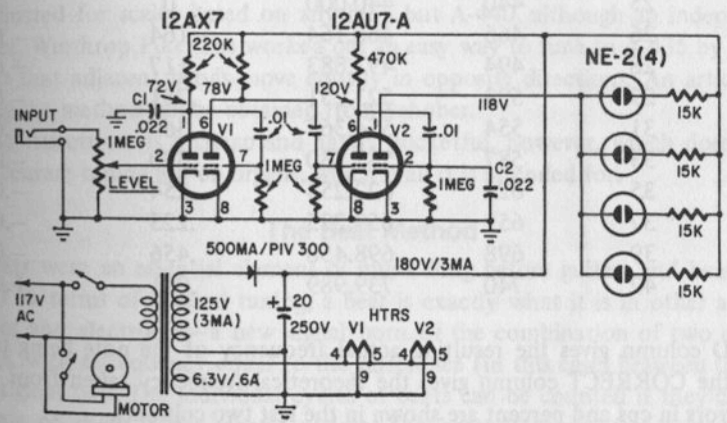


Fig. 8-6—Simple power supply and four-triode amplifier comprise Autotuner circuitry.

phone is provided, but surer indications are obtained with the direct-connection cable, which can be clipped to a convenient point on the organ such as the speaker voice coil terminals.

The disc rotates at a constant 60 rpm, which is 1 revolution per second. The first step in tuning an organ is to play the second G below middle C with an 8′ stop. The nominal frequency of this note is 97.99 cps and there are 98 marks on the outer band of the disc. The note is tuned until the marks stand still, meaning that the G has been tuned to 98 cps. This is an error of a trifle over .01%.

If the organ has individual oscillators, the next step is to tune the G just above middle C to exact zero beat with the G-98 just tuned, which is easily done by ear. If the organ has frequency dividers, the middle G has automatically been tuned.

This middle G at 392 cps is now sounded *simultaneously* with the G# immed-

iately above it. The neon lamps blink at the rate of the *difference* between these two adjacent notes. There are 23 marks in the band labeled G# on the disc. The G# is tuned until the band appears to stand still. The G# frequency is now exactly 23 cps higher than the 392-cps G, or 415 cps. From Table 1-1, the correct G#-4 frequency is 415.305, so the error is .305 cps, which is a shade over .07% of 415.

Table 8-1 shows the remaining operations and their results. The NOTE column shows the note being tuned; it is sounded simultaneously with the note just below it (on the keyboard), which has just been tuned. The MARKS column shows the number of radial markings on the disc band labeled with the note. The

TABLE 8-1: AUTOTUNER ERRORS

NOTE	MARKS	SOUND	CORRECT	ERROR CPS	ERROR %
G	98	98	97.99	.010	+.010
G#	23	415	415.305	.305	−.073
A	25	440	440.000	0	0
A#	26	466	466.164	.164	−.035
B	28	494	493.883	.117	+.024
C	29	523	523.251	.251	−.048
C#	31	554	554.365	.365	−.066
D	33	587	587.330	.330	−.056
D#	35	622	622.254	.254	−.041
E	37	659	659.225	.225	−.034
F	39	698	698.456	.456	−.065
F#	42	740	739.989	.011	+.001

SOUND column gives the resulting actual frequency of the note being tuned, while the CORRECT column gives the theoretical frequency taken from Table 1-1. Errors in cps and percent are shown in the last two columns.

You can deduce at once that since there can only be integral numbers of marks in each band and the number of marks determines the difference between the frequency of one note and that of the next, all notes must end up on integral frequencies. The maximum possible error is therefore one-half cycle, which would occur on a note with a theoretical frequency of something-point-five cps. The final frequency would, of course, end up on the nearest whole number of cps, which would be 0.5 cps higher or lower than nominal. There is actually no note that answers that description, so that the actual error in cps is less than 0.5 as a maximum. In cps it is greatest on the F, but since the F being used is near the high end of the octave, the percentage error, which is what counts to the ear, is not maximum there. Maximum error of slightly over .07% is at the G# and it amounts to a touch over 1 cent, just very slightly more than the Strobotuner at its best calibration. Actually, these errors could have been cut in half by using tones an octave higher and having twice as many markings on the 12 bands. The maximum frequency error in theory would still be 0.5 cps but this

would be only half the percentage of tones having twice the fundamental frequencies. The octave used permits more easily visible markings on the disc, and the point is academic anyway, since the accuracy obtained is much greater than ears require. A common complaint, in fact, is (or was in vacuum-tube organs) that a half hour after tuning a recheck showed the bands moving slightly, obviously due to organ oscillator drift. People had to be assured that the stroboscopic principle gives such indication sensitivity and accuracy that no oscillator is able to maintain for any period of time the immense stability that would be required to show no pattern movement at all!

The Autotuner does not, of course, have the versatility of the Strobotuner. It can only tune one specified octave, so that individual-oscillator organs must thereafter be tuned by zero-beating the notes in remaining octaves to those already tuned. Because of its necessity to derive an intermodulation beat for indication, it requires a fairly positive input signal. This is no problem in organs, where two held notes continue to sound at constant volume, but for piano tuning it is a nuisance because the amplitude fades too fast. The instrument cannot be adjusted for scales based on anything but A-440, although an independent worker, Winthrop Pike, has worked out an easy way to tune to A-435 by adjusting so that adjacent bands move equally in opposite directions. An article describing the method can be obtained from Schober.

The Autotuner is a cheap and handy pocketful, however, which does a fast and accurate tuning job on organs, which is all it is intended for.

The Beat Method

Beats were an essential element of music long before guitars and beards. As meant in terms of musical tuning, a beat is exactly what it is in other areas of physics and electronics—a new signal born of the combination of two original signals, with a frequency equal to the difference (in this case) between those of the original two. The individual cycles of beats can be counted if they come at the rate of about 5 per second or less. The beats between, for instance, two tones 3 cps apart are heard as 3 amplitude pulsations per second; and indeed a voltmeter or oscilloscope (with its horizontal deflection removed) shows these amplitude pulsations clearly.

You will recall from Chapter 1 that in the ordinary tempered scale no notes except octaves have an exact harmonic relationship (integral-number ratio). The third harmonic of any note, for instance, is slightly sharp of the scale note which, without tempering, would have coincided with it. For instance, the exact third harmonic of middle C (261.626 cps) is 784.878. The fundamental of the second G above middle C—the note an octave and a half higher—is at 783.991, which is .887 cps flat of the perfect C harmonic. This lack of whole-number relationships is what makes tuning basically such a nightmare.

It is obvious, however, that a logical way to tune that G (assuming the C to be tuned already) would be to sound both notes simultaneously, adjusting the G first so that it is exactly 3 times the C frequency, then reducing the G pitch until the difference between the G fundamental and the C harmonic is .887 cps. This

sounds impossible but is actually quite easy, though it requires ordinary care.

Step 1 is easily brought off by zero-beating. While sounding both notes, you adjust the G until you begin to hear definite pulsations. As you bring the G closer to the C harmonic, the pulsations slow down. And when they have exactly stopped, you know the C harmonic and the G fundamental agree perfectly.

For step 2, the seemingly impossible task of identifying a beat rate of .887 times per second, you use the second hand of a watch and count the number of beats in *10* seconds! This should, of course, be 8.87, which you can round off to 9 with a total final error in the tuned note of only .003 cps, completely inaudible. Piano and organ tuners have actually tuned this way from time immemorial, though almost always they do not bother counting once they have been trained and have experience; they "feel" the beat rate, much as an organist's feet find the pedals without really looking. However, anyone at all can do the job without this professional "feel" by using a watch and counting.

The system is generally known as *tuning by fifths and fourths.* In our example we specified two notes an octave and a half apart. To make playing both notes possible with the same hand so the other can do the tuning, it is customary to play them in the same octave. Typically, in our example, we would play middle C and the G just above it. We would actually be tuning the third harmonic of the C and the second of the G, which amounts to the same thing as before. We could also, if we liked, play middle C and the G just *below* it, in which case we would actually be tuning the *fourth* harmonic of the G. The latter two notes are known as a fourth, since the two keys are four notes apart in the key of the basic note, C. Middle C and the G just above it are separated by five C-scale notes and the interval is therefore called a fifth. Thus the term "fifths and fourths." It all amounts to the same in the end.

Table 8-2 will enable you to tune any organ with no equipment but a watch or clock having a second hand—once you have correctly set the frequency of one note. This you must do preliminarily with a tuning fork or some other standard known to be very accurate. A single tuning fork is within anyone's budget. Harmonicas and pitchpipes are *not accurate.* You can get the 440-cps tone from WWV if you have a shortwave receiver. (They also have a 600-cps tone; don't confuse them.) If you buy a fork for this purpose get an A-440; if you borrow one from a piano tuner it will probably be a middle C-261, which you can also use.

Begin by turning on the organ and letting it stabilize for a half hour or so. Use only 8′ stops for the tuning, preferably one with good fundamental plus some harmonics, such as a diapason. Tune the note corresponding to the fork or standard you have, by listening to the standard and the organ note *at the same time,* tuning the organ note until you hear slow pulsations (the beats) and the pulsations decrease exactly to zero. To check yourself, tune first one way slightly and then the other, stopping right in the center, at zero.

The base note is now tuned. All your subsequent tuning will be done in the middle octave, using the notes from middle C to the B above it, inclusive. Use no other notes, and always use an 8′ stop.

Now go down the table, beginning at the line whose first note in the SOUND column is the one you have tuned. Assuming, for example, that you have tuned the A, hold down *both* the A and E keys. Tune the note in the TUNE column (the E in this case) first for zero beat. Test each side of center to be sure there are beats both ways and you really are at center. Continue to hold both notes.

Now tune the note in the TUNE column in the *flat—lower-frequency*—direction until you hear slow beats. Use the watch to see how many beats you count in 10 seconds. Your trial count will indicate roughly how far you are from your goal, as given in the BEATS column. Just tune the note slightly flatter for more

TABLE 8-2: TUNING BY BEATS

SOUND	TUNE	BEATS IN 10 SECONDS
A-E	E	15
E-B	B	11
B-F#	F#	17
F#-C#	C#	12
C#-G#	G#	10
G#-D#	D#	14
D#-A#	A#	10
A#-F	F	16
F-C	C	12
C-G	G	9
G-D	D	13
D-A	A	10

beats or sharper (back toward the zero-beat point) for fewer. After a few trials you will find the number correct and can go on to the next note.

Next, hold down the two notes in the SOUND column on the next line. The first of them is the note you have just tuned. Go through the procedure again.

If you started at the A, simply go down the table, tuning as you go, including the next to last line. On the last line, however, you will see that the note in the TUNE column is the A—which you tuned at first with your fork. *Do not tune this same note again by using beats.* It is there for checking, which is a necessity since the process is cumulative. In the last line, hold the D and A and count the beats. If you find 10 in 10 seconds, your work has been exactly correct. If not you have accumulated some error.

If this is your first stab at beat tuning, you will probably want to start over and see whether you finish with less error the next time. However, do not expect to come out with exactly 10 beats between the D and A at the end except as a lucky accident. It is almost impossible to be that precise, and it is entirely unnecessary. The percentage of error in the beat rate is astronomically greater than the actual error in frequency of the note tuned.

For example, suppose you find the D-A beat is 5 rather than 10. Since this is the beat rate in 10 seconds, the real error is only 0.5 beat per second. This rep-

resents an error of 0.5 cps at the second harmonic of the A, 880 cps, which is less than .06%. People are easily fooled, as you can see; a 50% error in beat rate means an error of only .06%—a hundredth of a semitone—in actual frequency. And in all probability, this error is not on just one note but is shared between the two, so it is really even less. So don't spend all day trying to get results precisely per the table!

If the organ has frequency dividers it is now entirely tuned. If it has individual oscillators you can tune the rest of the octaves and ranks by zero-beating each note with its counterpart in the tuned octave or another one already tuned.

N. B. If you have started at C, begin with the line C-G after tuning the C. When you reach the bottom line of the table, go right on to the top line and down through the A#-F line in which you tune the F. Use the F-C line for checking, *but not for tuning the C again.*

In the same way, you may start with any note for which you have a fork or standard. Go to the top line after the bottom one. Use the line just above the one you started with for checking only; do not tune by beats the note for which you have the fork.

Other Tuning Methods

One handy way to tune an organ is to use a phonograph record having 12 steady tones on it, each exactly on the frequency of one of the 12 tones in the middle octave. Such a record is available from Schober.* (If you send for it, be sure to specify the *tuning* record as suggested in this book, as they have other records as well.) Using this record requires a turntable of very steady speed, although not necessarily the exactly correct speed. If the speed is more or less than 33⅓ rpm, the entire organ will be slightly sharp or flat of the A-440 standard. The ratios between the notes will still be right (as witness the fact that an ordinary phonograph record does not sound out of tune as played on a somewhat fast or slow turntable) and you will never know the difference unless you try to play with some other instrument that *is* exactly on the standard. The disc, in fact, used on a variable-speed turntable, is an easy way to tune an instrument to a standard above or below the nominal A-440 if you want to do it.

Still another tuning method is to go out and buy a complete set of 12 tuning forks. As a decent set will probably cost you a minimum of around $55, it is not exactly a cheap method, but the case the forks usually come in fits a (large) pocket and you will not need any electricity as you would for the other devices.

Servicing

There are some people who have a special talent for diagnosing trouble in electronic equipment and take a special pride and pleasure in putting an ailing patient back on its feet. (Although a good service technician must know electronic principles and be familiar with what he is working on, the most effective servicemen are not necessarily those with the profoundest theoretical understanding.) I must admit at the outset that I am not one of these repair-oriented people, which is one reason why this book does not pretend to be a course for maintenance people. My immediate instinct when something does not work right is to throw it away and start over with a new one, whether it is a ball-point pen or a yacht (not that I have ever had one). I haven't the patience or sympathy to deal with a sick circuit.

*The tuning record goes automatically to purchasers of Schober Organ Kits, without charge. Others can buy it separately for $1.00.

But since my company deals with people who have constructed organs from kits, I have had much necessity to cure performance problems, both on the bench and by letter at distances of thousands of miles, which is worse. I have discovered the happy fact that organ servicing is easier than many other kinds, because the nature of an organ provides it with a great many built-in testing and self analyzing facilities. In a radio receiver, for instance, signal goes in one end from the antenna and (hopefully) comes out the other end at the speaker. It can go dead for a variety of causes, all of which are located at some indeterminate point between these two ends.

In an organ, on the other hand, the whole thing rarely comes to a grinding halt. Almost always one stop does not work, or all stops of one pitch register on one manual fail. Or the vibrato stops vibrating. Or certain keys are scratchy. And so on. The very nature of the fault usually indicates where the trouble is, and the area can almost always be narrowed down to perhaps three or four components by testing with the organ's own controls to see what works under what conditions.

A few examples will drive the point home. If all the 8′ Great stops are dead, probably the 8′ bus amplifier for the keyboard concerned is responsible—a bad transistor or capacitor. If all the D tones below a certain octave are silent in a frequency-divider organ, the divider for the highest silent note on that generator is at fault—involving two transistors and a few capacitors. (I have found only one bad resistor in all my life, so I ignore them.) No vibrato? The vibrato oscillator is probably not oscillating. All generators out of sync? Power supply regulator or last filter capacitor out of whack so supply's impedance is high, causing generator interaction or higher than normal voltage. When all pedals above a certain one are silent on a monophonic-pedal instrument, the through contact on the highest good pedal is not making. And so on. Even a completely dead organ can only involve the final amplifier stages or the power supply.

Naturally, easy diagnoses like these do require a knowledge of how the organ is put together and works. Imparting that knowledge should be the function of the service manual which, unfortunately, is completely lacking in some cases or consists solely of schematics with inadequate explanations or none at all. Oddly, some companies make a practice of putting out beautiful service manuals (Conn is the outstanding example), while others often do nothing.

Another very useful technique is signal tracing. It can be used by any knowledgeable serviceman having an organ schematic, but its application is best carried out when, as is sometimes the case, the manufacturer provides diagrams and photos showing physical locations of things.

The whole trick is really to *localize* the trouble to a small area. Thereafter it

is usually just a matter of discovering which of a very few components is the culprit. One would think that any decent radio-TV technician could handle an organ job—and indeed he could, if he took the time and trouble to bone up on organs in general and study the manual for the particular model. These men are usually too busy, however, with more customary work, so intervention of a regular serviceman not an organ specialist usually accomplishes little more than eating up some cash. Things are even tougher now that transistors are so common. Before, the man could at least check all the tubes in sight and often fix the trouble just by tube replacement without ever knowing what the tube did.

Since organs differ so much from each other in principles and circuitry (as you will see beginning with the next chapter), there is almost nothing more specific that I can think of to say about servicing. It is a question of knowing the subject of organs and getting to know the specific one you must work on. When you have read and understood what is in this book, you will be on your way!

Chapter 9

Baldwin Organs

The D. H. Baldwin Company, of 1801 Gilbert Avenue in Cincinnati, Ohio, is one of the two principal makers of grand pianos in the United States (the other being Steinway, which makes no organs) and the producer of the first commercially important all-electronic organ, the justly famous Model 5, which in principle still serves as the basic prototype for almost all modern frequency-divider instruments. Dr. Winston Kock was the inventor named in the patents covering the original Baldwin organs; his researches and ingenious designs under the Baldwin aegis were of the highest importance in delineating the technical course of musical instruments which could sound amazingly pipelike without costing very large amounts of money.

Baldwin is an "old line" company in the music business and its design trends have always been in the direction of instruments of musical quality rather than minimum price—though it has a very active line of vertical Acrosonic pianos and today even deals in equipment for rock-and-rollers (including an amplified harpsichord for this purpose!). This musical conservativeness results in organs which, even in lower-priced models, lack some of the jazzy musical (or nonmusical) chromium to be found elsewhere in the trade but present instead large amounts of well engineered and musically desirable resources. Whether or not you happen to like Baldwin's voicing, if you are knowledgeable about these things, is a personal matter; but honest and varied voicing is definitely there, and in the more expensive models, which nowadays go up to the area of $20,000, money has been well spent on such refinements as obtaining the benefits of chorus without relinquishing the technically desirable frequency dividers, amplification of adequate quantity and quality, and similar good things.

It is very true that "Electronic organs . . . have become much more complex since the days of your first book," as Baldwin's Chief Electrical Engineer, M. B. Gregory, said to me in a recent letter. The changes in Baldwin organs, as I see them, however, have not really been very basic. I would rather say that the circuitry has been improved and brought up to date. Details like the use of transistors have made the usual improvement in reliability, plus greater versatility due to conservation of space and money. And certain effects have been added which simply were not thought of in the early days, such as sustain and reverberation. The additional complexities of trick effects more useful to the salesman than to the purchaser have not figured to any great extent in Baldwin organs.

Like all other organ manufacturers, Baldwin makes a large variety of instruments, from spinet models for the home (sold under the general name of Orga-Sonic) to 3-manual classic church-type organs. As is common in the industry,

the largest number of sales dollars comes from the home instruments. In this chapter we shall describe the Orga-Sonic 54A of Fig. 9-1 which, in Mr. Gregory's words, "defines most of the systems in common usage" at Baldwin. To illustrate a much larger Baldwin organ, Fig. 9-2 shows the Model 11, a 2-manual church organ with 37 speaking stops.

The 54A is a spinet organ with 19 speaking stops in four pitch registers–16′, 8′, 4′, and 2′ on the Solo (upper) manual, 8′ and 4′ on the Accompaniment, and



Fig. 9-1–The Baldwin Model 54A spinet organ is described in detail in this chapter.

16′ and 8′ on the pedals. There is one coupler, Accompaniment To Solo 16′, and there are six percussion (actually sustain) voices. Basically this is a single-channel organ, with built-in speakers and provision for an optional external tone cabinet. However, a Leslie speaker (page 124) is normally installed in the console, and when that is done a separate amplification channel is provided for the flute tones. Panoramic Tone, Baldwin's reverberation device (page 129), may also be added, in either a single- or dual-channel version.

A useful peculiarity of the 54A is that only the lower manual has the usual 44-key F-C compass. The Solo manual has 49 keys, C-C, meaning that the lowest

of its octaves is complete. The pitch range of the instrument covers 6 octaves in the main tone generators (plus one note–the low C–making 73 tones) so that the last octave of the upper manual is complete for the 4′ register and breaks back only for the 2′. The lowest 16′ octave is created for the pedals only by the



Fig. 9-2–Baldwin makes several models of intermediate and concert size. This is the 11.

usual "pedal generator," a flipflop. The total compass of the instrument, therefore, is 7 octaves–Octaves 1 through 7 in Table 1-1.

Physically, the 54A is in a console which is Baldwin's particular variant of the spinet case, with speakers in front, under the lower manual support.

Unlike most spinet-type organs, this one has its stop tablets above the upper manual right across the console. This has been a Baldwin habit (but not an invariable one) since the first Orga-Sonic spinets. It is a great convenience for the player, compared to the more usual small-organ arrangement, which has all the tablets on the cheek blocks at the left ends of the keyboards. The tablets are

rocker types, and they are parts of the tone color box, a long chassis which contains all the voicing circuitry. The console is 37½ inches high (without the music rack) and 45¾ wide.

Tone Generators

Fig. 9-3 is a photograph and Fig. 9-4 a schematic diagram of a tone generator, all 12 of which are identical except for one resistor and two capacitor values in the master oscillators. As the photo indicates, there are two generators to a printed board, and thus six generator boards for the organ. There is also an accessory board containing flipflops for the low C and the pedal 16′ tones, and a vibrato generator.

All Baldwin organs are of the frequency-divider type, and the transistor models use one of the variants of the common flipflop for division. All the dividers are the same, so we have shown in Fig. 9-4 only the first of them, with boxes representing the rest. The six outputs are shown as numbers in circles, the numbers corresponding to the octaves and frequencies in Table 1-1. As you can gather from the photo, Baldwin has followed the common economical practice of placing each divider's passive components in a packaged circuit.

The master oscillator is a Hartley, which is fairly usual except for R1 and C1. C2 is the main tuning capacitor, which ranges from 0.1 to .27 μf for the various generators. C1 ranges from .0056 to .015 μf, and R1 is individually chosen for

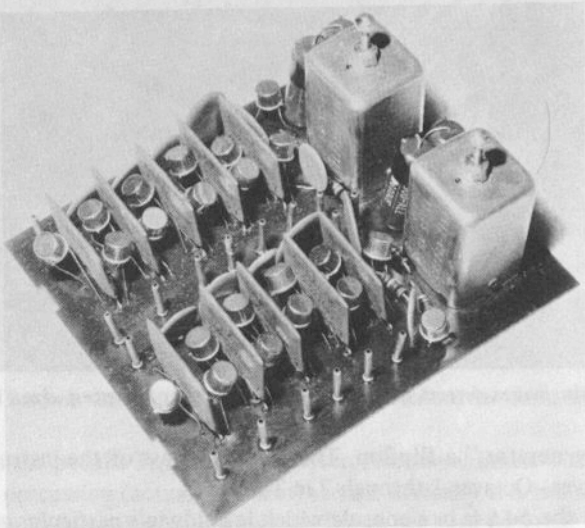


Fig. 9-3—Each printed board holds two tone generators, like this typical example.

each generator so that when the 6.9-cps vibrato signal is applied to the vibrato bus at a peak-to-peak voltage of 6, the Q of all coils is the same and allows a total frequency deviation of 2.5%. The vibrato signal produces frequency change by altering the conduction time of the transistor.

The vibrato generator and control circuit is shown in Fig. 9-5. The phase-shift oscillator TR1 looks normal; R3, one of the phase-shifting components, is selected to give the 6.9-cps frequency.

The one peculiarity is TR2, which is a method of increasing the loop gain of the oscillator while keeping the source impedance for the phase-shift circuit (the

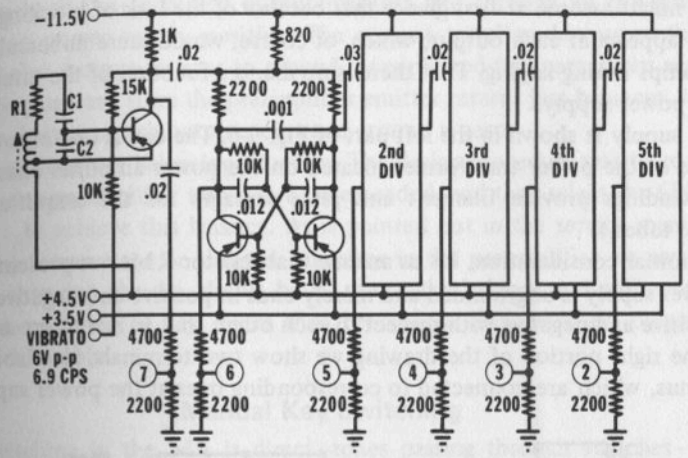


Fig. 9-4—Generator schematic. Dividers in blocks are identical to those diagrammed fully.

common collector resistor) low. Signal applied to the TR1 base appears at its collector in opposite phase. It also appears at the TR1 emitter in the same phase and goes from the TR2 collector to the common collector resistor in the same phase as signal going right through TR1. The outputs of the two collectors reinforce. R1 is selected for a reading of −5 volts at point A.

Signal from point A is also fed to emitter follower TR3, from whose emitter it passes through R2 and the 100-μf capacitor to the two vibrato-tablet switches.

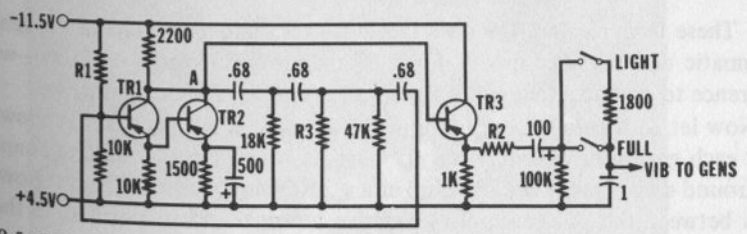


Fig. 9-5—Vibrato oscillator and emitter follower supply signal to master oscillators.

R2 is selected so that 6 volts peak-to-peak appears on the vibrato bus, which is shown in Fig. 9-4, when the Full switch is closed. When the Light switch is closed instead, 1800 ohms is placed between the vibrato source and the vibrato bus, reducing the vibrato signal applied to the master oscillators and consequently their frequency deviation.

Power Supply

Looking at the generator diagram, Fig. 9-4, notice the 4700-ohm and 2200-ohm resistors associated with each of the six outputs. These resistors serve to provide some isolation and to standardize the load on each divider collector when each tone is used with a number of key switches for various pitch registers on both manuals.

One might assume at first glance that because of the lack of blocking capacitors d.c. appears at each output, which, of course, would cause unbearable clicks and thumps during keying. That there is no d.c. is a function of the unusual low-voltage power supply.

This supply is shown in the left part of Fig. 9-6. The coil is one of several secondaries of the power transformer located on the power amplifier chassis. (The other windings provide filament and plate voltages for the amplifier, which contains tubes.)

For initial consideration, let us assume that resistor R1 is not present, so that the power supply is ungrounded and merely ends in positive and negative terminals—positive and negative with respect to each other, that is, not to ground.

In the right portion of the drawing we show two terminals, also labeled plus and minus, which are connected to corresponding ones at the power supply out-

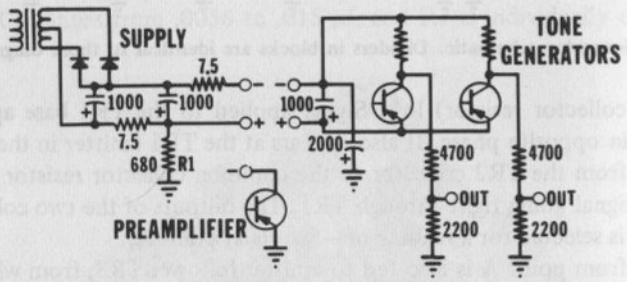


Fig. 9-6—The low-voltage power supply and how it supplies the tone generators.

put. These terminals are the ones labeled −11.5 and +4.5 volts in the generator schematic and the d.c. supply for the generators is exactly as shown—with no reference to ground. (Ignore the figures 11.5 and 4.5 for the moment.)

Now let us mentally concede that R1 actually is in the circuit as shown, and that each generator assembly has six outputs which cause a resistive connection to ground consisting of a 4700-ohm and a 2200-ohm resistor. There is now a d.c. path between the power supply's negative terminal and its positive. In the right part of Fig. 9-6 we have shown two of the 73 generator outputs on the organ, each connected to the collector of a divider or oscillator transistor. Direct current flows downward through each set of output resistors to ground, thence upward through R1 to the positive terminal of the supply. The result is a direct voltage at each output which is negative with respect to ground.

Below the main sections of the drawing we show a transistor which is intended to represent all the transistors present in the organ's preamplifier (in the tone

color box). These transistors derive their B-supply from the negative terminal of the power supply. Their emitter returns are not connected to the positive supply terminal as are those of the generators, but go to ground. Current through the preamplifier transistors travels upward through R1 to power supply positive to complete the circuit.

However, some of the current from the preamplifier emitters also travels upward through all the generator output resistors and through all the generator transistors to power supply positive. The current traveling downward through each 2200-ohm output resistor to ground, as explained two paragraphs ago, and that traveling upward from the preamplifier emitter returns just balances, so that the net d.c. between each output point and ground is zero.

This is obviously a balancing situation. The design procedure was to go ahead with whatever preamplifier transistors were needed, and then select the 680-ohm resistor R1 to achieve this balance. As is pointed out in the service manual, of course, a sick transistor or shorted capacitor in the preamplifier, or any other condition changing the d.c. drawn by it, upsets the balance and will cause key clicks. The presence of key clicks thus becomes a diagnostic aid pointing to preamplifier trouble.

Manual Key Switching

Key switching in the 54A is direct—tones passing through switches—rather than electronic, except for the sustain switching on the upper manual, of which more later. Fig. 9-7 shows the top of a typical key-switch board. It is made of phenolic and the moving switch members are spring wire. When its end is pushed down, a wire comes in contact with a bus (or, to use the word Baldwin has employed since the year 1, a *header*) made of an insulating material impregnated

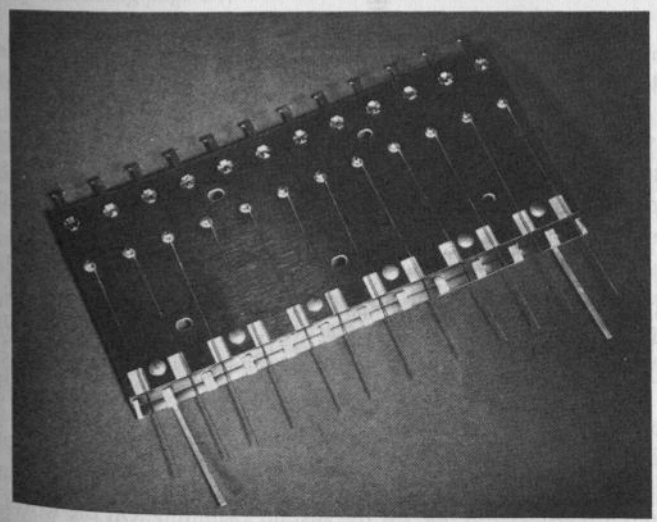


Fig. 9-7—Top view of one-octave key-switch board shows switch fingers and collector bus.

with a conductive composition. The resistance between switch spring and header is appreciable when the spring first comes in contact, and becomes negligible as the key is pressed down all the way and exerts pressure against the header material.

Desirability of gradual keying was discussed in Chapter 4, as were my reservations about this resistance material, which still stand. For many years Baldwin used a type of key switch which was actually a variable resistor—resistance material "painted" on insulating board, traversed by a bendable piece of metal as the key was pressed. After initial difficulties this proved quite reliable and gave a large change in volume as the key was pressed. The effect disturbed some organists, as it was necessary to bottom a key quite definitely to obtain full volume, which is not normal organ technique. The present system, which gives

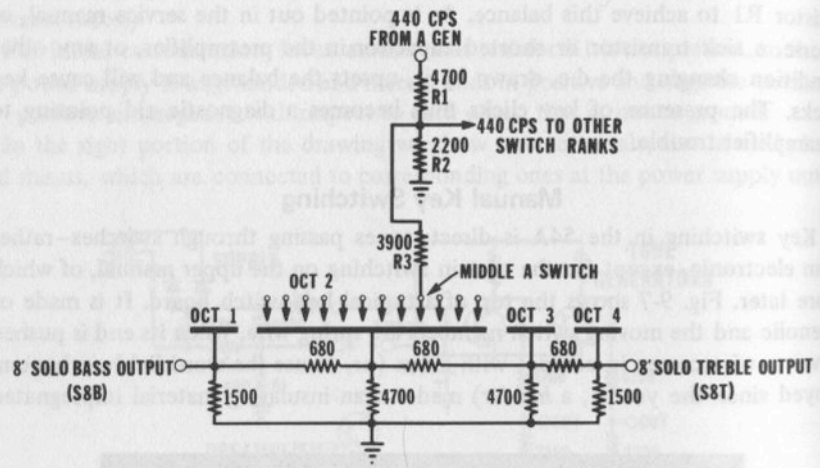


Fig. 9-8—Key-switching scheme derives bass and treble outputs for each pitch register of each manual. This diagram shows the 8-foot register for the Solo manual, which is typical.

much less volume change is welcome for this reason, but because the generated tones are square I see no reason for this monkey business to avoid clicks, so why take a chance on the resistance material? To be honest, I have heard no tales of trouble from Baldwin organs on this account, however.

The other side of the board has a resistor in series with each switch, and the terminals which can be seen at the rear are the connection points for the generated tone inputs. The long flat fingers attached to the headers at front are header connections. The keying boards are made with accommodation for 12 or 13 notes, and are lined up and stacked as needed, one lineup for each pitch register of each manual. On the upper manual, for instance, the 8′ switch rank has a 13-note board at the bass end, followed by 3 12-note boards, making a total of 49 switches. Boards for five such ranks are stacked up, four for the regular 16′, 8′, 4′, and 2′ pitch registers and one for the sustain switching. On the lower manual, two ranks are used, each consisting of four 12-switch boards, with the uppermost four switches simply omitted from the treble board of each rank, making 44.

The diagram of Fig. 9-8 shows how the switching works electrically. We have chosen to show what happens to one note, middle A, on one pitch register, the Solo 8′, with the rest indicated in skeleton form.

In Fig. 9-4 we showed each generator output passing through 4700- and 2200-ohm resistors to ground, with takeoff at the junction. These resistors are actually on resistor boards associated with the generators, but there is just one set of them for each generated signal. They are shown in Fig. 9-8 as R1 and R2, and the signal taken between them can really be considered the signal source for each pitch.

This signal is applied through R3 to a switch finger. Since we are illustrating an 8′ rank of switches, the 440-cps tone is, of course, applied to the switch operated by the middle A key. However, keep in mind that the same signal, from the same R1-R2 source, is also applied through other R3's to switches in each of the other pitch registers. As used in the 4′ switch rank, for instance, it would be connected through an R3 to the switch operated by the A key below middle C. And so on, in the usual way, on both manuals. The series resistors R3 prevent interaction between notes, as we explained in Chapter 4, as long as the keying bus contacted by the switches is terminated to ground.

The resistor network of Fig. 9-8 is the same as that of Fig. 4-7, page 84. As explained in Chapter 4, this system provides two outputs for each pitch register of each manual. One has gradual emphasis of all tones with the treble ones greatest (the treble output in Fig. 9-8) and the other tilts up the bass end. These two outputs are used as needed to offset the rolloffs in low-pass and high-pass voicing filters, as we shall see. It might be worthwhile, before going on, to refresh your memory by rereading page 84.

Manual Voicing

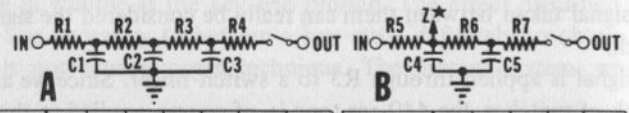
All Baldwin organs are voiced entirely with formant filters according to the theory discussed in detail in Chapter 5. Every attempt is made to obtain good voicing and good scaling for an organ of this size and price, and to this end each filter contains the components that are needed, without any skimping, to get what the designers were after in tone color. For that reason, we shall give information on all the filters, which should be helpful if you voice an organ.

Perhaps I should remark that not everyone fully agrees with Baldwin's voicing and there are a few organs in which formant filters are more elaborate than this. My point is that no really important tonal compromises have been made to save components, as they have to an extent in many other organs. This, then, is a fundamental, complete formant voicing system.

There are six manual key-switch headers in the organ—four for the Solo and two for the Accompaniment. As outlined in Fig. 9-8, each of these can provide two outputs—one at the treble end of the network, with treble tones emphasized, and one at the bass end, with bass tones emphasized. In fact, there are 11 of these key-switch system outputs rather than 12, since the Solo 16′ network

does not have a bass-end output; it is not needed, since the only 16′ Solo stop is a tibia. In discussing and showing the tones fed to the inputs of the filters, we shall save a little space by abbreviating. The 8′ treble-end signal from the Solo manual is S8T (Solo-8′-Treble); the Accompaniment 4′ bass-end signal is A4B (Accompaniment-4′-Bass); and so on.

All the flute stops for the Solo are tibias, and they are present at all four pitch registers available on the Solo. The tibia is actually a stopped flute—pipes which are closed at their upper ends and generate tones whose harmonics are all



SOLO	R1	R2	R3	R4	C1	C2	C3
TIBIA 2′	47K	10K	10K	27K	.001	.022	.01
TIBIA 4′	47K	10K	10K	22K	.001	.0047	.0047
TIBIA 8′	10K	10K	10K	27K	.01	.01	.033
TIBIA 16′	15K	15K	15K	22K	.022	.022	.022

ACC	R5	R6	R7	C4	C5
ORCH FL 4′	56K	10K	22K	.001	.022
FLUTE 8′	47K	10K	22K	.01	.033

★SEE FIG. 9-10

Fig. 9-9—Tibia and flute filters for Solo and Accompaniment manuals, with tabled values.

odd, with very nearly no even harmonics at all. Tibias are usually found only on theatre organs, and in general they are very dull and syrupy with very little (but not zero) harmonic development. (Stopped flutes on classic organs are the counterparts of the theatre tibias, but almost always they have at least slightly brighter tone colors. They are usually provided in some variety, with various names and various brightnesses and loudnesses.)

Section A of Fig. 9-9 shows the configuration of the filters which are used for all the Solo tibias. In each case the input of the filter is connected to the treble-end output of the key-switch rank of a particular register. For instance, the Solo 8′ Tibia filter derives tone from S8T. Since all the keyed tones are square waves derived from the generators, only odd harmonics are present, as desired for tibias. And since the filter has severe low-pass action, it chops off the upper harmonics so that the remaining tone is quite bland.

The values used in the filters for different pitch registers are, as you can see from Fig. 9-9, different. This is partly to set the volume of each output as the designer thinks most appropriate, but to a greater degree each set of values represents an effort to cope with the one problem which arises when a single filter is used to voice the entire pitch range of a manual. If the filter begins its attenuation curve between the fundamental and the first available harmonic (in this case the third) of the incoming tone, it can do a very good job of removing all brilliance. However, if it does this for one of the lowest-pitched tones of a manual, the higher-pitched tones will have fundamentals well down on the filter attenuation curve and will be very much lower in volume. This scaling deficiency can be made up only to a limited degree by the keying network's preemphasis of treble tones. If, on the other hand, the tone for which attenuation begins right above the fundamental is high on the manual, nass tones will transmit too many

harmonics with insufficient attenuation and the sound will be too bright.

The design job is to hit the best compromise between adequate harmonic attenuation and reasonably good scaling. The first method is to design the filter so that its three sections do not have identical turnovers; that is, the three sections are not there just to steepen the total frequency curve. What they actually do is to make the curve progressively steeper as frequency rises. You can see how this is done by noticing that the capacitor values become greater toward the output end of the filter in most cases. Thus the first section begins to attenuate at a fairly high frequency, the next section at a lower frequency, and the last still lower. As a result, the highest harmonics, which are beyond any of the fundamentals, are attenuated the most, while the lowest harmonics of the low notes and the fundamentals of the middle and high ones are attenuated the least.

The fact remains, of course, that a system like this must be a compromise (though it is really surprising how satisfactory a compromise it can be). To reach the ultimate, one cannot use a single filter for more than a certain number of notes, meaning several filters are needed. In this respect the Lowrey and other systems of cutoff filters for groups of notes are superior to what is shown here—though only for the tibias. Only one or two makers provide a separate filter for *every note,* which can give the same ideal results as for a pipe organ, in which each pipe is voiced separately. That method is certainly best (and most expensive), not only for flutes and tibias but also for all other voicing. It is doubtful, though, that the average organ buyer would be justified in spending a great deal of money to obtain better results than the single-filter formant-type system can give. The tonal results are not that obvious.

The Accompaniment flutes are called flutes rather than tibias. Like the tibias, they are derived from square waves and thus have the characteristic hollow tone of stopped pipes, but B in Fig. 9-9 shows that they have only 2-section filters, so they are not quite so bland as the Solo tibias. In both drawings, the switches shown are those operated by the stop tablets. The system is that known as *series switching,* in which the tab switch opens the filter output, in contradistinction to *shunt switching,* as in Schober Organs, where the switch is normally closed and shuts off a voice by shorting a point in the filter to ground.

To this point, we have blithely slid past the fact that the outputs of the generator flipflops are, as usual, square waves, which contain only odd harmonics and impart a characteristic hollow tone to any voices with which they are used, and nothing has been done at the generators, either to provide sawtooth waves with a shaper or to add even harmonics by staircasing, as explained in Chapter 3. Obviously, most stops other than tibias and stopped flutes require even harmonics as well as odd ones. Baldwin provides them by staircasing *at the filter inputs.* The scheme is extremely simple and is quickly illustrated by the diagrams of the filters for the four string-family stops in Fig. 9-10.

Look first at the Solo Salicet 4′ filter. Its input comes, not from a single key-switch output network as did all the flute signals, but from two—Solo 4′ and Solo 2′. The basic pitch of the stop is 4′; the input resistor for this S4T signal is 100,000 ohms. In addition, the filter is fed from the S2T keying output, which

provides tones an octave higher; and the S2T tones are admitted through a 220,000-ohm resistor, about twice the value of the other one. The 2′ tones thus arrive at the 470-pf capacitor at half the amplitude of the 4′, and this is exactly the effect which would have been obtained by mixing the same two tones for each key right at the generators with the more standard staircasing arrangement.

There are two advantages to this system. First, the keying system need provide only one waveform for all purposes—square—and square waves of different

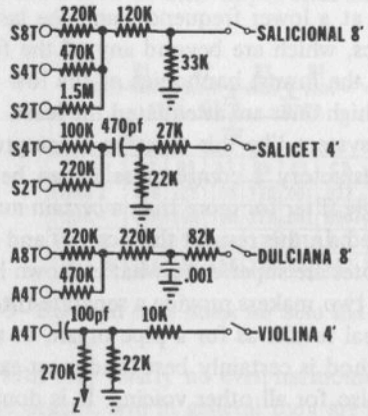


Fig. 9-10—Filters for the string stops.

pitches can be combined at filter inputs to yield staircased tones. Second, it sometimes gives a better result with a particular tone color to mix tones of different octaves in some other proportion than the standard ratio-of-two which yields the most normal distribution and comparative amplitudes of odd and even harmonics. You can see, for instance, that the Solo Salicional 8′ filter is fed by *three* octavely related tones, and that the highest, the 2′ from S2T, has less than the normal staircasing amplitude. (For normal proportions its resistor should be twice that for the 4′ tone, or 1 megohm; instead, it is 1.5 megohms.)

The system also has some disadvantages. Where staircasing is carried out at the generators, only the highest-pitched tone of each generator must remain square, since there is nothing an octave higher to mix with it and add even harmonics. This affects only the highest octave of the highest register. With this Baldwin system, however, the unavoidable square waves occur in an *entire pitch register*—the highest register of each manual. In this organ the damage is minimal since the only 2′ Solo stop is a tibia and the limitation is confined to the Great Violina 4′; a string ought not to sound hollow.

A second possible disadvantage would appear if the designer wished to staircase to the point of filling in as much of the missing even harmonics as possible, as in the organ of Chapter 10. The number of higher-octave tones available to supply missing even harmonics is limited to the number of pitch registers keyed.

On an organ of this type and size. I would guess Baldwin's choice of methods to be about even with generator staircasing; but for larger and more elaborate instruments, staircasing at the generators would be worthwhile even though the

even harmonics would then have to be removed by outphasers or "woodwind circuits" for those stops needing "stopped" tone (or extra key-switch ranks provided).

String tone should, of course, be fairly brilliant, at least compared to flutes and diapasons. In Fig. 9-10 the Salicional 8′, which should be quite buzzy, is simply raw neo-sawtooth tone obtained from staircased square waves with neither bass nor treble rolloff. The Salicet 4′ is a rather piercing string, obtained again from neo-sawtooth waveform but with fundamentals attenuated by the 470-pf series capacitor. The Dulciana is a rather soft accompaniment stop whose buzz or stringiness is greatly reduced by the single-section low-pass filter; it is still much more brilliant than a flute or diapason, but it is not obtrusive because of its relatively low volume.

The Violina 4′ is a combination of two factors. Its principal tone is very stringy because of the emphasis of higher harmonics afforded by the 100-pf input capacitor. However, some body is added separately by taking some signal from the first low-pass filter section of the Orchestral Flute 4′ stop, through 270,000 ohms connected to point Z (Fig. 9-9) on the Orchestral Flute filter.

The manual reed filters are diagrammed in Fig. 9-11. Signal for the Oboe is taken from the bass ends of three pitch registers; the three input capacitors are

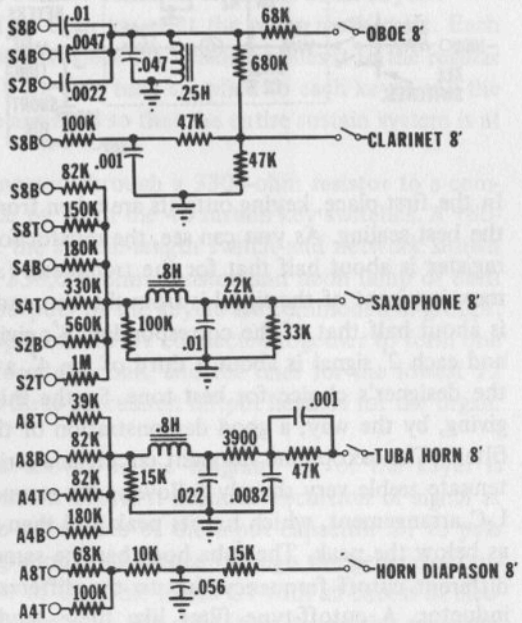


Fig. 9-11—Reed filters of the 54A.

graded in the 1:2:4 ratio which gives standard staircasing. A characteristic of a good oboe voice is that the fundamental is attenuated, which is the reason for input capacitors. Taking the signals from the bass ends of the keying networks helps hold up the volume of the stop at the lower notes. Like all typical reeds,

this voice has a pronounced formant, which is furnished by the L-C resonant circuit at about 1400 cps.

The clarinet filter does not have an L-C circuit of its own but takes a little of the very strong formant of the oboe. To this it adds 8′ square wave of which harmonics above about 1500 cps are attenuated. It is surprising, by the way, how much the sound of a raw square wave suggests clarinet tone, even though it contains no tuned formant and has all the harmonics one can hear, like a strong string. What does it is the hollow sound of the odd-harmonic-only square wave, and the fact that in addition to the harmonics there is a strong fundamental, giving body to the tone. In the filter shown here, the body comes from the somewhat rolled off square wave, the reedy top from the oboe formant, and the result can be a very good approximation of a clarinet, one of the most pleasing and useful organ voices available when well done. (It is less satisfying when rare or medium.)

The saxophone, which has been described as an ill wind which nobody blows good, has everything in it but the kitchen sink; but note the lack of randomness.

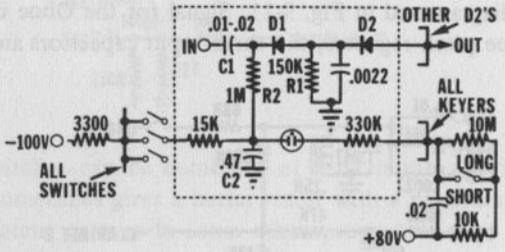


Fig. 9-12—A single sustain keyer, with keying supply, common output, and decay control.

In the first place, keying outputs are taken from both bass and treble ends to get the best scaling. As you can see, the resistor for the bass-end signal in each pitch register is about half that for the treble end (as nearly as RTMA values allow), meaning most of the signal is from the bass end. Second, note that each 4′ signal is about half that of the corresponding 8′, giving correct theoretical staircasing, and each 2′ signal is about a third of the 4′, a little less than correct staircasing, the designer's choice for best tone. So the thing is organized and makes sense, giving, by the way, a good demonstration of the flexibility of staircasing at the filters. The saxophone resonant circuit is a series cutoff type, which tends to attenuate treble very sharply following the resonant rise, as against the more usual L-C arrangement, which has its peak and then descends to about the same value as below the peak. The tuba horn has the same type of L-C circuit, but with a different cutoff frequency due to the different shunt capacitor following the inductor. A cutoff-type filter like these tends to have a quite horny sound (rather than reedy, speaking in orchestral terms), which makes it a good choice for the tuba-type voice. In the saxophone filter the mildly reedy character is preserved by adding 2′ tone, while the low-pass nature of the filter preserves the smooth body that used to make the saxomoan the foundation instrument in dance bands.

The horn diapason is a sort of hybrid stop, somewhat rolled off but retaining lots of harmonic development. It is reasonably loud and, aside from the flutes, it is the principal accompaniment stop. There are no real diapasons in this organ and the lack is not important if only light music is to be played.

Sustain Keying

The six "percussion" voices of the 54A are actually produced by a sustain circuit whose effect is much the same as the usual present sustains throughout the industry. Pressing a key and holding it produces a steady note from one of these voices; when the key is released the tone dies away slowly instead of stopping immediately. Percussive effects are produced by tapping keys in a very short staccato, so that the duration of steady tone is very brief. Unlike most organs with sustain, however, the 54A has a separate set of electronic keyers for the purpose instead of using electronic keying for both sustain and normal voices. Sustain is available on the uppermost 40 keys of the upper manual. In addition to the voicing tablets for the sustain stops, a tablet is provided to select long or short tone decay.

For the 40 keys there are 40 diode keyers, one of which is shown in Fig. 9-12. The elements inside the dashed box are peculiar to one keyer and are repeated 40 times. Items outside the box are common to several or all keyers.

One of 40 tones is connected to each keyer at the point marked IN. Each tone is obtained from the same resistive network which supplies it to the regular key switches (R1 and R2 in Fig. 9-8). The tone supplied to each keyer and the key which turns on that keyer are arranged so that the entire sustain system is at 8′ pitch only.

A source of −100 volts is connected through a 3300-ohm resistor to a common bus to which contact is made by all of the 40 sustain key switches. A +80-volt potential is supplied through the sustain-length switch and network shown to a common line to which the 330,000-ohm resistor and neon lamp of each keyer are connected. The audio outputs of the keyers are commoned in groups; the anodes of the D2's for the top 12 notes are connected together to form one group, those for the middle 16 form a second, and the ones for the lowest 12 notes form a third. There are thus three percussion output headers for the organ. Later we will see where they go.

With the key switch open, as shown in the diagram, D1 of the keyer is blocked by a self biasing arrangement. The first negative excursion of signal at the input causes electrons on the right plate of the input capacitor C1 to pass through D1, and through R1 to ground, and thence through the generator output network back to the left plate of C1. This leaves C1 with an excess of electrons on the left plate and a deficiency on the right; this means the capacitor plate connected to the cathode of D1 is positive with respect to its other plate, which has a d.c. path to ground and thus to the anode of D1. The next alternation of signal, which is positive, cannot discharge C1 because to do so would require the excess electrons to pass from the left plate of C1, through ground, through R1, and through D1 back to the right C1 plate. This they cannot do,

since the diode does not allow conduction in its reverse direction. C1 therefore remains charged positive at the plate connected to the D1 cathode, and the diode remains blocked, so that no further signal can pass through it. To relieve your mind if you are confused, if a rectifier is to rectify—that is, to pass current—as is usual in most applications, a d.c. path must be provided between anode and cathode. In the absence of one—as in this case with the key switch open—the diode simply blocks when a.c. is applied to it.

When the key switch is closed, a negative potential from the −100-volt source is applied to D1's cathode (and there is also established a d.c. path from cathode to ground through the power supply). The diode is now forward biased and generator signal can pass through it and D2 (which is also forward biased by the same keying voltage). The only function of D2 is to prevent d.c. interaction between keyers, preventing transfer of negative keying voltage from a keyed keyer to an unkeyed one.

When the key is pressed, the keying voltage charges C2 to about −100 volts. Ignoring the neon lamp circuit for the moment, when the key is released the negative charge on C2 keeps D1 conducting. As this charge leaks slowly away through R2, D1, and R1, conduction becomes less and less so that tone at the keyer output decays gradually. It ceases entirely when the charge on C2 has dissipated entirely and D1 is again blocked.

The neon-lamp circuit is used to obtain short sustain. When its switch is closed, pressing a key places −100 volts on the left neon electrode; the short-sustain switch has already supplied +80 volts through 10,000 and 330,000 ohms to the right electrode. The neon lamp fires when the key is pressed. As long as the key is held down, the voltage at the top of C2 remains at about −100 volts, keeping the gate open; but as soon as the key is released, C2 is allowed to discharge through the lamp. The resistance in series with the lamp assures that the discharge will not be immediate, but it is a considerably lower-resistance path than through R2, etc., and the discharge is thus much faster. As long as any appreciable negative voltage remains on C2, the lamp has the right applied potentials to remain ignited and act as a fairly low resistance. As soon as the charge on C2 has decayed to a negligible value—at which time the keyer is ready to block again—the lamp goes off and the circuit is ready to be keyed again.

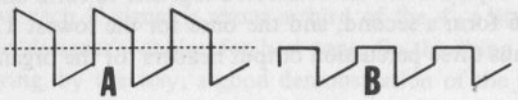


Fig. 9-13—Quasi-sawtooth waves are produced in keyers by differentiating and rectifying squares.

Input capacitor C1 is small enough, in conjunction with the resistance of the load seen by all the keyer outputs, to produce some differentiation—high-pass action. The horizontal parts of the square waves thus are made to slope toward the centerline as in A of Fig. 9-13. Rectification wipes out the positive part of the waveform and thus its symmetry. What is left (B in Fig. 9-13) is no longer symmetrical and now contains even as well as odd harmonics, suiting it for producing, through filtering, normal voices. Staircasing is not possible, of course,

with the percussion voices, because only a single pitch register is keyed. This rather clever waveshaping makes it unnecessary.

Sustain Tone Coloring

Fig. 9-14 shows the tone coloring circuitry for the percussion voices. Preceding the filtering there are three amplifier transistors, one for each of the three

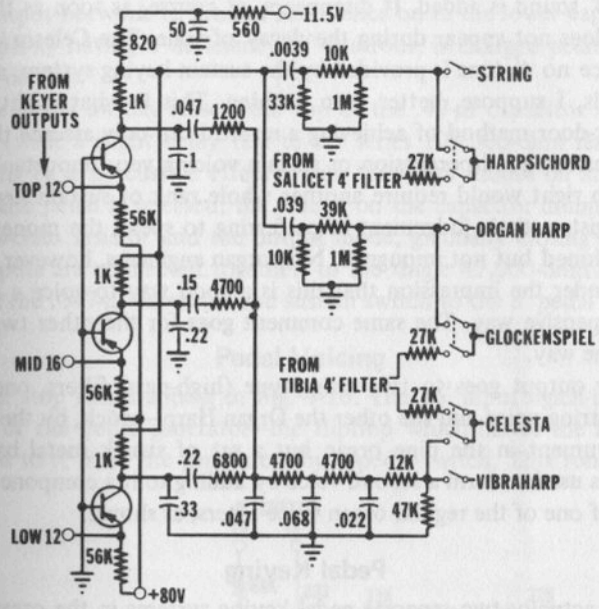


Fig. 9-14—Tone coloring system for sustain voices shows additions from organ stops.

note groups into which the percussion keyer outputs are divided. The amplifiers are basically identical, being grounded-base stages. They are used because they do not reverse phase as the common-emitter configuration does. If the sustain voices were 180° out of phase with the organ voices there would be some cancellation effects between the two groups. They would probably not be used at the same time, since with a sustain-type "percussion" system percussion is possible only with staccato playing. However, as you will see, percussion voices have some regular voicing added automatically.

There are two outputs from the amplifiers. You will notice that all three 1,000-ohm collector load resistors terminate at a single 820-ohm resistor; this is one output. Each amplifier also has a resistor and capacitor from its collector to a common line, which is the second output. Each collector is bypassed to ground to knock off some harmonics, the different capacitors giving optimized effect for each group; this has its effect on both outputs. In addition, the series resistors and capacitors leading to one of the outputs are graded for best aural results.

The last mentioned output goes directly to a low-pass flute-type filter. In the same manner as the regular organ voice filters, this one directly produces the Vibraharp stop. It is also one of two components fed to a 2-pole switch for the Celesta. The other component is tone from the Tibia 4′ filter, taken from the junction of the last resistor and the last capacitor of that filter through the 27,000-ohm resistor shown. The result is that the Celesta voice is exactly the same as the Vibraharp, except that during the brief moment a key is held down some Tibia 4′ sound is added. It disappears, of course, as soon as the key is released, and does not appear during the decay of tone. The Celesta is properly a 4′ voice. Since no 4′ tone is provided by the sustain keying system, adding some in this way is, I suppose, better than nothing. This is what might politely be called a back-door method of achieving a result. I can only assume that Baldwin does not consider good percussion or sustain voicing very important; and since doing the job right would require another whole rank of sustain keyers to provide a 4′ register, their judgement in preferring to spend the money elsewhere can be questioned but not impugned. New organ engineers, however, should not walk away under the impression that this is a good way to voice a celesta; it is only an inexpensive way. The same comment goes for the other two stops that work the same way.

The other output goes to two string-type (high-pass) filters, one producing the sustain String voice and the other the Organ Harp—which, by the way, is not a string instrument in the pipe organ but a set of struck metal bars. Each of these filters is used to form a second voice by adding to it a component from the last section of one of the regular organ voice filters, as shown.

Pedal Keying

There are actually two separate pedal keying systems in the organ, requiring two switches operated by each of the 13 pedals. One system is conventional and the other provides sustain, for the 8′ pedal voices only.

The conventional switches are each connected to the lowest-frequency output of one generator, except for the top C switch, which takes tone from the second

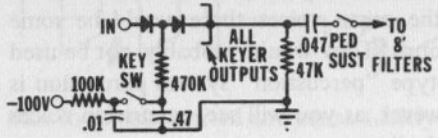


Fig. 9-15—One of pedal sustain keyers.

lowest C. (In this organ, as a matter of fact, C is the highest of the generators, so that a top C is provided to complete the top of the manual 4′ register. The bottom C, a sixth flipflop—the others have only five—is separate, on a board which also contains the vibrato generator and the pedal generator.)

Each switch comes in contact with a header or bus when the pedal is pressed. Tone goes from this bus to the 8′ pedal stops, and also to the input of the pedal generator, a flipflop. The output of the flipflop, reduced by an octave in frequency, is the tone source for the 16′ pedal stops.

Fig. 9-15 shows one of the 13 keyers for the pedal sustain system. The key switch shown is one of the second rank of 13 under the pedals, all of them supplied from the −100-volt source. The principle here is much the same as for the manual sustain gates, but this is simpler.

With the key switch open, the input diode blocks. The first positive excursion of the generator signal connected to the input causes electrons to be drawn from the upper plate of the .47-μf capacitor, passing through the diode and the generator output network to ground and thence on to the lower capacitor plate. The upper plate, having a deficiency of electrons, is charged positive and thus blocks both diodes.

When the pedal switch closes, the top of the .47-μf capacitor is charged to −100 volts (with a slight delay due to the series 100,000-ohm resistor—this is not meant to be a percussive effect). This turns the diodes on and opens the gate. When the pedal is released, the charge on the capacitor drains off through the 470,000-ohm resistor and the output diode, gradually closing the gate. All 13 gate outputs are connected together, to the single 47,000-ohm load resistor, and through the .047-μf capacitor and sustain switch to the 8′ pedal filters.

Pedal Voicing

The pedal stop filters appear in Fig. 9-16. The 16′ input signal is taken from the output of the pedal generator—the flipflop which halves the frequency of any tone fed to it from the generators by a pedal switch. This tone is a square

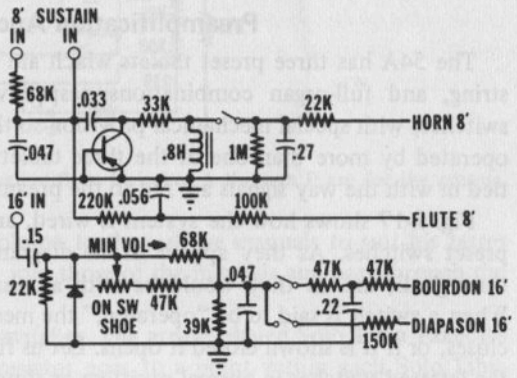


Fig. 9-16—The Pedal stop filters.

wave; it is differentiated by the input capacitor and following shunt resistor, and then rectified by the diode to destroy the symmetry and add even harmonics, on the same theory as was illustrated in Fig. 9-13.

The signal next passes through a potentiometer arrangement using a variable resistor which is part of the 3-gang unit operated by the swell shoe. The main potentiometer on the swell shoe controls the pedal volume as well as that of the manuals, and involves a capacitor to cause less attenuation of bass than of mid-range, to offset the fact that the sensitivity of the ear to bass decreases at lower voluem levels (the familiar Fletcher-Munson effect responsible for loudness controls in audio equipment). To add to this effect of bass compensation, the vari-

able resistor shown gets smaller as the swell shoe is moved in the direction of minimum volume, causing the output of the 16′ pedal stops actually to increase with respect to other organ signals. The total effect at the amplifier output is a decrease in all sounds, but the 16′ pedal tones decrease very much less than the rest so that even at low volume levels a good foundation bass is always audible. The two 16′ stops require no explanation, except that a bourdon should have a symmetrical waveform, which this one does not.

The 8′ input is taken directly from the pedal switch bus and is applied to two stop filters. The flute filter is conventional and is actually a 2-section filter because highs are attenuated before reaching it by the resistor and capacitor at the 8′ input. The horn filter is preceded by another differentiating and rectifying circuit to add even harmonics, this time using a transistor as the diode; presumably either its voltage-resistance curve seems advantageous for the purpose or Baldwin had some transistors they didn't know what else to do with. You should keep in mind, by the way, that this method of adding even harmonics to a wave that started life square can only be used in circuits through which a single tone passes at a time. If the trick were tried in, say, a manual stop filter, the results would be totally disastrous, as the distortion caused by the rectifier would set up intense intermodulation among any two or more tones being heard together. Distortion added to a single tone merely changes its timbre.

The output of the pedal sustain switch of Fig. 9-15 is simply added to the 8′ pedal input when the switch is closed, as Fig. 9-16 indicates.

Preamplification And Presets

The 54A has three preset tablets which are used to obtain prearranged tibia, string, and full-organ combinations, respectively. The system employs eight switches, with special mechanical provision so that most of these switches can be operated by more than one of the three tablets. The preset system is intimately tied in with the way signals are fed to the preamplifiers.

Fig. 9-17 shows how the system is wired, and it also shows five of the eight preset switches. As they appear in the diagram, these five switches, lettered A through E, are as they would be with all three combination tablets up—off. When a switch is said to be "operated," the meaning is that if it is shown open it closes, or if it is shown closed it opens. Let us first describe the connections with the lettered switches in normal positions as shown.

The arrow labeled Solo Tibia is the point at which all the outputs of the tibia filters for the Solo manual are tied together. All these signals join at the 47,000-ohm load resistor, passing through the 22,000-ohm and 6800-ohm series resistors, thence to the input of the flute preamplifier. Most 54A organs are equipped with the Leslie rotating speaker system (which Baldwin calls the *Baldwin-Leslie* in technical material and *theatrical tremolo* in advertising). Only flute-type signals from the manuals are sent to it, and a separate preamplifier is provided. Like the main preamplifier, it is not unconventional enough to warrant full diagramming.

The other Solo signals—joined outputs from the nontibia filters—go through a

similar resistor arrangement, and through 10,000 ohms to the main preamplifier.

The flute signals from the Accompaniment filters join those from the Solo tibias, and similarly Accompaniment strings join Solo strings. For the two types of signal for each manual there is a double-pole tablet operated switch. When open, these switches force the signals to pass through 22,000-ohm resistors, reducing their volume. When closed, they short the resistors and volume increases. These are, in effect, a type of balance switch system adding flexibility to the playing.

All signals from the pedal stop filters join and pass through a similar loud-soft switch, thence to a potentiometer, an internal adjustment with which the owner

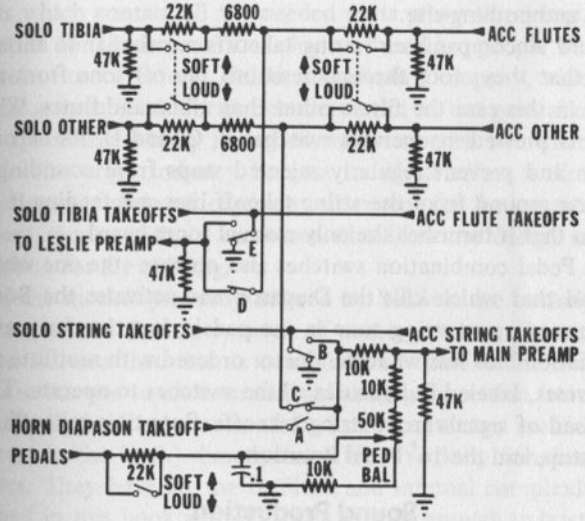


Fig. 9-17—How signals go to the preamplifiers. Switches A through E are for the presets.

can balance the normal pedal volume to that of the manuals to suit his tastes and the room. Pedal signals then join those of the manuals and pass through the main preamplifier.

Now let us look at the preset switches. The arrow labeled Solo Tibia Takeoffs leads to a point from which a resistor goes to a point within each Solo Tibia filter, before the normal tablet switch. The point labeled Accompaniment Flute Takeoffs similarly goes to the joint of two resistors, one leading to a takeoff place within each of the two Accompaniment flute filters. The joined Solo and Accompaniment tibia and flute takeoffs thus contain tone derived from all the flute-type manual filters. This line is normally grounded by switch E and does nothing.

When the preset tablet labeled Tibias is pressed, it operates switches C, D, and E. Switch C opens, so that no tone can go to the main preamplifier from the nonflute filters. Switch D opens, so that no tone from whatever tibia and flute stops may have been selected in the normal way can reach the flute preamplifier.

And it takes the arm of switch E off ground and feeds the tibia and flute takeoffs to the flute preamplifier. The total result is that all settings of the regular stop tablets are cancelled, and only the preselected takeoff tones go to the preamplifier.

There are three additional switches operated by the preset system in the Pedal filters; these switches, not shown, are also operated by the tibia preset. One of them is a separate flute-type stop filter which puts 8′ flute tone into the Pedal filter line shown in Fig. 9-17. The second stops 8′ Pedal tone from going to any regular stop filter. The third removes 16′ Pedal tone from the Diapason filter and feeds it instead into the Bourdon filter—after the Bourdon tablet switch. The result is that the pedals provide only a special 8′ flute tone plus regular Pedal Bourdon tone, and nothing else.

The Solo and Accompaniment string takeoffs are similar to those for flutes and tibias, in that they, too, through resistors, tap off tone from regular stop filters—though in this case the filters other than tibias and flutes. When the preset for strings is pressed, it operates switches B, C, and D. As before, switches C and D open and prevent regularly selected stops from sounding. Switch B moves, removing ground from the string takeoff line and feeding it to the main preamplifier, so that it furnishes the only manual tones heard.

Two of the Pedal combination switches also operate—the one which removes all 8′ stops and that which kills the Diapason and activates the Bourdon. This does not, of course, give string tone in the pedals, but the designers evidently assumed foundation bass was what the doctor ordered with nonflute manuals.

The final preset, labeled Full, causes all the switches to operate. The resulting tone is composed of signals from string takeoffs, flute-tibia takeoffs, the special 8′ Pedal flute stop, and the 16′ Pedal Bourdon.

Sound Production

Baldwin has chosen to use vacuum-tube amplifiers for the 54A. At this point in the state of the art, they are not worth while diagramming here.

The swell-shoe controls are the same for both amplifiers. Each consists of a variable resistor with one end of the element connected to the audio line between a preamplifier triode and the following grid. The other end of the element is unconnected, and the arm goes through a 1-μf capacitor to ground. As the shoe is rotated toward minimum volume, the resistor value decreases and bypasses audio, through the capacitor, to ground. The designer provided the capacitor, rather than a simple arm connection to ground, for bass compensation. With the resistor at maximum value (loudest sound) its resistance is so much greater than the reactance of the capacitor at all frequencies that the capacitor has no effect. As the resistor value decreases with lower volume settings, the reactance becomes more significant; the middle and high frequencies are shunted to ground more than the bass.

The swell shoe operates a gang of three variable resistors. One is for the main amplifier and another for the flute amplifier, operating as above. The third is the special control in the 16′ Pedal line described earlier.

Chapter 10

Schober Organs

A maverick in the organ business is The Schober Organ Corporation, 43 West 61st Street, New York. Since 1955 Schober has been designing and selling electronic organs in kit form, which the purchaser assembles from kits which contain all the needed parts plus voluminous, detailed instructions written for (and more often than not used by) people who have never worked with electronics before. Unlike the Heath Company, which markets a pair of Thomas organs in kit form, Schober is not in the general electronic kit business, but specializes wholly in organs and what goes into them. Schober is also unusual in the organ industry because its whole effort is directed toward kits, and it sells no completed organs at all except for the occasional custom-built instrument requested by a customer who likes the design and lacks the time or inclination for kit building.

All four of the models presently offered by Schober are entirely of its own design. Contrary to the first impression of some people on hearing of "organ kits," they are full-size instruments equivalent to those of more conventional manufacturers in the one to six thousand dollar range as sold in stores, though of course they actually cost very substantially less because they are in kit form and are sold (at least in the U. S.) directly from New York by Schober rather than through dealers. They have all the facilities and internal complexities of other organs described in this book and, if I may be rude enough to boast a little, are among the more pipelike and musically satisfying electronic organs made today. You may take it from the horse's mouth that no technical simplifications or performance compromises were made in any of the designs on account of the kit aspect. The fact that thousands of Schober Organ owners have conspired over the years to make the company's success constantly grow can be attributed to (a) the burgeoning do-it-yourself philosophy so kindly fostered in the electronic field by such companies as Heath and (b) the quality of Schober's instructions, materials, and organ designs. In a sense, the competition with conventional organ manufacturers is a trifle unfair because, due to the kit idea and elimination of a dealer's profit, Schober can afford to spend more money to produce a more elaborate instrument to sell at any given price level. To offset that, of course, only a minority of possible organ purchasers is willing to do kit assembly. As a matter of fact, I want to add a heartfelt tribute to the many individuals and organizations who have had the courage, faith, and enterprise to invest anything from several hundred to a couple of thousand dollars or more in piles of parts and to depend on their own efforts to make the whole thing worthwhile!

Of the four current Schober models, the Theatre Organ shown in Fig. 10-1 is

the newest and its innards will be discussed in this chapter. The Recital Organ of Fig. 10-2 is a full-size formal organ, built to AGO specifications and having 32 speaking stops, 6 couplers, and, among other options, a 13-piston combination action. It is unique among all organs in that any or all of its 32 voices can be changed in a few seconds by plugging in small printed-circuit cards containing different voicing filters and pressing differently engraved stop tablets into place. This is the trademarked Schober Library Of Stops system, which makes possible, in one sense, the largest tonal flexibility ever built into an electronic organ.

Fig. 10-3 shows the Schober Consolette II Organ, an unusual design which



Fig. 10-1—This is the Schober Theatre Organ, assembled from kits, described in this chapter.

combines the 61-key manuals of a large instrument with the handily small physical size of the usual spinet model. The Consolette has 22 speaking stops and a coupler, as well as 17 pedals instead of the usual 13, which put it into the class of a rather deluxe "home-size" instrument.

The smallest Schober Organ is the Spinet of Fig. 10-4, which has the usual 44-key manuals and 13 pedals. It is the only organ in the line which is not fully transistorized; it employs vacuum tubes and neon lamps, with generators like that explained beginning on page 52.

Other Schober products include the Reverbatape Unit (Chapter 7), the Autotuner (Chapter 8), and a 40-watt transistor amplifier designed specifically for organ service but highly suitable for the best of audio systems of any kind. Schober also makes two speaker systems, again designed specifically for its organs. Since Schober Organs are intended to be played through high-quality flat-response systems (many owners use their own hi-fi or stereo equipment, as no power amplifier and speaker system come as integral parts of any of the organs), the speaker systems are also of very high quality for general audio use.

About The Theatre Organ

The Schober Theatre Organ is one manufacturer's answer to the modern en-



Fig. 10-2—Schober Recital Organ is suitable for all sorts of liturgical and classic music.

thusiasm for electronic versions of the cinema instruments of a bygone day. The designers were enthusiastic as well, for a well equipped theatre organ is certainly the most effective instrument on which a single player can render theatre, popular, standard, and other light music. The curved stop board and the "traditional" theatre console design which can be seen in Fig. 10-1 (along with the crazy-legs bench!) are the most obvious points of identification to the eye, but the tonal results are much more vital.

Of the array of 48 stop tablets, 35 are speaking organ voices in pitch registers of 16′, 8′, 4′, 2′, and 1′. Because the organ actually generates eight octaves of

tone, fundamentals from 32.7 to 7902 cps, the 1′ register is complete to within one octave of the top of the upper manual, which is quite unusual in electronic organs and is almost never exceeded even by a pipe organ. Four of the tablets are couplers, another for vibrato on-off, and eight more for percussion voices of an unusually authentic type.

Both manuals have, of course, 61 keys, and there are 25 pedals, the clavier radiating but flat rather than concave. There are controls for placing vibrato on either manual or both, and for vibrato depth; no vibrato appears at all on the



Fig. 10-3–Schober Consolette II is "home size" but has 61-key manuals.

pedal tones (except when the optional Leslie speaker is in use), a traditional Schober feature which follows the pipe-organ philosophy (confirmable by experiment) that pedal vibrato is rather unpleasant. A 10-piston-plus-cancel combination action is available as an option.

All components of the organ except the power supply are built on printed circuit boards, each component (such as each tone generator, all coupler circuits, the stop filters, etc.) on a separate board. This separation makes understanding, building, selling, and servicing easier. Each circuit board and its components are sold as a separate kit when the customer desires it, and some people take advantage of this to spread out the expenditure.

Tone Generators

All Schober Organs belong to the class of frequency-divider instruments with formant-filter tone coloring. There are thus 12 tone generators, each consisting of a master oscillator and six frequency dividers. Seven octavely related tones are derived from each generator, so that 84 tones or seven complete octaves (Octaves 2 through 8, Table 1-1) are provided. As we shall see later, an eighth

octave, the lowest of the instrument, is added by the pedal generator. All the main generators are identical with the exception of two components in the master oscillators.

Fig. 10-5 shows the master oscillator system for one generator. The first transistor is a rather conventional tuned-collector oscillator. The capacitor across



Fig. 10-4–Schober Spinet Organ is only one without transistors. The generators employ neon-lamp divider circuits.

the collector winding of the variable-core tuning transformer differs from one generator to the next to give the right tuning ranges. The emitter resistor is selected for each generator to give the minimum loop gain required for reliable oscillation, so as to obtain maximum frequency stability. The organ power supply is regulated, and the combination of voltage-divider base bias and a good-sized emitter resistor effectively stabilizes the oscillator against differences in transistor characteristics and temperature changes.

It is also no accident that the oscillator can be designed for maximum stability by avoiding any necessity for it to change frequency for vibrato purposes. This was one of the reasons for adopting the phase-shift vibrato system in all

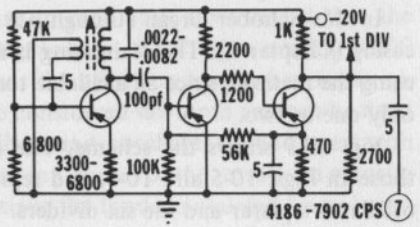


Fig. 10-5–Typical master oscillator circuit.

Schober Organs; if an oscillator's frequency must be "shaken" to create a vibrato, the oscillator can almost never be designed for maximum stability–for if it is, it won't shake! Vibrato is added in the preamplifier, as you will see.

Since the frequency dividers to come will produce square-wave outputs, so

must the tones taken from the master oscillators. The remaining transistors, therefore, are squarer stages, with direct coupling through the 1200-ohm resistor and d.c. feedback through the 56,000-ohm resistor. The excellent square wave appearing at the collector of the third transistor is fed to tone output 7, where it is used for the top octave of tones for the organ, and it is used as the trigger for the first frequency divider.

Fig. 10-6 is a diagram of the first frequency divider. It is a flipflop as described in Chapter 3, and needs no further explanation. Its output is, of course, one octave lower than that shown in Fig. 10-5. All six frequency dividers in each generator are exactly identical, so that we may now assume that each generator has

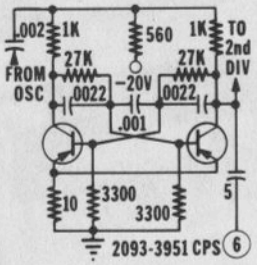


Fig. 10-6—All frequency dividers are identical to this one.

seven square-wave outputs. All the transistors in the generators and throughout the organ except in the percussion keyers are germanium. They are in TO-5 cases, and sockets are used throughout so that a kit constructor who may find the occasional bad transistor will not have to fight three leads and unsolder it. Special low-noise transistors are used in certain marked places, though not in the generators.

The square-wave outputs are not, of course, satisfactory for use throughout the organ, and several courses are open. Schober has chosen to convert the waves to their final neo-sawtooth form before they leave the generator boards, and to provide symmetrical waves later. This results in keying economy, since only one set of tones need be keyed; and in an organ of this size it provides more flexibility than combining pitch registers at the filters as in the Baldwin organ described in the last chapter.

In this Schober Organ (though not in all models) the method chosen is staircasing (Chapter 3). The staircasing is much more complete than in most organs using the method, since all available tones are used for every staircase rather than only one or two.

Fig. 10-7 shows the scheme. The circled numbers at the top correspond to those in Figs. 10-5 and 10-6, and represent the direct outputs from the master oscillator squarer and the six dividers. The numbers down the right represent the finished generator outputs after staircasing.

Notice final output 1. It derives tone from square output 1 through 22,000 ohms, from 2 an octave higher through 47,000 ohms, roughly twice the value, from 3 through 82,000 ohms, about 4 times the value, and so on up to 7. The staircase thus includes all seven available octaves, all in the correct amplitude re-

lationships, and the resulting waveshape at final output 1 is a staircase with such small steps that the audible result is the same as a linear sawtooth. There is a 2200-ohm load resistor from final output 1 (and each of these outputs) to ground. It acts as the shunt leg of the voltage divider for which each of the other resistors is a series leg, proportioning the amplitudes derived from each square source correctly, and also establishes a low impedance for the final generator output. The actual peak-to-peak amplitude of each output is about 2.5 volts.

Final output 2 derives tone from all the higher octaves in just the same way, except that only six sources exist for it instead of seven. And the same procedure is followed for each remaining output, until we finally arrive at 7, which has

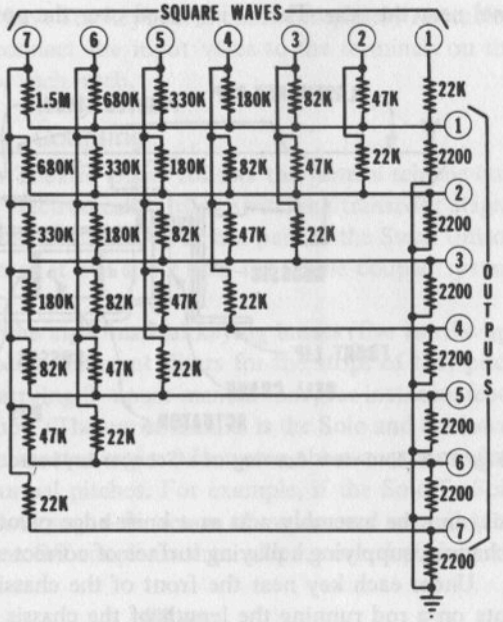


Fig. 10-7—Resistor network adds all available tones to the system of staircasing.

only one source to draw on and thus comes out as a square wave. It is obvious that the actual harmonic content of each output differs, since each is less complete than the last. What actually happens in each case, however, is that the first missing harmonic of even the lowest generator, the C, is at roughly 8,000 cps and the next about twice that, so that the effect on human hearing is approximately nil. The final outputs may be considered sawtooth and can be dealt with in the normal manner by formant filters and outphaser-type squarers with the same final audible results. As explained in Chapter 3, the results are actually better than with real nonlinear sawtooth, since the fundamental and lower harmonics remain at full amplitude with this method.

Manual Keying

The keying system for the manuals is perfectly straightforward, along the lines described in Chapter 4. On the lower manual, for instance, there are three

pitch registers, 4′, 8′, and 16′. There are therefore three moving switch arms under each key, each connected through an isolating resistor to the generator output of the required pitch. Each switch arm contacts a bus which runs the length of the keyboard, and there is one such bus for each pitch register of each manual.

Fig. 10-8 is an end view of the keyboard itself. It is built on a heavy steel chassis formed as shown, with an integral front lip and small angle brackets screwed to it at intervals along the rear. In the organ the whole affair rests on the front lip and rear brackets, on a wood manual support board which is a part of the console.

Each key is a steel channel with an inverted V cut into the sides of the channel near the rear. This V is placed over the upward bend at the rear of the chassis, and the assembly acts as a knife-edge pivot. A plastic cap fits over the steel channel, supplying a playing surface of correct shape and appearance.

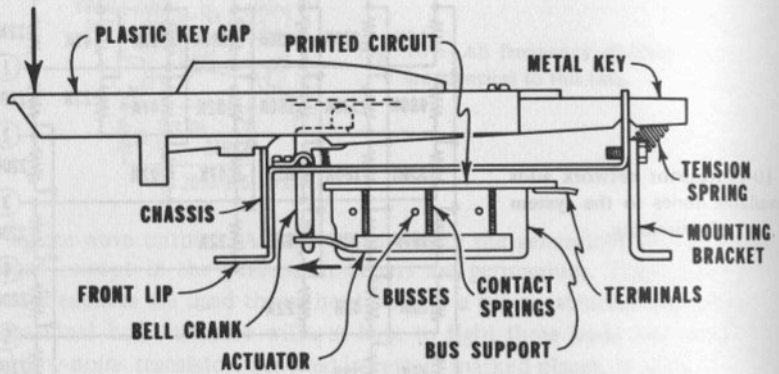


Fig. 10-8–End-view drawing of a Schober keyboard illustrates action and switching parts.

Under each key near the front of the chassis is a plastic bellcrank which pivots on a rod running the length of the chassis and which is in a slot cut in the chassis. When a key is pressed down, the upper section of the bellcrank moves downward and the lower section moves forward, as indicated by the arrow.

The switch mechanisms themselves are on a printed circuit which runs the whole length of the chassis and is supported resiliently by it. Each switch arm is a tightly coiled spring clad with gold. The upper end of each spring is soldered into a hole in the printed board and receives its tone through resistors and copper lines. The lower end of each spring is imbedded in a plastic actuator, the front end of which is attached to the bottom end of the bellcrank. There are three gold busses running the length of the printed board, supported off it by insulators at intervals.

When a key is pressed, the bottom end of the bellcrank pulls the actuator forward. This pulls forward the lower ends of the three contact springs imbedded in it, so that each spring comes into firm contact with one bus. The gold on both springs and busses, and the fact that the springs are pulled right over the busses, assure the best possible electrical results. The fact that the moving contacts are coil springs means that even though each contact takes quite a bit of movement as a whole, any small part of the spring wire bends only a trifle, so that many years of use will not cause metal fatigue or make the contacts take a set.

The upward restoring force for the key is supplied by a husky tension spring at the rear, fastened to a tab punched back out of the chassis. Two types of springs are used, one for the white keys and another, slightly weaker, for the blacks, which are shorter and therefore have less leverage and need weaker tension for the correct playing feel. The upstop for the keys is not shown.

The keyboard illustrated has three busses and three contacts for each key. The one for the upper manual, which carries five pitch registers, has, of course, five busses and five contacts per key. The keyboards come completely assembled so the constructor need only connect the input wires to the terminals on the rear of the printed board, one for each pitch.

Coupling

The coupling on the Schober Theatre Organ follows the general scheme outlined in Chapter 4 but is done electronically, using switching transistor stages. One of these stages is diagrammed in Fig. 10-9. It is a part of the Swell Unison Off coupler, and an explanation of it will show how the whole coupler system works.

Normally tone from each of the eight manual keying busses (five for the upper and three for the lower) goes to formant filters for the stops of that pitch register. That is, the gold bus carrying 8′ upper-manual tone, for instance, feeds tone to the 8′ upper-manual stops. (The upper manual is the Solo and the lower the Accompaniment, as usual in theatre organs.) However, there are times when stops are not wanted at their normal pitches. For example, if the Solo To Solo 4′ coupler is on, all Solo stops will sound an octave above normal as well as at normal pitches. If the Solo Unison Off coupler is then added, normal pitches will disappear, leaving only the raised ones. The Solo Unison Off (which we will call SUO) differs from other couplers only in that it removes something when turned on rather than adding something. All this means is that its tablet switching contacts are reversed–normally closed instead of normally open.

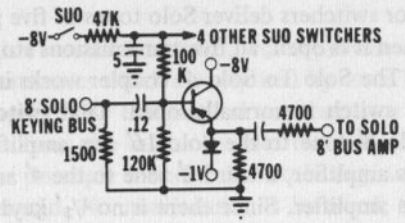


Fig. 10-9–One of 16 coupler switchers.

In Fig. 10-9 the tone input terminal is supplied by the 8′ gold bus of the Solo manual. The 1500-ohm resistor terminates this bus to complete the isolation system explained in Chapter 4. Each keyboard bus has such a terminating resistor.

The 8′ Solo tones then go through a .12-μf d.c. blocking capacitor to the base of a special kind of emitter follower. The stage is normal in that its collector is connected directly to a B-supply of −8 volts, its emitter is grounded through a 4700-ohm resistor, and audio output is taken from it through a blocking capacitor and the 4700-ohm resistor in series.

Two abnormalities are apparent, however. First is the fact that no on bias is supplied to the base except through the SUO tablet switch. The switch is shown open, as it would be with the tablet *on* (down) to help our explanation. With the switch open and no bias applied to the base, the stage is cut off and theoretically none of the tone applied to the base reaches the emitter. In actual fact, however, such a stage leaks some tone through the diode formed of the base and emitter. To kill the leakage the diode is added. With its 1-volt forward bias the diode conducts and forms an audio short circuit from emitter to ground. Since the base-emitter leakage path is of high impedance, this short, which amounts to about 10 ohms, brings down the level of tone at the emitter to over 100 db less than when the stage is conducting normally, reducing leakage to inaudibility.

When the SUO tablet switch closes (SUO coupler off) −8 volts is supplied to the base of the transistor through the 47,000-ohm-5-μf delay network, so that the transistor is turned on gradually without a pop. With current flowing through the transistor, the d.c. at the emitter is greater than 1 volt negative, so that the −1-volt bias on the diode cathode is overcome and the diode no longer conducts. This brings the stage to the condition of an ordinary emitter follower and normal signal appears at the output.

The function of this particular switcher transistor is to pass tone from the 8′ Solo keying bus to the 8′ Solo bus amplifier, which feeds the 8′ Solo stop filters. The SUO coupler system includes four more identical transistor stages, one each for the Solo 16′, 4′, 2′, and 1′ tones. Each of them feeds a corresponding bus amplifier and is separate in almost every way. However, all of them derive their on bias from the junction of the 47,000-ohm resistor and the 5-μf capacitor following the single SUO tablet contact. Thus, when this switch is closed, five transistor switchers deliver Solo tones of five pitch registers to five bus amplifiers, and when it is open, all five transmissions stop.

The Solo To Solo 4′ coupler works in just the same way, except that the tablet switch is normally open. That switch turns on four switchers. One delivers Solo 8′ tone to the Solo 16′ bus amplifier and stops, another 4′ tone to the 8′ bus amplifier, a third 2′ tone to the 4′ amplifier, and the fourth 1′ tone to the 2′ bus amplifier. Since there is no ½′ keying bus, the 1′ stops do not go up an octave but are unaffected. In the same way, the Solo To Solo 16′ coupler sends tone from keying busses an octave lower to each bus amplifier except the 16′, for which no 32′ tone is available. And the Solo To Accompaniment 8′ coupler sends 16′, 8′, and 4′ tones from the Accompaniment manual keying busses to Solo bus amplifiers of corresponding pitch registers. The coupler switcher system is all on one printed circuit. It includes 16 transistors, five for Solo Unison Off, four each for Solo To Solo 4′ and Solo To Solo 16′, and three for Solo To Accompaniment 8′. No switchers are interposed between the three Accompaniment keying busses and their bus amplifiers, since there is no Accompaniment Unison Off. The busses are connected directly to the amplifiers through capacitors, though of course each bus has a terminating resistor.

Bus Amplifiers

The organ contains a bus amplifier for each pitch register of each manual, eight in all. Each of them is like transistor Q1 or Q3 in Fig. 10-10, a simple voltage amplifier followed by an emitter follower to provide a low-impedance line for wiring to the stop filters. Those shown in Fig. 10-10 are for the 8′ and 4′

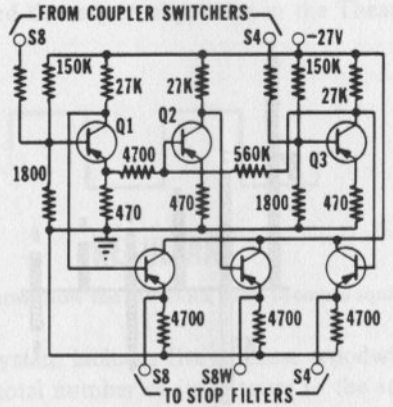


Fig. 10-10—Two of the bus amplifiers and the woodwind circuit which takes signals from both of them.

Solo tones, which come in at the top of the drawing from the coupler switchers (along with tones of any other pitch registers that may be coupled in as we have described), and go out to the stop filters from the terminals at the bottom of the diagram. The emitter followers are direct coupled to the amplifiers, so that the amplifier stabilization applies equally well to the emitter followers. D.c. blocking capacitors are provided at the coupler switcher outputs (Fig. 10-9) and at the stop filters as you will see.

Q2 in Fig. 10-10 is what Schober calls a *woodwind circuit,* which provides the symmetrical waveforms lacking even harmonics which are required by some hollow, woody voices, as discussed in Chapter 5. By this method one transistor can synthesize or reconstruct square waves from the sawtooth ones for each pitch register in which they are required. The principle is known as *outphasing,* and it was originated by Baldwin many years ago. No nonlinearity is involved, so a single transistor stage can be used for any number of simultaneous tones.

Fig. 10-11 can be used to explain how it works. Let us assume we wish to create an 8′ woodwind tone. This means that the original 8′ sawtooth tone which we have available and which has all harmonics, each with amplitude inversely proportional to its ordinal number, must be robbed of all its even harmonics, leaving only the odds.

In Fig. 10-11 we have shown the fundamental and harmonics of the 8′ sawtooth tone as vertical lines. For each harmonic, the height of the line is propor-

tional to its amplitude. (Disregard anything below the horizontal line for the moment.) To make life easy, we have assumed that the fundamental frequency of the 8′ tone is 1 cps, so that all the numbers we shall deal with represent frequencies and also harmonics of the 8′ tone. Note that the second harmonic has half the amplitude of the fundamental, the third harmonic 1/3, the fourth 1/4, etc. We show only six harmonics, but the same story continues in actuality.

Now let us consider a 4′ tone, which has, of course, a fundamental frequency exactly twice that of the 8′ tone—2 cps. Let us also imagine that the 8′ and 4′ tones are added together in a linear amplifier stage with two special provisions. First, the 4′ tone has been reversed in phase with respect to the 8′; and second,

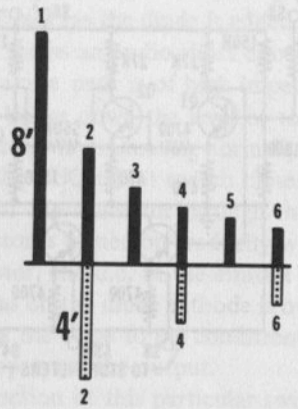


Fig. 10-11—Bar graph shows how outphasing idea gets rid of the even harmonics by cancellation.

its amplitude has been reduced to half that of the 8′ tone. The fundamental and harmonics of the 4′ tone are shown beneath the horizontal line in Fig. 10-11, lined up so that harmonic frequencies of the two tones coincide. The fundamental of the 4′ tone at 2 cps goes in the downward direction because we have arranged for it to be 180° out of phase with the 8′. The 2-cps fundamental is also at half the amplitude of the 8′ tone's 1-cps fundamental, which makes it the same amplitude as the 8′ tone's 2-cps second harmonic.

The second harmonic of the 2-cps fundamental of the 4′ tone is 4 cps. It is at half the amplitude of its fundamental, as is normal with any sawtooth, which makes it the same as the amplitude of the 4-cps fourth harmonic of the 8′ tone. And so on.

You will note now that for each even harmonic of the 8′ tone there is a 4′-tone harmonic of the same frequency and the same amplitude but reversed phase. Naturally, these corresponding frequencies and amplitudes cancel. As a result, the 8′ tone is left with all its odd harmonics intact, but with all its even harmonics wiped out.

The same thing can be shown in a more usual graphical waveform construction by Fig. 10-12, remembering that a symmetrical waveform always lacks even harmonics. The 8′ wave (solid line) is mixed with amplitude-halved, phase-reversed 4′ sawtooth, and the resultant at the right is a square wave.

In Fig. 10-10 transistor Q2 does this outphasing job. Its base is fed 8′ tone from the emitter of the 8′ bus amplifier, so that the collector of Q2 produces 8′ tone in the same phase as the collector of Q1. The base of Q2 is also fed 4′ tone from the *collector* of the 4′ bus amplifier, so that the 4′ tone appearing at the Q2 collector is *opposite* in phase to that on the Q3 collector. The 560,000-ohm and 4700-ohm resistors are so proportioned with regard to the point from which each takes its signal that the 4′ tone applied to Q2 is half the amplitude of the 8′ tone. We have thus satisfied the requirements of what we have discussed; and what appears at the collector of Q2 is an 8′ square wave. The S8W output is therefore of the same fundamental pitch as that at S8, but it lacks even harmonics and is suitable for filters like stopped flutes (actually tibias in the Theatre

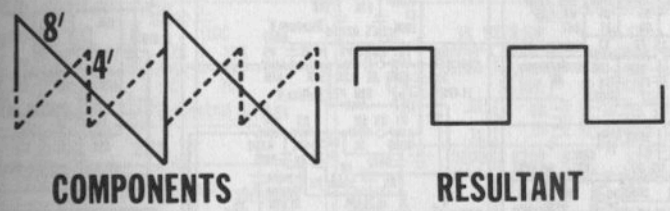


Fig. 10-12—Standard graphical construction shows how the sawtooth wave becomes square.

Organ) and the like. The bus amplifier system includes five of these woodwind circuits at various pitches, so that the total number of its outputs to the stop filters is 13—eight sawtooth outputs for the two manuals, plus five woodwinds.

Theatre Organ Voicing

The Schober Theatre Organ conforms fully to the theories of formant filter voicing as discussed in Chapter 5. It is perhaps the best example in this book of an organ voiced completely in this way, for two reasons. First, there is no "telescoping" of any kind; that is, no part of any filter is used as part of another. Every stop is voiced by a completely separate filter, and no economies have been allowed to affect any filter. Second, each filter has been worked out to obtain the best possible scaling—the most even apparent loudness from one end of the keyboard to the other—obtainable from a system of this type, in addition to good voicing.

I have therefore diagrammed the full manual voicing system in Figs. 10-13 and 10-14. Taking Fig. 10-13 first, the Solo division, keep in mind that the bus amplifiers provide eight outputs for the Solo—sawtooth tones in five pitch registers and woodwind tones at 16′, 8′, and 4′ pitches. These bus amplifier outputs are represented in Fig. 10-13 by the letter S, the pitch register, and, if applicable, the letter W. At the upper left of the diagram, for example, S16W indicates the input point for Solo 16′ woodwind signal. The 5-μf capacitor furnishes isolation for d.c. from the bus amplifier, and the 47,000-ohm resistor grounds the far end of the capacitor to prevent an accumulation of charge. Tone then goes to the Tibia 16′ filter, a 3-section low-pass affair. The terminal labeled BS is the point

from which a wire goes to the stop-tablet switch. When the stop is to be off, this switch grounds the terminal, shorting the filter at this point, a more positive method of shutoff than a series switch, since no leakage of tone through an open switch or the wires leading to it (by capacitance or radiation) is possible. Pushing down the stop tablet removes the short and the stop sounds. Since the Solo has no other 16′ stops requiring woodwind tone, no other filters are fed by it.

Three stops require 16′ sawtooth tone, the Stentorphone, Cello, and Tuba Mirabilis. The S16 input, therefore, after being d.c.-blocked and terminated, is fed to all of them. Each has a terminal for grounding through its stop switch, but there is no interaction on the tone source from these switches because the source

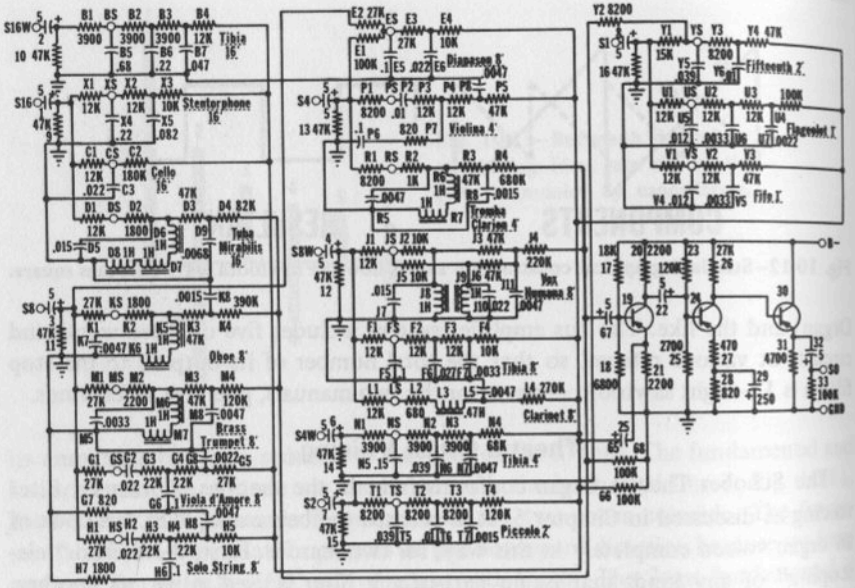


Fig. 10-13–Stop filters and amplifier stages for the Solo division of the Theatre Organ.

is the emitter follower in the bus amplifier system, which can be assumed to be about 50 ohms; the input resistor to each filter is a great many times that, so the tone source cares not at all whether one or more stop tablet terminals are grounded.

The remaining filters are set up in the same way, each fed by appropriate tone of the appropriate pitch register. Diapason stops are fed by tones of *two* registers to supply the desirable emphasized second harmonic (page 108). You will note that most of the reed stops have two inductors in series. The purpose is to get the total series value of inductance without having to stock inductors of several different values. The ones used are wound on ferrite cores and look like tubular capacitors with axial leads; they are conveniently small for their inductance and are more easily mounted in printed boards than other types. The Vox Humana 8′ is notable for the fact that it has two separate tuned circuits (what you see is

not a transformer) because this distinctive tone color has two definite formants.

The outputs of all filters are connected together in two groups–the reeds and flues (nonreeds). The reeds (except for the Clarinet) are fed to the base, and the other stops to the emitter of transistor 19, which is the start of a 2-stage amplifier followed by an emitter follower. The reason for the separation is that the phase shift in the reed stops is different from that in the flues. It was found that

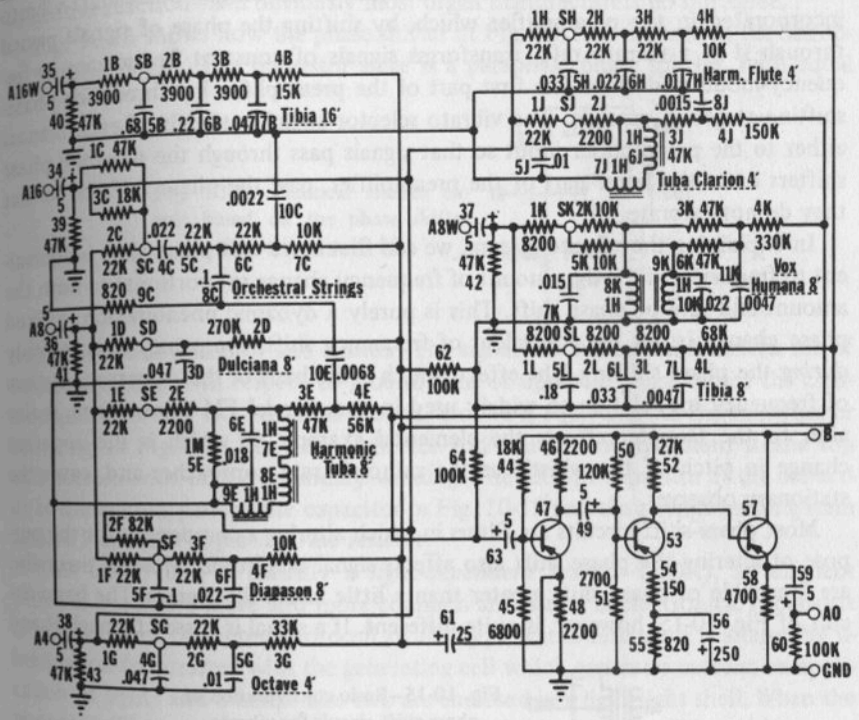


Fig. 10-14–Accompaniment stop filters. Note how amplifier has two input points.

if all stops were fed to the same amplifier input so that all signals were in phase at the output, reeds and flues tended to cancel each other on certain notes and harmonics with certain stop combinations.

The Accompaniment stop filters of Fig. 10-14 are assembled along exactly the same lines. They draw tone from five bus-amplifier sources–the three sawtooth ones, plus woodwinds at 16′ and 8′ pitches.

The Schober Vibrato

Signals from the two stop filter amplifiers reach the organ preamplifier by way of a manual balance switch and a vibrato selector switch. The manual balance is a rotary switch with three positions. At center, both signals go to the preamplifier through equal resistors. At one side position, greater resistance is in series with the Solo signals and less with the Accompaniment, and in the other

side position the reverse is true. At center, therefore, stops of both manuals have their normal relative amplitudes, while at each side position one manual is emphasized over the other.

The vibrato selector is another 3-position rotary switch. At center, both manual signals have vibrato; at each side position one manual has vibrato and the other has not. The ability to do this despite the fact that the organ has only one set of frequency-divider generators is a result of the unique phase-shift vibrato incorporated in the preamplifier which, by shifting the phase of signals passed through it at a vibrato rate, transforms signals of constant frequencies to frequency-modulated ones. The first part of the preamplifier has three such phase-shifting stages in cascade. The vibrato selector switches signals of each manual either to the preamplifier input so that signals pass through the dynamic phase shifters or to the latter part of the preamplifier, past the phase shifters, so that they do not "vibrate."

In explaining the vibrato system, we can first agree that phase shift is equivalent to frequency shift, the amount of frequency change proportional to both the amount and rate of phase shift. This is purely a dynamic phenomenon; a fixed phase change is not the equivalent of frequency shift. Frequency changes only *during* the phase shifting. The effect, which is the basis of the Armstrong system of frequency modulation so widely used in commercial FM transmitters, is also akin to the Doppler effect, the elemental example of which is the apparent change in pitch of the whistle as the railroad train approaches and leaves the stationary observer.

Most phase-shift circuits are filters in which altering a component for the purpose of altering the phase shift also affects signal amplitude. Most of them also are incapable of phase shift greater than a little under 90 degrees. The basic circuit of Fig. 10-15, however, is quite different. If a signal is passed through it and

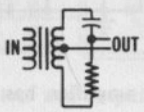


Fig. 10-15–Basic configuration of phase-shift circuit for vibrato.

either the R or C element varied, the signal phase can be made to change over a range of nearly 180 degrees without changing its amplitude. Unlike other shifters, it is also important that this one is not a filter and its frequency *vs* transmission curve is flat over an extremely wide band. While the absolute phase of an output signal of particular frequency with respect to the phase at the input depends on frequency, the variation with respect to resting phase when a component is varied is quite wide over a large band, decreasing gradually above and below that band. This latter point is a fortunate one, for when organ signals are passed through this circuit and a component varied at vibrato rate, maximum vibrato is obtained around the center octaves, for which the components are chosen, and vibrato depth decreases at the higher harmonics and the lower fundamentals.

The resulting sound is more pleasant than when all fundamentals and harmonics are vibrated by the same percentage. The reason for this at low frequencies is that the modulation frequency (around 6 to 7 cps) becomes too large a percentage of the tone frequency—which is also why pedal vibrato, which can, of course, be avoided by this system, is unpleasant. At the higher harmonics, a vibrato percentage equal to that applied to the midrange is described by many people as a sort of squeal. I have not yet found a rationalization for this quite subjective reaction—and obviously most organ manufacturers do not agree.

Fig. 10-16 shows how the phase shifter of Fig. 10-15 is realized in the Schober Theatre Organ. The transistor stage is a paraphase phase splitter, with equal

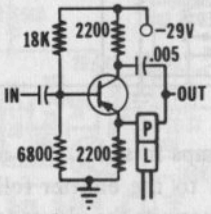


Fig. 10-16–Practical vibrato circuit based on the phase shifter shown in Fig. 10-15.

load resistors at collector and emitter. The signals at collector and emitter are of equal amplitude with respect to ground, and of opposite phase. Since the common point for the two signals is ground, ground is equivalent to the transformer centertap in Fig. 10-15 and the collector and emitter are equivalent to the top and bottom ends of the secondary winding. The .005-μf capacitor at the collector is the counterpart of the capacitor in Fig. 10-15 and the strange looking item labeled P and L is in place of the resistor.

This latter item is actually a *light-dependent resistor* (LDR), an element which is becoming more and more common and useful in electronics. It consists of a small semiconductor photocell of the type whose resistance is altered by illumination (contrasted with the generating cell which generates current on application of light) and a lamp. The two are enclosed in a light-tight shell. When the photocell (P) is placed in a circuit and current fed to the lamp (L) we have a resistor whose value can be controlled over very wide ranges (from several megohms down to a few hundred ohms or less, for example) by the lamp brightness. The control current to the lamp is entirely independent of the circuit in which the cell is placed, and since lamps operating on very low currents and voltages are available, lamp current can be controlled by a small-signal transistor as well as by physical means such as a potentiometer or switch. The lamps used in Schober's LDR's are rated at 10 volts at 20 ma, which can easily be supplied through a small transistor. When the LDR lamp in this circuit is energized by a vibrato-frequency current, the resistance of the cell in the phase-shift circuit varies widely at the same rate, producing a dynamic phase shift and resultant vibrato of any organ signals passed through the circuit. Three of these stages are used in cascade, with amplifiers between them to restore lost gain and load each stage with the very high impedance needed for maximum phase shift, to achieve as wide a theatrical vibrato as anyone might wish.

Fig. 10-17 shows the source which drives the three lamps of the LDR's. The

first transistor is a conventional phase-shift oscillator with three phase-shift sections in the feedback network. A part of the resistance of the first section is variable on the printed circuit to adjust the vibrato speed. Output is taken from the oscillator collector to an emitter follower which furnishes oscillating current

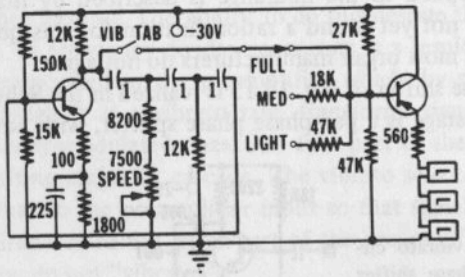


Fig. 10-17—Low-frequency oscillator and emitter follower supply current for vibrato LDR lamps.

to the three lamps in series. The normally open vibrato tablet switch is in series with the signal to the emitter follower, as is a 3-position rotary switch on the console which permits the player to adjust vibrato depth.

The preamplifier is in two parts, the first of which is the series of vibrato stages. The vibrato selector switch on the console can therefore switch the signal of either manual either through or around the vibrato stages. The second part of the preamplifier also has inputs for the pedal signals and any auxiliaries, such as a chime unit, that may be used. It also incorporates the swell shoe control, which is a 10,000-ohm Allen Bradley Type J potentiometer which forms part of a signal feedback circuit. The preamplifier output is an emitter follower.

Pedal Generator

The pedal switching and signal handling is carried out quite separately from the manuals, as is fairly common in nonclassic organs. Each of the 25 pedal switches is a single-pole double-throw, of which the three lowest and three highest are shown in Fig. 10-18, all in the normal (no pedal pressed) position, so that the line to the pedal generator is grounded. The lowest 25 notes of the main generators—the lowest one of which is at 65.4 cps, the lowest 8′ C of an organ—are fed to the lower contacts of the switches. When any switch goes down, the output line picks up the corresponding note from the main generators. If two or

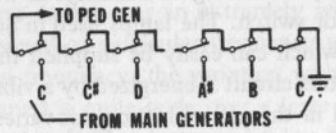


Fig. 10-18—Pedal switching idea is illustrated by three lowest and three highest switches.

more pedals are pressed at the same time, the lowest note of those pressed is the only one passed to the output. The switches themselves are flat-blade stacks with gold contacts. They are mounted on a rail in the console and are pressed by the tips of the pedals.

The pedal generator, diagrammed in Fig. 10-19, takes the output signal from

the pedal switch system at terminal PS and passes it first through a 2-stage squarer. An 8′ output is taken from the collector of the second transistor for the 8′ pedal stop filters. The same signal is differentiated, further sharpened by diode 15, and used as trigger for the flipflop, transistors 25 and 26, from which signal for the 16′ stops is derived.

The diapason-type pedal filters (Diaphone and Dulciana) contain an element of 8′ tone as with the manual filters. The reed filters are unusual. In the Tuba Profunda, for example, the 16′ square-wave input is differentiated by capacitor

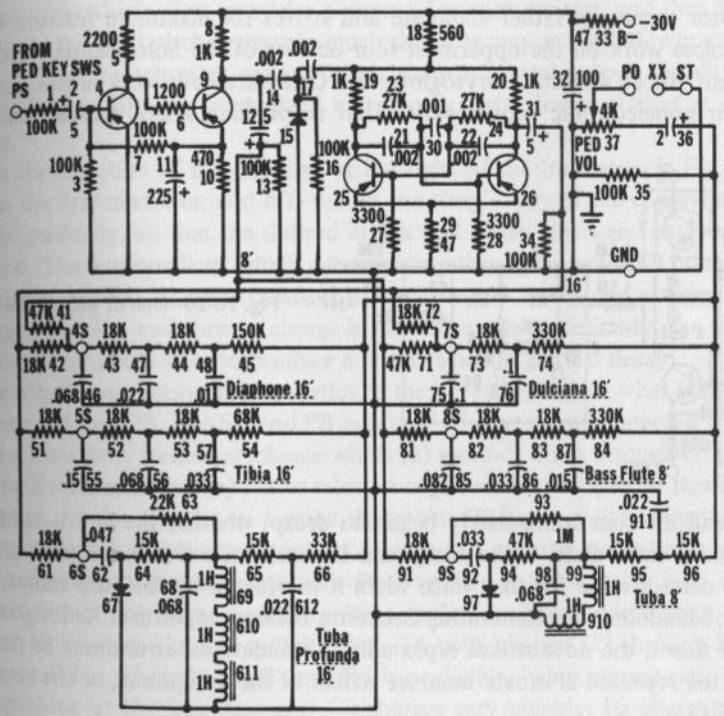


Fig. 10-19—Theatre Organ pedal generator has flipflop plus all formant filters for stops.

62 and following shunt impedances and rectified by diode 67. The result is a series of strong negative pulses which shock-excite the tuned circuit and produce the pronounced formant desirable for a good, penetrating pedal reed, without the poor scaling which would result from simply passing tone right through a fairly sharply tuned filter. The output is a series of damped wave trains of the formant frequency, at the repetition rate of the notes themselves. Some of the original note is passed through the filter via resistor 63 to fatten the sound. The same procedure is followed with the Tuba 8′.* All filter outputs are connected

*Don't try this dodge with a manual filter or with a polyphonic pedal system. If more than one note passes through the diode at a time, horrible distortion will ensue!

together and to the hot end of preset pedal volume control 37 and output is taken from the arm at terminal PO. This output is passed through a 3-position rotary switch on the console, which interposes three net values of resistance before sending the signal into the preamplifier, so that in addition to the varying volumes of the pedal stops the player has the flexibility of a pedal balance switch to vary total pedal volume a few decibels up and down.

Percussion

The Schober Percussion Group, which is available optionally for all Schober transistor organs, is rather elaborate and strives for maximum realism. Percussion voices work on the uppermost four octaves of the Solo manual, and there are eight voices. Celesta, Chrysoglott, and Orchestra Bells work in the same way as their nonelectronic counterparts. That is, pressing a key causes a strike of

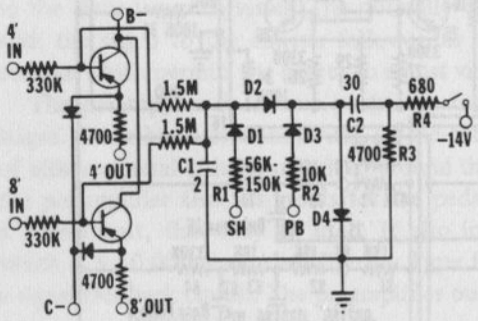


Fig. 10-20—One of percussion keyers.

tone, and the tone immediately begins to decay, whether the key is held down or released immediately. The decay time for each voice is automatically adjusted to the correct value for that voice when it is selected by the stop tablet. Xylophone, Mandolin, and Reiterating Orchestra Bells are repetitive. As long as a key is held down, the note strikes repeatedly to simulate the strumming of the mandolin, the repeated alternate hammer strikes of the xylophone, or the ringing of the bells.

The Piano and Harpsichord stops have damper action. When the key is pressed and held down, the tone strikes and immediately begins to decay. This decay takes quite a long time. If the key is released at any time before the decay is complete, the sound stops almost immediately as it does in the genuine instrument when the felt damper drops on the string.

All this requires a good deal of circuitry, much of it repetitive. We shall be fairly sketchy in our explanations here; if you want full details, you can send for Schober's 25-cent booklet, "How Schober Organs Work," which contains full circuitry and expanded explanations.

The system is operated by a single-pole, single-throw, normally open switch under each key. This switch is in addition and unrelated to those described earlier for the normal organ tone switching. The key switch appears at the right of Fig. 10-20, which is a diagram of one of the 48 almost identical keyers.

Looking first at the transistor end of the diagram, each of the two transistors takes care of one of the two percussion pitch registers, 4′ and 8′. In each case, the appropriate tone from the main generators is permanently connected through a 330,000-ohm resistor to the transistor base. Each transistor is an emitter-follower switcher of the same kind as in the coupler switchers, Fig. 10-9; notice the emitter diode in each case, biased negative, so that in the absence of a negative on bias applied to the base, not only is the transistor turned off but the diode also shorts the emitter to ground and prevents any tone leaking through the transistor from going to the output. The output terminals shown go to a resistor network which, first, provides a ground for each emitter, and then gathers all outputs for each pitch register in much the same way as the Baldwin network of Fig. 9-8. For each pitch register, then, there is a total output for all keyers with the higher pitches emphasized and another with lower pitches at greater volume.

It is the function of the circuitry at the right of the transistors in Fig. 10-20 to turn the transistors on and off—not in the simple way of the coupler switchers, but gradually, so that the desired attack and decay times and envelopes are obtained. The intermediary which accomplishes this is capacitor C1, the charge on which is applied to both transistor bases through 1.5-megohm resistors. If C1 is charged quickly and then its charge is allowed to dissipate slowly, the transistors control the audio so as to produce a fast attack and gradual decay.

The attack and decay characteristics of the keyer depend on what happens to terminals SH and PB. The SH and PB points of all keyers are connected together and to a transistor switching scheme which (a) grounds them through effectively very small resistance and (b) places selected negative bias voltages on them at the same time. At the start let us assume that point PB has very little negative bias, so that diode D3 is willing to conduct whenever a negative potential is applied to its cathode; but that the negative voltage applied to SH is rather high, so that D1 can conduct only if the negative voltage applied to its cathode is great.

Now let us close the key switch. The −14 volts charges C2 through R2 and D3. Since R2 is relatively small (and R4 is negligible, being present only to prevent sparking at the key contacts) C2 charges very quickly. Its charge current causes a pulse of negative voltage across the D3-R2 combination. This pulse passes through D2 and immediately charges C1 fully.

As soon as C2 has become charged, the closed key switch does nothing further. C1, meantime, begins to discharge as soon as it is charged. It cannot discharge through anything to the right of D2, as the C1 charge is negative and D2 is turned the wrong way. Its initial discharge takes place through the D1-R1 combination. D1, as we have said, is biased externally, but its off bias is not quite equal to the full charge potential placed on C1. Initially, therefore, C1 can discharge through D1-R1, which it does until the remaining charge is not negative enough to keep D1 conducting. Thereafter, its only discharge path is through the 1.5-megohm resistors and the transistor base circuitry.

In response to all this, the initial fast charge of C1 has caused both transistors to be turned fully on at once, sending full tone to both outputs. During the in-

itial phase of C1 discharge through D1-R1, transistor on bias recedes rather quickly, so the tone amplitude fades down fairly fast. Thereafter, the discharge through the larger resistances causes relatively slow decay of tone. This quick strike, fairly fast initial descent, then slow decay, is the ideal pattern for a voice such as Chrysoglott or Celesta. These voices ought to have a long decay, but amplitude must descend from its peak value to some lower value at an initially higher rate, to achieve realism and make it possible to play the same note repeatedly and hear the strike each time.

An orchestra bell ought to have a shorter decay time. To obtain it, the negative potential applied at SH is reduced. Now a greater part of the C1 discharge can take place through the D1-R1 combination (and the abbreviation SH–*shortener*–begins to make sense). A xylophone sound lasts only a very short time indeed. For that, the bias at SH is reduced still further, so that almost the whole of the C1 discharge takes place through it.

What happens when the key is released? Obviously, C2 discharges; the excess negative electrons on the right plate pass quickly down through R3 and up through D4 (which is polarized correctly for C2 discharge but has no effect on its charge) to the left plate, and C2 is ready for another switch closure.

The so called *piano circuit* operates through terminal PB–an abbreviation for *piano bus.* When the Piano or Harpsichord is turned on, a rather high negative voltage is applied to PB so that D3 remains nonconducting through the entire operation. At the same time, almost all bias is removed from SH, so that D1 does conduct whenever its cathode sees negative potential.

Now when the key is pressed, C2 again begins to charge. This time, however, it cannot charge through R2 (since D3 is biased off), but must charge through D2, D1, and R1 instead. R1 is much larger than R2, so C2 charges at quite a slow rate. At the start of the charge, the charging current through R1 is high, and as the charge proceeds, while the key is held down, the charging current gradually diminishes in the normal exponential manner. Since it is the voltage drop across R1 which is applied to C1 and the transistor bases, the transistors quickly start to conduct when the key is pressed, but the decay of the on bias applied to their bases is very slow, because it is not now controlled by C1 but by the charging of C2. As long as the key is held down, therefore, the tone decays very slowly.

Actually, bias is not *completely* removed from SH. Therefore, after a certain time, the D1-R1 circuit ceases to conduct and the remainder of the decay is controlled by the discharge of C1 through the base resistors, which are larger. Thus, again, we have an initial decay rate, followed by a slower decay rate, though the total now is relatively very slow. The R1 resistors in the 48 keyers vary in value, with the smaller resistors near the top. This makes the decay somewhat faster for treble notes to add realism, since piano and harpsichord treble notes always decay faster than bass ones. These resistor value variations are not large enough to cause appreciable differences with more "normal" percussion voices.

So far the key has been held down. If the key is released, C2 immediately discharges through R3 and D4, just as we have described for other stops. The only

remaining on bias for the transistors is whatever charge is on C1 at the time. And since D1 is almost unbiased, C1 quickly discharges through D1-R1 and the note stops sounding very quickly–as quickly, in fact, as would a Xylophone note, since the same bias is applied to SH for Piano and Harpsichord as for Xylophone.

We thus have a Piano action that comes very close to the real thing as far as attack and decay are concerned. No tone coloring system can make a real piano sound without the very complex harmonic and time effects of the strings in a piano, which to my ear have never yet been imitated electronically; but the total effect is surprisingly convincing.

The last of the three different kinds of percussion stops is the reiterating or repetitive type–Mandolin, Xylophone, and the reiterative version of the Orchestra Bells. The obvious way of creating the repetitions is to change the steady –14 volts at the key switch in Fig. 10-20 to a pulsating voltage. That is what is done, but with some necessary complications.

Fig. 10-21 shows everything having to do with both the steady and pulsating keying voltages. Q2 and Q3 form a multivibrator. The emitter of Q2 is connected to ground through Q1, which is used as a switch. Normally Q1 does not con-

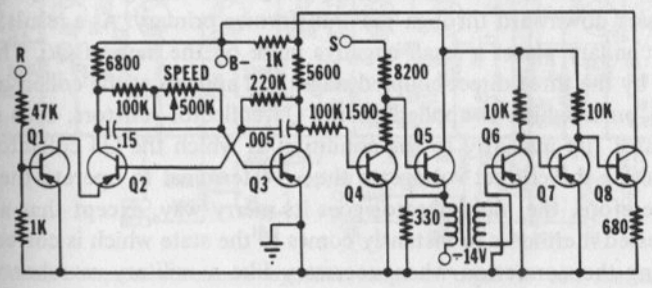


Fig. 10-21–Multivibrator and trigger stages make up keying voltage source for percussion.

duct and the multivibrator does not oscillate. When terminal R is furnished a negative on bias, Q1 conducts fully and the multivibrator goes on. The negative voltage is switched to point R and to a number of other points for various control purposes by the appropriate stop tablets. All this d.c. switching is done through diodes, so that each stop switch need be only a single-pole, single-throw, normally open type, with the diodes inserted wherever necessary to prevent backups–voltages applied to common connections, thence to unwanted points.

With terminal R unconnected to a d.c. source, only Q3 conducts. An appropriate voltage is connected from its collector through direct-coupled Q4 and Q5 and the primary of the transformer, and this turns out to be the steady –14 volts shown in Fig. 10-20. Terminal S is furnished B– whenever any of the percussion tablets is on.

When a repeating stop is used, Q1 conducts, the multivibrator multivibrates, and its output reaches the –14-volt point so that the keyer of Fig. 10-20 is furnished a series of pulses and the sound is keyed repeatedly. The three voices used with repeat have very short (but not instant) decays, so they sound realist-

ic. A simpler circuit formerly used by Schober just keyed an amplifier on and off with instant switching and was far less effective.

The remaining circuitry solves an important problem. Suppose the system is set up for repeat and the organist plays something which includes both very short and longer notes. For a note which is to be held for some time, it is not terribly important that it sound at the very instant the key is pressed—as it might well not if the key happens to be pressed at an instant which comes between repetitions of the circuitry. But a short note, which might last only for the time of one sounding, might very well be pressed when the keying pulse is not present and released before it comes along. Thus the note might be missed entirely. In any case, it gives the player a very unpleasant feeling if responses to his keying are sometimes delayed.

Let us suppose that this has happened. The voltage at the −14 point must be at −14 volts for a note to strike. Tracing back through the normal phase reversals, this means the square wave at the Q3 collector must be at its positive peak. Suppose it is negative instead, meaning that Q5 is almost (but not quite) nonconducting, instead of conducting heavily as it should be. When the key is pressed, the keyer still draws a small current (though not enough to key effectively) which passes downward through the transformer primary. As a result, the transformer secondary places a small negative pulse on the base of Q6. This pulse is amplified by the three direct-coupled stages and appears at the collector of Q8 as a positive pulse which is applied to the Q3 collector resistors. This makes the multivibrator flip instantly to the condition in which the Q3 collector is positive, furnishing the correct voltage to the −14 terminal to operate the keyer. In effect, therefore, the multivibrator goes its merry way, except that any time a key is pressed it either is or instantly comes to the state which is correct for keying, making the correction when necessary like a military marcher who finds himself out of step and corrects it with a practiced little skip.

The remainder of the percussion system circuitry consists of a gate transistor for each stop (again using the emitter diode idea to prevent leakage through a closed gate) followed by a formant filter for voicing. All the filter outputs are gathered together and amplified, and fed to the mixer-compressor.

The Mixer-Compressor

The Schober Theatre Organ (and, optionally, other Schober transistor instruments) contains a most unusual item, a volume compressor. Ideally, whenever a stop is added to an existing ensemble, it should audibly add to the voices already on, not merely modify them to produce a resultant tone color and volume. Methods of arranging loads and other impedances in Schober Organs does yield this result, so that as voices are added the added voices increase both the complexity of the timbre and the volume.

The result is that there is a very wide difference in volume between a single small solo voice and a full ensemble of stops. This is not an unmixed blessing, because if the soft voices are made loud enough to be useful (by arrangement of the fixed volume controls), the full, loud ensembles may be so loud as to knock

down the walls of an average living room or, at the least, overload a moderate-capacity amplifier and speaker system. In a church, auditorium, or large home room, the result is excellent, for in good-sized spaces wide dynamic ranges can be accommodated without discomfort and as large amplifier-speaker systems can be used as are necessary.

To make the organ equally satisfactory in small rooms without penalizing possibilities in larger ones, a compressor is furnished. Amplifier volume can be set so that sound is adequate with soft stops. Then, using the loudest ensembles, the owner adjusts the compressor so the volume will not exceed what can be tolerated.

Compressor problems can be severe with an organ; this one was designed to eliminate them. Fig. 10-22 is a full diagram of the mixer-compressor. Organ output is applied to the PR terminal and goes to transistor 36 through a voltage

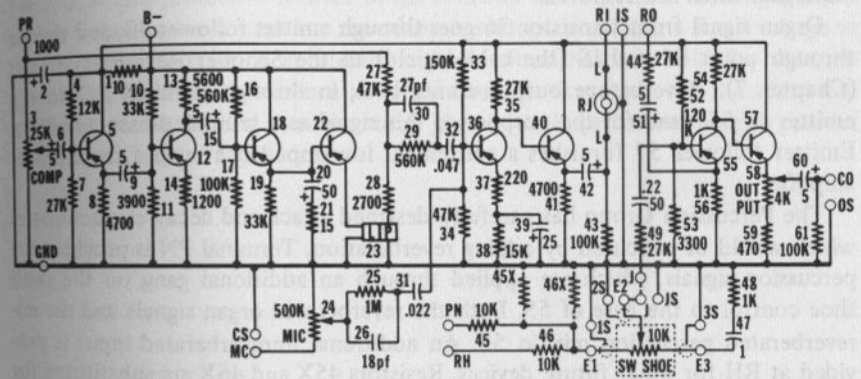


Fig. 10-22—Mixer-compressor controls dynamic range of organ to suit the installation.

divider composed of resistors 27 and 28 and the photocell part of a light-dependent resistor (LDR) identical to the one used in the vibrato circuits, Fig. 10-16. If the photocell is at maximum resistance, signal to the base of 36 is maximum. If the cell resistance gets smaller, the signal at 36 is reduced or compressed.

Transistors 5, 12, 18, and 22 form the compressor circuitry, whose task is to energize the lamp section of the LDR so as to cause the photocell to reduce resistance and cause compression. The audio input signal from PR is also fed, through the compression control, to the base of isolating emitter follower 5. Let us assume that compression control 3 is set for maximum—the arm at the top. Output of 5 feeds amplifier 12, which is linear until, when the signal at the arm of control 3 reaches about 2 volts, transistor 12 begins to clip symmetrically, preventing overdrive to stage 22 and the lamp. It also limits compression so that output signals due to an extremely loud input do not eventually become softer than those due to moderate input.

Transistor 18 is biased only very slightly on. In the absence of signal, its emitter connection to 22 causes 22 to conduct only slightly, which keeps the lamp barely illuminated to reduce its thermal inertia so that when it is asked to go on

and begin compressing the signal, it will do so without appreciable delay.

The low on bias of 18 also makes 18 a rectifier for all but the very smallest signals, since only negative audio alternations can cause significant emitter current. As a result, audio applied to the control system appears at the emitter of 18 as varying negative d.c. which charges capacitor 20.

When a big signal peak comes along, 20 charges quickly and compression is very fast, so much so that it is not even apparent except in the most extreme cases. When the signal is removed or reduced, the only on voltage at the base of 18 is the slight fixed negative bias and 18 is no longer a low-impedance source connected to 20. Thus 20 has no low impedance through which to discharge quickly; it must discharge through 19 and the base of 22, which are fairly high-impedance paths and cause a slow discharge. The result is fast compression attack when a loud sound comes through, followed by a relatively slow return to maximum gain when it is removed.

Organ signal from transistor 36 goes through emitter follower 40, and thence through point RI and IS (the cable shield) to the Schober Reverbatape Unit (Chapter 7). Reverbatape output comes back in through terminal RO to the emitter of 55, used for the purpose of this signal as a grounded-base amplifier. Emitter follower 57 furnishes a convenient low-impedance output for a power amplifier.

The Percussion Group has carefully designed attack and decay characteristics which would be obscured by adding reverberation. Terminal PN is provided for percussion signals, which are applied through an additional gang on the swell shoe control to the base of 55. Both the reverberated organ signals and the unreverberated percussion mix in 55. An additional unreverberated input is provided at RH for other future devices. Resistors 45X and 46X are substituted for 45 and 46 when either of these special input channels is unused. As an additional

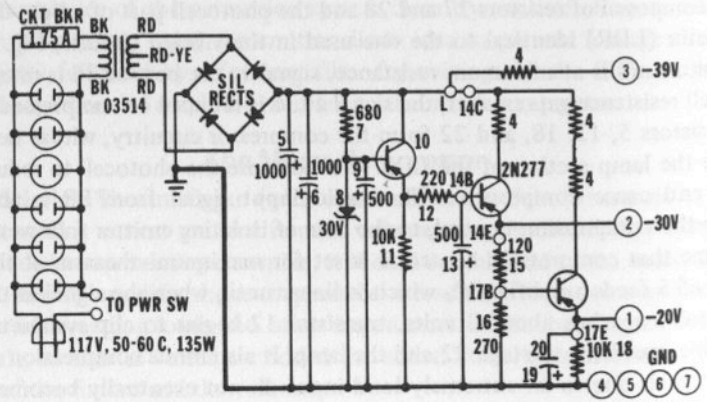


Fig. 10-23—All supply voltages used in the organ are regulated by zener and transistors.

wild gambit, a microphone input is also furnished. As you can see, it passes signals through the Reverbatape Unit, along with those of the organ, so that as

Mama plays the organ, Papa and offspring can sing through the mike and sound as though they were in Grand Central Station.

Power Supply

The Theatre Organ power supply diagrammed in Fig. 10-23 furnishes two regulated voltages for the entire organ, plus the unregulated −39 volts for the lights buried under the front of the lid, without whose rays shining on the stop tabs no dyed in the wool theatre organ fan could play a note. The main regulator begins with small-signal transistor 10. It is an emitter follower, with its base voltage furnished by a voltage divider which includes a zener diode in the shunt leg. This supplies a constant voltage to the base, and as always in an emitter follower, almost the same voltage appears at the emitter. The emitter voltage is furnished to the base of the 2N277 power transistor. This is a second emitter follower (with a 4-ohm protective resistor in the collector circuit) which actually handles all the organ power except for the tone generators, providing a fixed voltage at its emitter, −30 volts. A subsidiary regulator furnishes −20 volts for the tone generators, deriving its reference potential from the first regulator; the additional regulation is not needed as such, but it gives a low-impedance source to prevent interaction among the generators.

Chapter 11

Thomas Organs

The Thomas organ, made by Thomas Organ Company, 8345 Hayvenhurst Avenue, Sepulveda, Calif., first saw the light of day in the late fifties as one of the very first real attempts at an extremely low-priced instrument for the home. This first model (briefly described in the second edition of this book) was no great shakes musically. It had a single manual and a system of tone coloring controlled by five knobs (for flute, string, diapason, horn, and reed) which could be adjusted to provide each of these colors in varying volumes. One of its major sources of economy was the use of about one third as many oscillators as keys, with a sharing arrangement whereby pressing a key would tune an oscillator to one of three adjacent notes.

Thomas now has a good sized line of electronic organs, concentrating, like Lowrey (which is probably its closest competitor) on instruments for the general public, mostly in the lower price ranges. Like Lowrey, Thomas concentrates on design economy coupled with a display of "features," rather than on the sort of circuit elaboration and authenticity which characterize some others such as Baldwin. While I suppose you will find me somewhat prejudiced in favor of the more straight and pipelike, less gadgety designs, there is an excellent case for the Thomas-Lowrey approach. The general public does not have a particularly musical or traditional background and apparently likes an instrument which does interesting things and which does not cost as much as those designed more traditionally—or which does cost as much but spends the money on gee-whiz tricks.

From the engineering viewpoint, I readily admit that it takes more ingenuity to design economically than to have the freedom to throw in anything that a particular result seems to require, or to think up some of the wild effects that appear in organs of this kind. This makes Thomas and Lowrey organs especially interesting from the standpoint of this book. The balance of economy *vs* results or of sound effects *vs* music is presumably decided by each company on the basis of marketing research among its prospective customers, and since the law in most states says the primary purpose of a corporation is to make as much money as it can, any profitable decision is a right one.

Be these points of philosophy what they may, Thomas organs are sold in vast quantities and are even sold in kit form by Heath. Unlike Lowrey's, Thomas' designs are fairly straightforward and contain nothing especially wild, so they make a good introduction to the economical branch of organ construction.

The Thomas organ described in this chapter is one of the larger-than-spinet models, the Celebrity 800. As the photo of Fig. 11-1 shows, it is a theatre-type organ with two 61-key manuals and a 25-note flat pedal clavier. It has 27 speaking stops, plus three sustain voices, in 16′, 8′, 4′, 2⅔′, and 2′ registers on the upper manual and 8′ and 4′ on the lower. The pedals are monophonic, with 16′ and 8′ registers, and there is a pedal sustain. A species of attack percussion can be applied to certain groups of manual voices, and there is a 28-note chime range. Other points will appear as we progress.

Tone Generators

The Thomas frequency sharing has long been a thing of the past; like all modern instruments of Thomas, the Celebrity contains 12 frequency-divider tone generators. Each one consists of a master oscillator and the sort of flipflop



Fig. 11-1—This is the Thomas Celebrity, Model 800, described in detail in this chapter.

dividers which have become standard in the field. Fig. 11-2 shows the master oscillator schematic. Both collector and base circuits are resonated with coupling through a centertapped, adjustable-core coil. The output is marked 6 to show that it generates notes in Octave 6 of Table 1-1, except for the C generator, which is the highest of the string rather than the lowest and whose master oscillator generates the C of Octave 7. Throughout this chapter we shall use numbers

identifying the frequencies with the octaves of Table 1-1 rather than those used by Thomas, which are based on 1 as the lowest octave actually supplied by the main generators (69 to 131 cps).

The master oscillator output is used directly for the highest tones. The diode clips positive excursions, so that the output waveform has some harmonics and

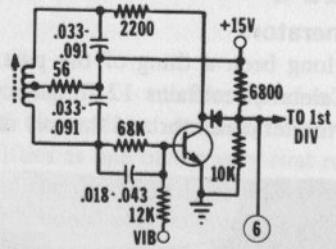


Fig. 11-2—The master oscillator.

is not a sine wave. The output is also fed as trigger signal to the first divider.

Fig. 11-3 shows the first of the four frequency dividers in each generator. All four are identical and all components of each, except the transistors, are contained in a circuit pack like that shown in Fig. 3-17. The main generators thus produce five octaves of tone, the highest in Octave 6, so that the highest two oc-

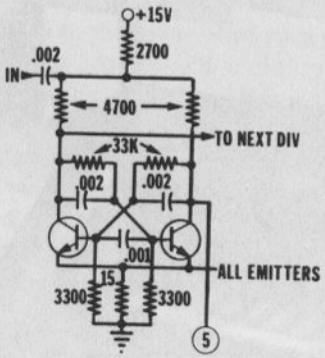


Fig. 11-3—All frequency-divider flipflops are identical to this one.

taves of keys repeat the last 2′ octave of tone and the 4′ keys are complete to within one octave of the top, with consequent one-octave repeat.

The generator outputs are all (except for the master oscillators) square waves, and square they remain right up to the voicing filters, as you will see. Vibrato is provided by an a.c. signal fed to the master oscillator bases, as indicated in Fig. 11-2. Its source involves other circuitry and we will come to it later.

Manual Main Key Switching

There is a "main" keying system for each manual, and in addition a sustain keying setup for the 8′ voices of the Solo (upper) manual. The main keying systems are identical for the two manuals, except that the Accompaniment (lower) is simpler because there are only two pitch registers instead of five. We shall describe the simpler Accompaniment and thereby understand the Solo as well.

The main keying idea is simple and straightforward. Under each key two

switch wires rise when a key is pressed, and each touches a bus running the length of the manual. One rank of wires carries tones from the generators for the 8′ pitches and the other for the 4′. In series with each switch wire there is an isolating resistor, all of them 100,000 ohms. To show how the system works, we shall deal only with the 8′ keying.

The 8′ bus is actually divided into five sections, each insulated from the others. The first bus picks up 8′ tones from the first three octaves of keys. Each of the following four busses picks up tone from a succeeding half-octave of keys. These five separate sections of the 8′ bus are shown by the terminal circles labeled 8′ Acc Keying Bus Sections in Fig. 11-4.

Signal from each bus section is passed through a 10,000-ohm resistor and the common junction is connected to the input of a one-stage amplifier, whose out-

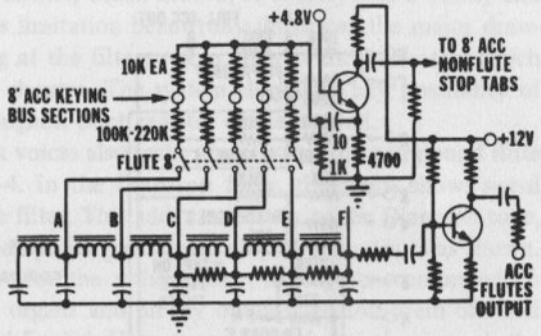


Fig. 11-4—Compound Accompaniment flute filter and multi-arm stop switch for one of the two Accompaniment flute voices. All 8-foot Accompaniment tones join at top for "complex" (nonflute) tone filters.

put thus contains all 8′ keyed Accompaniment tones, just as though the bus were solid as in the Schober Organ of Chapter 10. This output is used to feed tone to the nonflute stop filters. The common junction of the 10,000-ohm resistors is terminated by the 1,000-ohm resistor to ground. Transfer of tone, therefore, from one bus section to another through the 10,000-ohm resistors is small. That is, while the signals of all busses meet equally at the emitter of the grounded-base amplifier, they remain relatively separate at the bus sections themselves.*

The flute filtering in this organ is reminiscent of that used by Lowrey, which you will read about in Chapter 12, in that simple R-C filtering, one filter to a stop, is not used as in Schober and Baldwin (and some other) organs. Instead, a system of L-C cutoff filters (or the equivalent in the case of Lowrey) is used to knock off harmonics much more sharply than R-C components economically could. Since this would destroy the scaling if one filter were used for several octaves of tone, the tones are divided up into groups. This is the function of the separate bus sections.

*The 1,000-ohm terminating resistor is actually connected to the +4.8-volt line rather than ground; for signal purposes, this is the same as ground, but throughout the organ signal grounds are made to points found to avoid loops and hum production. We shall mark all signal grounds in the same way, without showing whether they are actually chassis grounds or not, to keep things simpler.

The flute filter is a six-section L-C filter, with output taken from the right end. Near the left end, therefore, the cutoff frequency is lowest and the tones from the lowest section of the 8′ keying bus are therefore fed to it—point A in Fig. 11-4. Near the right end, point E, cutoff is highest and the bus section carrying the highest-frequency tones is connected to it. We thus have a system which efficiently removes harmonics to convert the generated square waves to near-sine form for the required bland theatrical flute tones, while having very little effect on fundamentals so as to avoid a scaling problem. The output of the compound flute filter goes to an amplifier, as shown.

The 4′ system for the lower manual is almost identical to that shown, differing only in that the notes covered by sections of the 4′ keying bus differ somewhat. Another complete set of isolating resistors, another grounded-base amplifier, and another multisection tablet switch are provided, but the outputs of the tab switch go to the same points on the same compound flute filter, so that the flute amplifier carries flute tones of both 4′ and 8′ pitches.

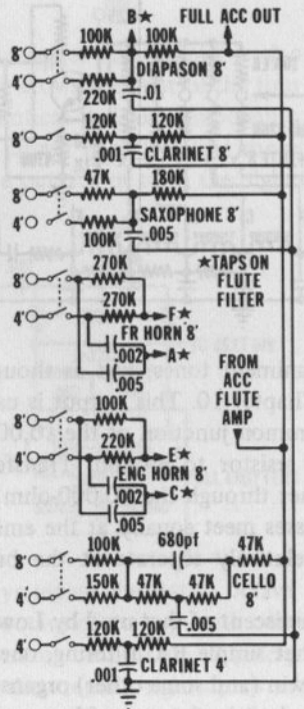


Fig. 11-5—Voicing for the Accompaniment manual.

The main keying system for the Solo manual is identical except in details. There are five keying busses, each in sections; the sections are not of the same length for all pitches. There is a single compound flute filter, as in Fig. 11-4, but with more sections, and a separate set of isolation resistors and grounded-base amplifier for each pitch register. There is, of course, a separate multisection flute switch for each of the five registers, providing three tibias, Nazard, and Piccolo.

Complex Tone Voicing

The nonflute voices of the Thomas Celebrity are voiced by formant filters. Fig. 11-5 shows the voicing for the lower manual. The terminals marked 4′ and 8′ are the outputs from the grounded-base amplifiers, one of which appears in Fig. 11-4. These are, of course, square waves, and before they can be used for most filters, some even harmonics must be added. This is done by staircasing in the same manner as Baldwin (Chapter 9), as can be seen in the Diapason filter, which is typical. The stop tablet switch is double-pole, each pole connected to square tone of one pitch register. The 4′ input resistor is twice the value of the 8′ so that a standard 2-step staircase results. The same procedure governs the Saxophone and, to a modified extent, the Cello. Without a 2′ register to call upon, there is no way of producing staircased 4′ tone. The only nonflute 4′ stop on the lower manual is therefore a Clarinet, which should, of course, sound woody and so can use square waves. This limitation beautifully illustrates the major drawback of the idea of staircasing at the filters rather than at the generators which we discussed in the Baldwin chapter. The system eliminates any possibility of "normal" tone colors for the highest pitch register of the manual.

The lower manual complex voices also make some use of the compound flute filter diagrammed in Fig. 11-4. In the Diapason filter, the arrow shows signal carried to point B on the flute filter. This adds some body to the Diapason tone, since the output of the flute amplifier joins these stop filter outputs as shown. Practices of this type account for the rather heavy, muddy tone colors which tend to characterize Thomas organs and, in my opinion, to rob them of tonal interest. The French Horn and English Horn tones are fed entirely through the flute filter, with 8′ and 4′ tones mixed in desired proportions first by the resistors and capacitors. The resistors feed tone to a comparatively high point on the flute filter, so that for most tones of the range some harmonics are practically unattenuated—low ones. A cutoff filter which cuts off, not right after the fundamental but at a higher point—and I mean *cuts off*, not slopes off as an R-C filter would—imparts a horny quality to the tone. Thus the same filter that makes some tones into flutes can make others into horns, albeit rather dull ones. From each of these two filters, additional tone, staircased by the capacitor values, is fed to the beginning of the flute filter to add body to the tone. The method is economical in that it avoids all inductors outside the flute filter.

The output terminal at upper right in Fig. 11-5 carries all Accompaniment voices.

Fig. 11-6 shows the complex-tone voicing for the Solo manual, which works in the same way, except that no use is made by these voices of the compound flute filter. Here the reed voices have actual L-C formant filters, one for the Oboe and another for the two Clarinets, Bass Saxophone, and Trumpet. There is an L-C filter in the Violin and Violina voices, though these are not, of course, reeds. Obviously, the designer felt that adding a bit of definite formant helped make these voices more realistic. I have found this to be true with a guitar voice in a percussion system, even though a guitar is also far from a reed.

While the upper manual provides five pitch registers, you will notice only 16′,

8′ and 4′ inputs in Fig. 11-6. The answer is that there is only one voice each in the 2⅔′ and 2′ pitch registers, both of them flutes which are handled by the compound flute filter. There is, however, a 2′ complex tone amplifier, but instead of feeding its output to a switch so as to staircase the 4′ Violina, it is added directly to the output of the 4′ amplifier and appears in all 4′ inputs. This adds a third step to the staircasing of 8′ voices.

In looking over the voicing of this organ as a whole, you may notice the paucity of voicing at the higher pitches. On the lower manual, the only 4′ voices are a bland flute and a rather harmless clarinet. Even on the Solo manual, where

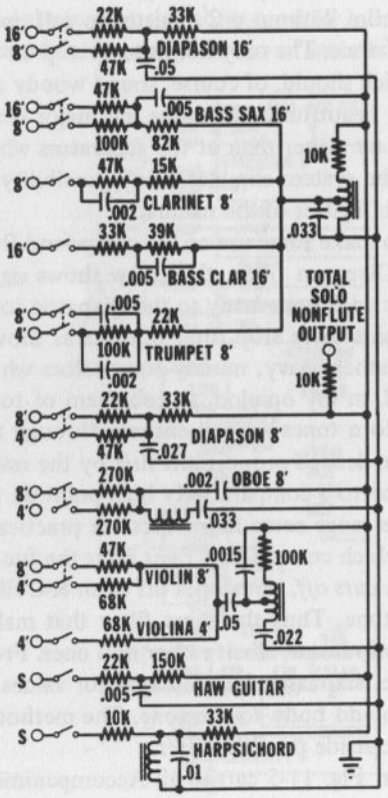


Fig. 11-6—Solo manual voicing includes two of four sustain stops. Chimes and Vibraharp use flute filter.

the more cutting voices would normally be found, the Violina is the only non-flute voice at 4′ pitch and there are none at higher pitches. The instrument, therefore, lacks any great tonal brilliance. The party line throughout much of the industry seems to be that the average householder of no real organistic ability, who is the largest buyer of organs, is rather afraid of much brilliance, and Thomas, in common with purveyors of goods and entertainment in many fields, probably does not feel called upon to educate the taste of the public at the expense of sales.

The two stops whose input tone source is shown as S are for the sustain system. The input tones are at 8′ pitch, and we shall now show where they come from.

Manual Sustain

The sustain is played in the usual way—when a key is pressed, tone comes on immediately, and when the key is released it fades away gradually. There are three sustain stops—the Harpsichord and Hawaiian Guitar shown in Fig. 11-6 and a Vibraharp which is simply voiced as a flute. There is also a 28-note chime.

In Fig. 11-7 the key switch shown exists under each key of the upper manual and is in addition to the five switches which key the regular organ tones. When the key is pressed this switch causes 8′ tone to be sent through the sustain system. There are also three additional switch arms under each of 28 of the keys. For each key these arms are connected to the sustain circuits of the notes neces-

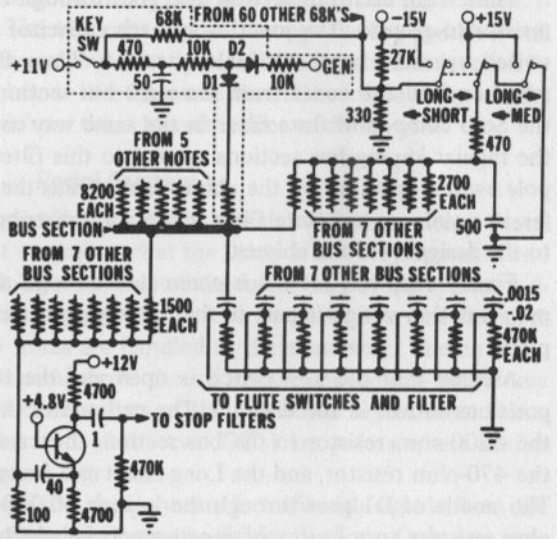


Fig. 11-7—Sustain system. The dashed outline includes components for one note only.

sary to make a major chord based on the note of the particular key, with a lowered third. For instance, the key whose normal 8′ note is C5 also has contacts connected to the sustain circuits of E4, G5, and C6. When these additional contacts are fed keying voltage, all four notes sound through the sustain system when the key is pressed. They cause an approximation of a chime tone.

The whole sustain system can be understood with the aid of Fig. 11-7. The components enclosed within the dashed lines are peculiar to one key and are repeated for each key; the others suffice for the entire system. The GEN terminal carries tone from a main generator stage, and the key switch shown is the pair of contacts for the 8′ regular sustain. (Having briefly explained how the chime works, we now forget about it and its contacts to simplify matters.) Whether the sustain is on or off merely depends on whether the stop tabs for its three voices are on; the circuit is actually working at all times. There are two control tablets whose switches are shown. With both switches in the positions of the diagram

the sustain decay is at its longest. One tablet can be used to set the time at medium and the other at short.

Before talking about the d.c. keying itself, let us find what the signal does. Signal for each note is applied at the GEN terminal. It comes out of the keyer through the 8200-ohm resistor to a pickup bus which is divided into eight sections, each section picking up tones from a group of keyers. The number of notes to a bus section varies from 26 to 5. We have chosen to show just one bus section which happens to handle 6 notes. We have shown five more 8200-ohm resistors to indicate the five other keyers which feed audio to this bus section.

From each of the eight bus sections, tone goes through a 1500-ohm resistor to a common junction connected to the input of a one-stage grounded-base amplifier. The output of this amplifier feeds the two "complex tone" sustain stops at the points marked S in Fig. 11-6.

Tone from each bus section also goes through a capacitor and 470,000-ohm resistor to ground. The junction of each capacitor and resistor goes to an 8-pole switch operated by the Vibraharp tablet. When this switch is closed the eight switches feed the tones from the eight bus sections to various input points on the Solo compound flute filter in the same way as groups of regular tones from the regular keying bus sections are fed to this filter *à la* Fig. 11-4. A second 8-pole switch operated by the chime tablet feeds the same tones to somewhat different points on the flute filter to obtain a distribution of effect more pleasing to the designer for the chimes.

Finally each bus section is connected through a 2700-ohm resistor to a common junction which is part of the d.c. keying system and which we will now explore.

Assume that the key switch is open and the two tablet switches are in the positions shown in the diagram. The cathode of diode D1 is connected through the 8200-ohm resistor to the bus section, thence through a 2700-ohm resistor, the 470-ohm resistor, and the Long-Short and Long-Medium switches to ground. The anode of D1 goes through the keyer's 10,000-ohm, 470-ohm, and 68,000-ohm resistors to a source of negative voltage which is the tap on the voltage divider between the −15-volt source and ground. D1 is therefore blocked. The anode of D2 is connected to the same negative voltage source, while its cathode is connected directly to a generator output. The generator output is a varying positive d.c., ranging from a very small value to nearly 15 volts as the square wave goes through its cycle. D2 is therefore also blocked. No tone can reach the bus section.

When the key switch closes +11 volts, slightly delayed to avoid click by the 470-ohm resistor and 50-μf capacitor, is applied through the 10,000-ohm resistor to the anodes of both diodes. Because of the impedances, the voltage from the source of −15 is entirely blocked off. This causes both diodes to conduct and tone goes through them to the bus section.

When the key is released and the key switch opens, the potential from the −15-volt source is once again applied to the diode anodes to cut them off. It is applied, however, through a time constant composed of the 68,000-ohm series resistor (and the insignificant 470-ohm resistor) and the 50-μf capacitor in shunt. It takes time for the capacitor to charge, and this is the time taken for the diodes to shut off and cause audio transmission to be reduced gradually. This is how the circuit works.

Suppose now that the Long-Short switch is used, so that the arms of this double-pole switch are in the positions *not* shown in the diagram. Now, before the key switch closes the full −15 volts is placed (through resistances) on the diode anodes and +15 (through resistances) on the cathode of D1, leaving matters much as before, so that when the key switch closes the diode still conducts fully. However, the 27,000-ohm part of the resistance through which the 50-μf capacitor must charge when the key is lifted, to cut off the diode, has been shorted, so that the charge takes place faster and the decay does not last so long.

When, instead, the Long-Medium switch is placed in the Medium position, opposite to that shown, the cathode of D1, instead of merely being grounded, is biased off by 15 volts more. The +11 volts applied through the key switch is still enough to turn the diode fully on, but less time than the maximum is taken to turn the diode off when the key is lifted because the full eventual charge of the 50-μf capacitor is not required from the −15-volt source to do the job.

Pedal Sustain

The entire pedal switching system is another sustain, with three decay times, the shortest of which is short enough so that the decay is not very apparent and pedal keying is about normal. The working of the major part of the system is illustrated in Fig. 11-8, in which we have shown two pedal keyers, one for an A note and one for a B. All 25 notes are handled in the same way. The circuitry individual to each note is shown above the corresponding double-throw key switch; the two points where components of the individual keyers meet (junctions of the .001-μf capacitors and of the two D2's) are points where similar components of all 25 keyers come together.

The purpose of this part of the pedal system is to furnish signals from the main generators to a pair of cascaded frequency-divider flipflops. Only one note at a time can be fed to the dividers, so the key switches themselves, labeled simply A and B, are interlocked in the common manner for monophonic pedal systems. All 25 switches are connected in the string in the same way.

In a pedal sustain, pressing a pedal must send one signal to the frequency divider and that tone must continue to go to the divider after the pedal is released, for a period at least somewhat longer than the time required for the note to decay. At the same time, however, if a second pedal is pressed, the lingering note caused by the first pedal must stop immediately and give way to the second. This requires a sort of interlock effect much like the familiar row of interlocked mechanical pushbutton switches; pressing any button causes any button previously in to spring out and the last one pushed remains in. Indeed, Hammond actually uses a mechanical interlock system in the pedal sustain of its spinet organs. Here, however, a purely electronic interlock is employed. Appropriately, Thomas calls it the "pedal cancel," since its function is to cancel anything re-

maining of one note when the next is required. The pedal cancel circuit is embodied in the two transistors of Fig. 11-8.

Signal is furnished to each keyer from the main generators at the terminals shown at the top. The signal used is in Octaves 3 and 4, Table 1-1, rather than the lowest notes available, as Thomas finds it more convenient to use two cascaded flipflops to furnish 8′ and 16′ tones; their outputs are similar and can be handled alike.

The signals are blocked by diodes D1. When the A pedal is pressed, potential from the +18-volt source is applied through intervening resistors and the arm of the A switch, which is now touching the lower contact, to capacitor CA. Since the series resistance from the voltage source is quite low, CA charges almost immediately and biases the anode of diode D1-A so that it conducts and passes the

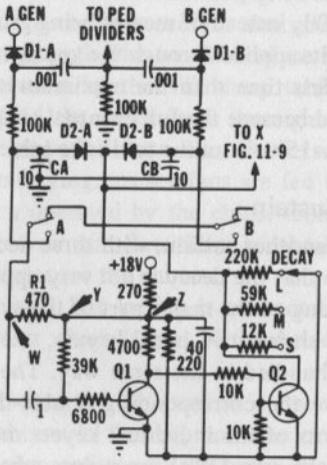


Fig. 11-8—Functioning of pedal sustain keying is illustrated with two pedal notes, an A and a B.

A signal through its .001-μf capacitor to the pedal dividers. When the A pedal is released and the A switch returns to the position shown, the charge on CA dissipates through D2-A, Q2, and a resistor which depends on whether the switching is set for long (L), medium (M), or short (S) decay. In the Long position, the resistor is maximum so CA discharges slowly; in the Short position it is minimum, causing a fast decay. (The resistor switching actually involves tablets and two s.p.d.t. switches, but it is more simply shown as a 3-position selector.)

Let us now suppose we have keyed the A tone and then released the A pedal with decay set for Long. *While the A tone is still coming through to the dividers* in the course of its long decay, we press the B pedal.

First, positive potential from the 18-volt supply is applied to capacitor CB, quickly charging it and turning on the D1-B diode.

During the short time required for charge, CB draws a large pulse of current. This current is drawn through resistor R1 and creates a sizeable pulse of voltage across R1, making point W negative with respect to point Y. The negative pulse at point W is transmitted by the 1-μf capacitor and 6800-ohm resistor to the base of transistor Q1, but has little effect on the supply voltage for the collector

of Q1 because point Z, from which the voltage comes, is the result of a voltage divider from the 18-volt supply and is heavily bypassed to ground.

The negative pulse thus applied to the base of Q1 appears at the Q1 collector and thence the Q2 base as a positive pulse. At this time Q2 is working for a living since the positive potential being gradually drained from CA is being passed through D2-A and is appearing at the Q2 collector as the normal positive collector potential. There appears, therefore, at the Q2 collector, a negative pulse. This pulse is heavy enough so that when passed through D2-A to CA it quickly drains what is left of the original CA charge and shuts off the A keyer. It also, of course, passes through D2-B to CB; but since the B switch is closed at this time and positive potential is being directly applied to CB through quite small resistances, CB stands up against the assault and remains charged so that the B note is sounding.

By the time the B pedal is released, the pulse signals have long since done their work and disappeared. Thus the new B note can now continue to sound in peace for as long as it takes to decay—or until another pedal is pressed and the B is cancelled.

Fig. 11-9 shows the rest of the pedal circuitry. The 8′ signal is taken from the output of the first pedal flipflop and 16′ from the second. The two are equal in

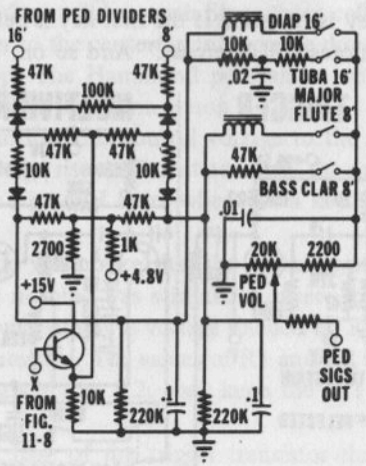


Fig. 11-9—Actual sustain of pedal notes is controlled by diode gates.

level and, of course, are square waves. Each is passed through an identical channel to the inputs of the stop filters. Each channel includes two diodes. Those nearest the tone inputs have cathodes biased somewhat positive from the flipflops themselves; the others are biased off by the 4.8-volt source.

Point X in Fig. 11-8 carries positive potential which is actually the decaying potential of whatever 10-μf keyer capacitor has last been charged by pressing its pedal. The decay curve of that positive potential is determined by the resistor selected by the switch in Fig. 11-8. This same voltage is applied to the base of the emitter follower in Fig. 11-9, whose emitter applies a similar voltage to the four diodes. Since the potential is maximum positive at the time a pedal is

pressed, the four diodes of Fig. 11-9 go fully on at that time, passing signal to the stop filters; and of course it remains at this maximum as long as the pedal is held down. When the pedal is released the potential begins to fade, and so does the tone as the diode conduction decreases. The circuit values are so arranged that the minimum positive potential required for the four diodes of Fig. 11-9 to remain conducting is greater than for the D1 diodes of Fig. 11-8, so that the two channels of Fig. 11-9 stop conducting well before a D1 of Fig. 11-8 stops sending signal to the dividers. The two time factors remain in proportion at all times since both depend on the same resistor selection.

The filters of Fig. 11-9 are, to say the least, elemental. The inductors are not for producing formants; they merely roll off treble good like a series inductor should. The two "reeds" just give a little bite for variety without any attempt at authenticity. The pedal volume control is available to the organist.

Percussion

There is a percussion system in the Thomas Celebrity organ which is entirely separate from the sustain. It can be applied to Solo flutes, Solo nonflutes, or total Accompaniment voices, or any combination of these three groups. When a key is pressed the note comes on immediately and at once begins to fade away slowly. It continues to fade as long as the key is held down, but stops immediately if the key is released. If the first key is held down and a second key is then pressed both notes will "percuss." And so on. The percussion has no voices of its own,

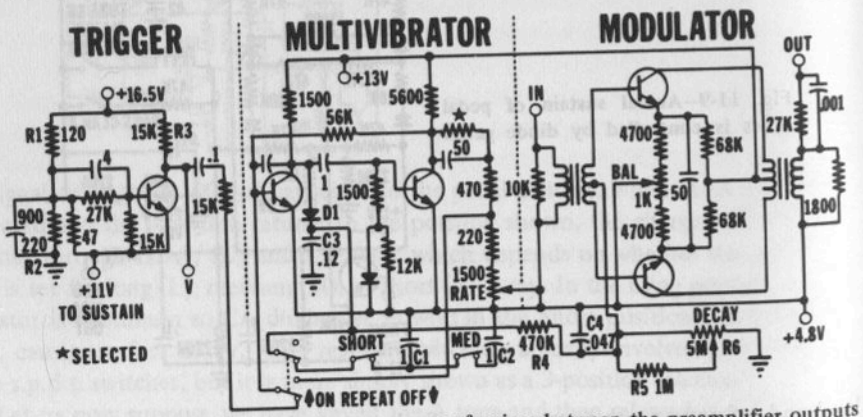


Fig. 11-10—The percussion operates on groups of voices between the preamplifier outputs and the amplifier inputs.

working entirely with the one or more regular groups of stops I have mentioned. It also provides a repeat percussion effect (for the same voice groups) so that whatever keys are held down sound repeatedly, something like the strumming of a stringed instrument.

The percussion works by passing the selected voice groups through a modulator rather than directly to the amplifier-speaker systems. The modulator is simply an amplifier stage which is normally cut off. When a key is pressed the stage is

turned on immediately, after which its conduction gradually stops. When the key is released the conduction of the stage is not stopped, but the tone comes to a grinding halt (just a figure of speech) simply because the key switch is no longer supplying it to the stop filters.

Fig. 11-10 shows all the substantive parts of the percussion. The switches which determine whether tonal groups are fed through the percussion modulator or directly out are not shown. These switches follow three preamplifier stages, one for each tone group. Each preamplifier is fed through balancing potentiometers which adjust the relative levels of the groups. The system holds no mysteries. When a group is to have percussion it is switched to the IN terminal of the modulator in Fig. 11-10 and the OUT terminal goes to the main organ amplifier.

Beginning at the end, let us first look at the modulator. This is a push-pull stage whose transistors are normally biased to cutoff by the positive voltage from the 13-volt source fed to the emitters through the 68,000-ohm resistors. If a sufficient positive voltage is applied via the centertap of the input transformer to the two bases the stage will amplify. If that positive voltage then fades away gradually so will tone transmitted through the stage. The function of the balance control in the emitter circuits is to adjust the two transistors so that the pulses of collector current caused in both transistors by the sudden application of on bias to the input transformer secondary will be equal. Since these collector pulses go through the output transformer to the centertap in opposite directions they cancel and do not produce a pop. The Hammond percussion switcher (Chapter 16) uses a push-pull stage for exactly the same reason.

The task of the trigger stage is to furnish this control voltage to the input transformer centertap—an immediate voltage rise sufficient to cause the modulator stage to conduct fully, followed by a gradual fade—whenever a key of the upper manual is pressed.

Looking again briefly at Fig. 11-7, you will notice that the sustain system is activated by a source of +11 volts, but nothing was said about where it came from. Fig. 11-10 reveals that. The +16.5-volt supply is voltage divided by R1 and R2, with an insignificant 47-ohm series resistor. The values of R1 and R2 are so small that, together with the large bypass across R2, they keep the +11 volts constant.

The same 11 volts is applied to the base of the trigger transistor through 27,000 ohms and 4 μf as on bias, and the +16.5 volts is the B+ for the transistor. You will note, however, that the emitter of the transistor is also held at +11 volts, instead of being grounded as a respectable emitter should.

Glancing again at Fig. 11-7, notice that whenever the sustain contacts of a key are closed the +11-volt point is connected through a relatively small resistor (470 ohms) across a relatively large capacitor (50 μf). At the instant of key closure, current through the 50-μf capacitor is high, limited only by the 470-ohm resistor—so that in effect the +11-volt point is brought within 470 ohms of ground. This lasts only for an instant, of course, for the 50-μf capacitor charges quickly and as soon as it has charged the +11-volt point is as far above ground as

it ever was. But during that instant, when the emitter of the trigger triode of Fig. 11-10 is close to ground, that triode conducts.

When it is not conducting, its collector voltage is the same as the +16.5-volt supply. During the pulse of conduction, however, the collector voltage goes negative because the collector current passing through R3 creates a drop across R3. The net effect, therefore, is that each time a Solo manual key is pressed a negative pulse is created at the collector of the trigger transistor.

Let us assume that the three percussion control switches of Fig. 11-10 are in the positions shown. The negative trigger pulse is then passed through 0.1 μf and 15,000 ohms to the left base of the multivibrator. This multivibrator is not actually multivibrating at the moment because the essential feedback connection which passes through the lower section of the Repeat switch is open, and the feedback provided by the components which are permanently in the circuit is not enough to cause continuous oscillation. However, the negative pulse from the trigger is enough to cause it to flip once and flop once, producing a standard-size positive excursion from the left collector followed by a standard negative one. Only the positive pulse can pass through diode D1. If you will follow it through the closed switches you will find that this positive pulse charges C1 and C2 in parallel to the full value of the positive voltage. The charge potential is connected through a voltage divider consisting of R4 as the series leg and R5-R6 as the shunt leg to the centertap of the modulator input transformer. The voltage at the divider tap is enough to turn the modulator on. R4, R5, and R6 then act in series as a shunt across C1 and C2 to drain away their charge, which they do over a period of time whose length is basically controlled by the setting of R6.

With both Short and Medium switches closed as shown, decay time is longest. If the Medium switch is opened C2 is removed from the circuit. Since the total C of the R-C time constant is reduced by half, the decay takes place faster. If the Short switch is opened, the only C remaining in the time constant is C3, which was formerly so insignificant by comparison that it was not worth mentioning, plus C4, same comment. This gives a very short decay indeed and the sound from each note is just a "plunk."

For the repeat percussion the multivibrator is brought to life by the lower section of the d.p.d.t. switch with the switch arms in the upper "on" position. At the same time the upper section of the switch does just what opening the Short switch did—removes both 1-μf capacitors from the timing circuit so decay is very short. (Obviously decay must be very short or the repeats would not be heard.) The trigger circuit is now in a condition of siesta and all triggering pulses for the modulator are furnished automatically by the multivibrator at a repetition speed determined by the setting of the Rate control.

You may have been wondering about the V terminal connected to the trigger transistor collector. You are about to discover its function.

Vibrato And Vibra-Magic

The vibrato for the tone generators is provided by the circuitry of Fig. 11-11.

The signal is generated by a phase-shift oscillator. The second leg of the shifting network contains an internal potentiometer to adjust basic vibrato speed, plus a tablet-operated switch for normal speed and a slower speed obtained by adding 10,000 ohms to the resistance of the leg. Output of the oscillator is taken from a tap on the collector load. The upper part of the load is bypassed by a 4-μf capacitor, which is too small to short the fundamental but big enough to help eliminate any harmonics which might rear their ugly heads.

The vibrato amplifier stage is, as we have shown, direct-coupled to the oscillator and d.c. stabilizing feedback is provided between the two stages by a split in the d.c. collector load of the amplifier which provides operating potential to the emitter of the oscillator. Note well that this amplifier transistor is a PNP—the voltages shown would not make sense otherwise.

Output of the vibrato amplifier is taken from the collector through an a.c. voltage divider C1-R1-R2-C2. The "lower" end of this voltage divider actually goes to the B—, but this is the same as ground as far as signal is concerned. When the Vibrato switch is closed, signal from the collector is put directly on the vibrato output terminal, which goes to the tone generators. When the Tremolo switch is closed instead, the signal encounters 1500 ohms of series resistance and is thus reduced. The result is not, of course, a tremolo, but simply a reduced-depth vibrato, and like the stuffed shirt I am I should like to protest the misleading terminology. When both switches are open the vibrato line is 1 megohm from the collector and only 4700 ohms from signal ground, so you would be able to detect very little vibrato without a seismograph. The depth control is an internal preset, varying the emitter potential and thus the conduction of the stage.

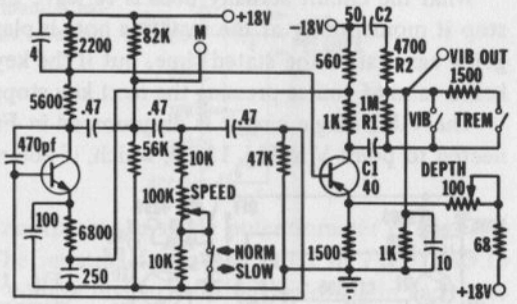


Fig. 11-11—Vibrato oscillator and amplifier feed signal to master oscillators in the tone generators.

Thomas has yet one more trick in its bag called Vibra-Magic. When Vibra-Magic is turned on, there is no vibrato when a key of the upper manual is first pressed. If the key is held down, however, vibrato comes on after a certain interval of time, and it comes on fairly gradually, building up to full intensity. The effect is interesting, though tiring if used much. It might have at least one legitimate justification. When vibrato is on all the time and the organist plays Solo melody which includes notes of both short and long durations, he will very likely strike any or all of the short notes at an instant when the note is at something other than its correct center frequency. In fact, since the frequency varia-

tion imparted by vibrato is of sine shape, the probability of striking any note when the sine wave is on the baseline and frequency is dead center is very small. If the duration of a note is too small to encompass sufficient cycles of vibrato to give the brain the message as to what the average frequency is, the listener receives an out-of-tune impression, especially for notes which happen to come when the frequency is at or near one of its extremes with a heavy vibrato. Vibra-Magic corrects this problem simply by not allowing vibrato to be heard at all until a note has been established and held for a given time.

In actual practice, experience tells me that many if not most players use Vibra-Magic for the bizarre impression it makes—as though a violinist played without vibrato, but on each longer note slowly began to wiggle his fingers well after the note began. This technique, which is looked upon with horror by serious musicians, is extremely common among popular singers. Didactically speaking, it would be better if there were no vibrato on short notes, but a vibrato which began immediately on any note to be held. Since this would require the instrument to read the player's mind, I hold out little hope.

What the circuit actually does is to leave the vibrato running constantly, but stop it momentarily at the instant a note is played. The vibrato then always gets going again after the stated time, but if the key is released soon enough it is not heard—and of course pressing the next key stops it again momentarily.

The Vibra-Magic circuit is diagrammed in Fig. 11-12. The V terminal is connected to point V in Fig. 11-10, which, if you remember, carries a negative pulse

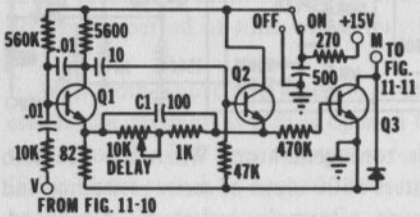


Fig. 11-12—Vibra-Magic turns vibrato off temporarily each time a Solo key is played.

every time a Solo manual key is pressed. Fed to the base of transistor Q1 in Fig. 11-12, this negative pulse appears as a positive pulse at the Q1 collector. Q2 is an emitter follower, so the pulse is still positive at the Q2 emitter.

The Q2 emitter load is an R-C time constant, with C1 immediately charged positive with respect to ground (which it reaches through the small Q1 emitter resistor for a little friendly feedback). The Delay control determines how fast the positive charge leaks off C1. This signal, which is now an immediate rise to a positive peak followed by a gradual fall to nothing, is direct-coupled to the base of Q3. The collector of Q3 goes to point M in Fig. 11-11, from whence it receives its B-supply through the 82,000-ohm resistor and also plants its collector-ground impedance across the first section of the oscillator phase-shift network.

When the Vibra-Magic circuit is turned off by the switch in Fig. 11-12, there is no B+ applied to the transistors, so that the resistance between point M and ground is high enough to have negligible effect. When it is turned on, there is B+, but since the Q2 base is at ground potential there is no conduction through Q2

and thus no positive on bias applied to the base of Q3. Q3 is therefore nonconducting and still does not affect the oscillator. However, when a positive pulse is applied to the Q3 base as explained, Q3 conducts heavily enough to stop the oscillator. As time and a half goes by, the positive Q3 voltage gradually subsides and Q3 eventually becomes nonconducting again, allowing the vibrato oscillator to go to work again. This instant murder followed by gradual revival takes place every time a Solo key is pressed.

Sound Production

Signals from the three tonal divisions—Solo flutes, Solo nonflutes, and Accompaniment—emerge from the voicing circuits and go through balancing controls which allow the player to adjust the relative volumes of the two manuals

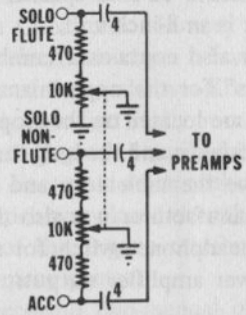


Fig. 11-13—A ganged manual balance control precedes preamp.

when he wishes. The control is continuous, a single potentiometer of two sections as shown in Fig. 11-13. The resistors from each input of Fig. 11-13 to ground form the shunt leg of a voltage divider with the series leg in each case formed by resistors in the voicing circuits; thus, the greater the resistance to ground for each channel in the balancing circuits, the greater the signal output. Operation should be obvious.

Each of the three signals goes to a conventional single-stage voltage amplifier. Following the amplifiers the signals are switched either directly to the inputs of the two power amplifiers or to the amplifiers through the percussion modulator of Fig. 11-10.

There are two identical power amplifiers, one for the Leslie dual-speed speaker. The switching permits a wide range of choices as to which channel(s) sound through main and/or Leslie speakers.

The signal to each amplifier passes through a potentiometer-type swell-shoe control, both controls being ganged. The manufacturer's leaflet speaks of a 150-watt "peak" power amplifier. Whether this refers to each amplifier or both together (much more probable) is not clear, but a peak power rating is meaningless anyway and is just playing the numbers game. A "peak" number can be arrived at by doubling "music power" (or slightly more legitimately sine-wave power). "Music power," a term which I have never been able to bring myself to write without quotation marks, is sufficiently meaningless with respect to radio and phonograph reproduction; it is a term authorized by an association of manu-

facturers to give a surface appearance of legitimacy to unnecessarily inflated ratings placed on their equipment for advertising purposes by these same manufacturers. In association with organs, it has not the slightest semblance of significance, as organ power must be produced for any length of time the player decides to hold down keys and pedals. Peak power could have meaning only if it meant that power at which an amplifier just began to clip. It is not used that way and is therefore nothing more than what some very tolerant and polite people call advertising "puffery."

The presence of two amplifiers and speakers (one of them the Leslie) is referred to as "stereo" sound, but audio people will recognize the inaccuracy of the term, since there is no spatial *distribution* of signals but only placement of certain tonal groups on either one side or the other. The main speaker system is a 15-inch and a 12-inch speaker paralleled, without a dividing network; the Leslie speaker is an 8-inch unit.

This organ also contains a combination action which is purely mechanical. Four "pistons" for the upper manual and two for the lower, plus a cancel for each manual, are located on the stop board above the tablets. These buttons protrude some distance and the long throw this allows works the mechanical system inside to move the tablets up and down. The original combinations can be altered. The manufacturer was also thoughtful enough to provide a headset jack and speaker-headphone switch for silent practice. The headphones work right from the power amplifier outputs, so that modern quality units can be used.

Chapter 12

Lowrey Organs

The Lowrey Company, a division of Chicago Musical Instrument Co., 7373 N. Cicero Ave., Lincolnwood, Ill., manufactures the Lowrey organ, one of the most widely sold brands. The company was started by F. C. Lowrey, a mining and oil man and manufacturer of structural materials, who first became interested in home organs as far back as 1918 when he bought the rights to a keyboard instrument called the Choralcello. Lowrey and James A. Koehl, a patent attorney, sponsored small electronic organ laboratories during the twenties, thirties, and early forties. One of the results was Koehl's ingenious invention of the Lowrey Organo, the company's first really successful musical product, first sold in 1949. The now defunct Organo, described in the first two editions of this book, was in effect a small electronic organ without keyboards. The key switches were embodied in a rail which was placed over the backs of the keys of a piano. After the initial success of the Organo attachment, Janssen, Story & Clark, and Kimball built them into many pianos.

In 1955 Lowrey introduced its first successful two-manual organ, a model called the Berkshire. At the time the company was a division of Central Commercial Company, whose business was entirely unrelated to musical instruments. In 1955 Lowrey died and in 1956 his son sold the organ rights to Chicago Musical Instrument Company, a large manufacturer of guitars and other instruments.

The present line of Lowrey organs presents a wide variety, both in "models" as visible to the naked eye and in internal circuitry. The concentration is on low and medium prices, the models all leaning rather heavily on the special effects which characterize many modern home organs. Some fairly expensive organs are made by Lowrey, but they are few and not in the company's usual genre, which aims at the mass market. Because of this concentration on that section of the public which is culturally and economically the recognized "mass," Lowrey and Thomas have made the principal inroads on the home market which have been responsible for gradually pushing Hammond back from its traditional clear and overwhelming domination of the field.

Since the parent company's acquisition of Lowrey in the middle fifties all Lowrey organs had been manufactured under contract by Hallicrafters Company, the well known maker of radio transmitters and receivers. On February 1, 1966, Chicago Musical Instrument purchased the Hallicrafters organ facility, which is now Lowrey Electronics Co., Chicago.

The Theatre Spinet, Model H, on which we shall concentrate in this chapter, shows much of what is present in most current Lowrey organs and definitely illustrates the trend of thought which sparks Lowrey design. Figs. 12-1 and 12-2

show two other members of the line and Fig. 12-3 is a photograph of the H.

General Description

The Lowrey Theatre Spinet externally conforms to two trends in the electronic organ world. As a "spinet," a member of a class originated by Hammond when it first went to short keyboards, it has manuals shorter than the standard



Fig. 12-1–The Lowrey Starlet is a very inexpensive model.

61 keys. There are 44 keys each, which is just par for the spinet course.* It also has a neo-horseshoe console† in superficial imitation of the theatre pipe organs which reigned during the era of the silent movies. These "horseshoe" consoles are becoming very popular, along with, in some cases, voicing designed to imitate the "mighty Wurlitzer" of yore (no tonal relation to the modern Wurlitzer).

In accordance with old theatre practice, the two manuals are known as Solo (upper) and Accompaniment (lower). The Theatre Spinet has 11 voices on the Solo, seven on the Accompaniment, and, in effect, two on the pedals. It has a great many special effects and devices, which we will go into later.

Functionally, the organ begins with the 12 tone-generator assemblies customary for frequency-divider organs. The tones are fed directly to the lower manual, where they are keyed by switches, and to diode keyers controlled by the upper manual. All tones then go through a series of tablet controlled voicing filters, thence to power amplifier and speakers.

Main Tone Generators

The 12 main tone generators produce about 4½ octaves, from F2 (87 cps) to C7 (2093 cps). This provides the minimum necessary number of pitches for the

*A spinet was actually a harpsichord with the strings running at an angle to the keyboard. We now have spinet organs and spinet pianos, and the word has come to mean "smaller than standard" or words to that effect–generally, in organs, at least, with abbreviated keyboards.

†The old theatre organ had an almost semicircular stopboard because it had so many stop tablets that if it had been straight the organist would have needed telescopic arms to reach the ones at the ends. Almost no modern electronics have real need for the curve; it is almost always just a stylistic imitation. In the theatre the organist usually sat inside the semicircle and thus needed the "crazylegs" bench. This, too, in the modern case, is just style, as the modern bench is outside the smaller console and could just as well have a full-length top. And of course its legs need not spread like this to span just 13 pedals!

upper manual, the only one which has stops at 16′ pitch. The upper manual has the span most usual for spinets–F below middle C to the C three octaves above middle C. The lower, however, begins with the second A below middle C, rather than the more usual second F, and goes up to the third E above middle C rather



Fig. 12-2–Festival Deluxe is one of the larger Lowrey organs with 61-key manuals.

than the second C. This places the lower manual four semitones higher than in most spinets. It has only 8′ stops, plus a 4′ coupler, so its range does not affect the lowest manual pitch required from the generators.

It is interesting to note that this slightly odd disposition of the two manuals comes about for a reason you would probably never guess. Most if not all Lowrey spinet manuals are made by sawing apart a standard 88-key piano keyboard,



Fig. 12-3–The Lowrey Theatre Spinet described in this chapter has a curved stop board.

between the E and the F! Thus mundanely are the world's decisions made.

Since the upper manual has a 16′ register, the lowest tone needed from the generators is that required to produce 16′ pitch from the F key whose unison pitch is 174 cps. The highest generator frequency of 2093 also represents maximum economy, as it just allows the top C of the upper manual to produce its 8′ tone. In the top octave, therefore, 8′, 4′, and 2′ tones would be all the same if it were not for the ingenious frequency "doubler" used at a later point to derive an extra octave so that only the 4′ and 2′ tones are the same. It must still, of course, seem that these are extremely stingy generators but it is Lowrey habit to stint on tone generators (and organ-type voices) so that money can be spent on special effects while the price of the organ remains low.

Each of the 12 generators consists of a master oscillator and either three or four flipflop dividers—three for C#, D, D#, and E, and four for the rest. In Fig.

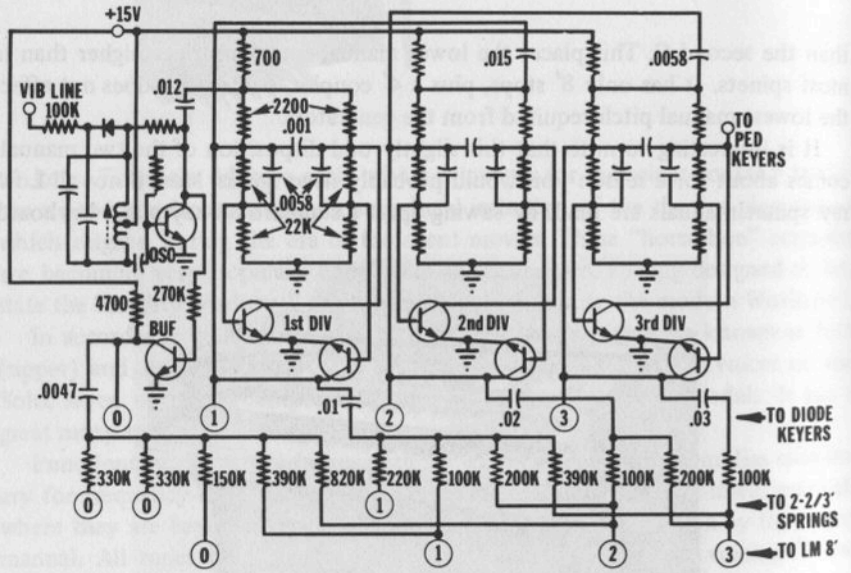


Fig. 12-4—The Theatre Spinet tone generator, one of 12. The divider components above each pair of transistors are contained in a packaged circuit, all of them the same.

12-4 appears the schematic for one of the smaller generators. The others are the same with one additional flipflop.

I have redrawn the oscillator in Fig. 12-5 to make it easier to understand. It is essentially a slug-tuned Hartley, and it conforms to the general rule that every designer has his own little touch of originality when approaching a Hartley.

The transistor is forward biased by the 220,000-ohm resistor, which derives its positive voltage (these are NPN transistors) from the collector rather than from B+, a form of negative a.c. and d.c. feedback tending to stabilize the stage. The collector load is part of the tuned circuit (that section of the coil at the right in the drawing) in series with 10,000 ohms. Since the organ has the custom-

ary low-impedance power supply, ground and B+ may be considered common for a.c. so that the centerpoint of the coil and the emitter are connected to the same a.c. point and the load properly appears between collector and emitter. Ignore the rectifier and the .039-μf capacitor for the moment.

The signal appearing in the collector part of the coil is transferred to the left portion by mutual inductance. This left portion is connected from base to emitter (through the .0068-μf d.c. blocking capacitor) and thus furnishes input for the transistor. An arrangement like this, with two parts of the coil transferring signal through mutual inductance, is regenerative (has positive feedback), so that it oscillates. The capacitor values shown are those for C# and D; they change for other notes.

A d.c. blocking capacitor for the transistor base is usually found in Hartley organ oscillators, even those with other configurations. Ideally, this capacitor should merely block d.c. and should have too low a reactance to affect frequency. Unfortunately, a capacitor large enough not to affect frequency, even when

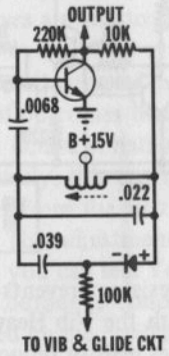


Fig. 12-5—Master oscillator, redrawn to show how circuit works.

the oscillator runs an octave or two above those in the Lowrey organ, runs into microfarads and has to be electrolytic, which costs something. A 5-μf unit is used in the Schober Organ of Chapter 10, for example. The other approach, typified in the Lowrey generator, is to go ahead and use a small value (the .0068 in this case) and treat it as a frequency determining component, scaling it as necessary and saving a few cents per oscillator. I can think of nothing wrong with that except that the fewer the components which affect frequency, the more stable the oscillator is likely to be. Such a thought is more theoretical than practical, however, as long as a ceramic capacitor is avoided, since other common types like paper and Mylar are negligibly affected by heat and time.

The 10,000-ohm resistor between collector and tank serves to throw away some signal and bring loop gain to the reliable minimum, and also to avoid shunting the tank directly with the relatively low transistor collector impedance, reducing its Q. I have often thought how lucky it is that resistors, the most plentiful components in electronics, are also the cheapest, so that a good designer doesn't spend too many hours considering economies before throwing in an extra resistor if it will do some good!

In most frequency-divider organs vibrato is applied directly to the master

oscillators—by various methods—and this is where the diode comes in. Look at Fig. 12-6, the vibrato circuit. The basis is the phase-shift oscillator consisting of the first transistor with the conventional three series capacitors between collector and base, with resistance between each capacitor junction and ground. The first resistor is a variable one for setting speed, paralleled with a switch-operated resistor which enables instant speed change with a tablet. Closing the switch raises speed. The final resistor is the input resistance of the transistor base.

Oscillator signal goes through an emitter follower and a d.c. blocking capacitor, through a pair of tab-operated switches to a load of 10,000 ohms, with a 1-μf capacitor across it. The capacitor is a low impedance at tone oscillator frequencies and thus shorts the vibrato line to ground for audio, which might otherwise leak into voicing wires along the way. It also filters out the tick caused by the sudden d.c. change when the Glide switch is used. With the Vib On switch open,

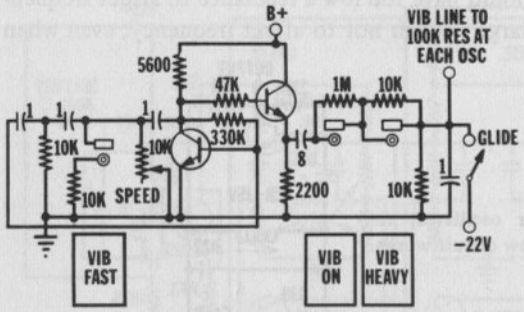


Fig. 12-6—The vibrato generator and control circuit. Boxes representing tablets here and in subsequent drawings do not necessarily show actual lettering appearing on organ, nor do contact springs necessarily move in same directions as in the organ.

the 1-megohm resistor prevents any but a negligible oscillator signal from reaching the load. With the Vib Heavy switch open (but the Vib On switch closed), a 10,000-ohm series resistor reduces signal to the load.

The resistor-capacitor oscillator load is also connected to each of the 12 tone-generator oscillators through a 100,000-ohm resistor. There is one resistor per generator and it is shown in Figs. 12-4 and 12-5. Fig. 12-5 shows that the anode of the diode is biased positive by connection through the tuning coil to B+. The cathode goes through the 100,000-ohm resistor, through the 10,000-ohm load resistor in Fig. 12-6 to ground. The diode thus conducts somewhat at all times, though not heavily because of the current limiting effect of the high resistance. As the vibrato oscillator causes the potential at the common vibrato line to rise and fall in a sine fashion, the conduction of all the 12 diodes varies. In effect, the diode in each oscillator is a variable resistor.

In series with this variable resistor is the .0068-μf capacitor (Fig. 12-5); and both are across the tuning coil. The capacitive reactance appearing across the coil as the result of the capacitor varies in accord with the vibrato oscillator frequency, and this, of course, causes the master oscillator frequencies to vary smoothly up and down, creating the vibrato effect. One might wish to sound elegant and call this a reactance modulator, which indeed it is. The reactance of the diode-capacitor combination varies and modulates the frequency of the master oscillator.

One more item appears in Fig. 12-6. One of the famous Lowrey effects is the "glide."* When the Glide switch is closed, a husky −22 volts is applied to the common vibrato line. This immediately causes maximum conduction of all the diodes and consequently lowers the pitch of each oscillator about a half tone. The switch is attached to the left side of the swell shoe so that it can be operated by pivoting the foot slightly on the heel without removing it from the swell shoe. If you press the switch at the same instant a key goes down, and then release the switch immediately, it gives an effect reminiscent of a trombone "smear" or a Hawaiian guitar slide.

Returning to Fig. 12-4, the flipflop frequency dividers are conventional, as explained in Chapter 3. All of them are identical except that the trigger capacitors coupling one divider's output to the input of the next vary. All resistors and capacitors for each stage are contained in a packaged circuit.

Each generator has two kinds of outputs. One type comes directly from each flipflop and consists of the square wave produced by the flipflop—the same one used to trigger the next stage. These square waves are fed to the diode keying system controlled by the upper manual.

The second group of outputs, used for the direct keying of the lower manual, is the result of staircasing. The output of a flipflop comes through a 100,000-ohm resistor. To it is added tone from the next octave at half voltage and from the third higher octave at quarter voltage. Obviously, this can be done only for the fourth, third, and second divider tones. That from the first divider has two components, and from the oscillator just one. The adjustment of component voltages is by choice of series resistor values, as you can see. You will note that the resistors are different where the oscillator output is concerned, since the output transistor in that case is a buffer stage fed by the oscillator, and its basic output voltage is different from that of the flipflops. The purpose of the buffer is to provide isolation between keyers and master oscillator. The oscillator output waveform, by the way, contains over 50% second harmonic, which makes it possible for Lowrey to "generate" an extra octave of tone with a unique "doubler" system, of which you will read later.

In addition to the staircased outputs, there are some additional outputs marked $2\frac{2}{3}'$, which are simply square waves taken through resistors. These go to a set of upper-manual contacts which key the $2\frac{2}{3}'$ register directly.

Lower Manual Keying

The keying circuits are easily the most complicated section of this organ; they are complicated even by comparison to *most* keying circuits, which tend to be cluttery simply because each key usually has multiple contacts and there must be as many sets of contacts as there are keys. In most organs, however, you can establish a basic scheme and carry it out right up the keyboard. Here there is a great deal of irregularity, largely because there is an absolute minimum number of pitches and there are pitch registers calling for many more under a "straight"

*Wurlitzer has it now, too, calling it "slide." See Chapter 14.

scheme. This necessitates lots of jumpering and cross-connecting. Other special provisions of this organ conspire, too, to complicate matters. So to get a really complete picture of just what is happening in detail would require more space than is available. This book is intended to teach, not as a service manual for any organ; if you have any business servicing a Lowrey organ you will be able to get the set of diagrams which shows everything. We will, therefore, show you enough to make it clear how the design objectives are reached, but we will mostly ignore incomplete and repeated "ranks" and the routing details of exactly what each wire is connected to.

The lower manual has four ranks of contacts, each of which, like all the key contacts in the organ, is a spring which is moved into contact with a stationary contact or bus when a key is pressed. To make things clearer from here on, we shall refer to individual moving contacts as *springs* and to the pieces of metal with which they come into contact when a key is pressed as *contacts* when the stator is individual to one spring or *busses* when the stator is contacted by several springs.

One complete row of springs is provided for the basic 8′ tone keying. Each of these springs is connected to the staircased output of a generator—the output of the appropriate 8′ pitch for the key.

We have shown this in Fig. 12-7. Under the middle C key is the 8′ spring connected to the appropriate output on the C generator. Above the spring is a contact against which the spring rests when the key is *not* played. (Note that our

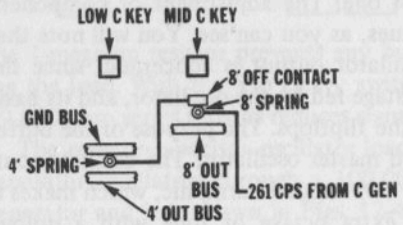


Fig. 12-7—Basic keying system for lower manual. Both 8-foot and 4-foot springs actually exist under each of the keys.

diagrams are purely schematic. Physical placement of springs in the organ may be above when we show below, etc.)

A second row of springs under each key (except the top octave) is represented in Fig. 12-7 by the 4′ spring lower and to the left, under the C key an octave lower than the middle C. Actually, each of the keys has both 8′ and 4′ springs; we are showing only one of each. Assume that here—and in all our illustrations of keying on this organ—a spring shown against its upper contact or bus is *off*; that is, it is controlled by a key that is *not* being played.

With neither of these C keys pressed, the generator output is grounded through the 8′ spring, 8′ off contact, 4′ spring, and ground bus. This grounding is an important point. Referring back to Fig. 12-4 for a moment, notice that the outputs of the staircasing networks are not terminated to ground. As discussed in Chapter 3, this would cause coupling between stages. Grounding each output in the switch system avoids this; and when the ground is removed, the impedance of following circuitry is low enough to supply the missing termination.

Returning to Fig. 12-7, if the middle C key is pressed the 8′ spring moves down and touches the 8′ output bus, from which the tone is connected to following filter circuits.

If the low C key is pressed, the 4′ spring descends and strikes the 4′ output bus, which carries the signal to the following filters.

It is important to note that the lower manual has only 8′ stops. The 4′ provision is strictly an intramanual coupler; that is, it adds tones an octave higher to the standard 8′ pitches, but with the same tone colors as the 8′ stops in use. This explains what you have undoubtedly noticed—that if both the C keys are pressed, only the 8′ output bus receives tone; there is none for the 4′ output bus. Well, none is needed. Think of it in terms of a rank of pipes, normally played at 8′ pitch but with a 4′ coupler. The object of a coupler would be to hear both the low C and middle C pipes at the same time when the low C key is pressed. You key the low C and the electrical system opens the valves of both pipes; they both sound. Now you also press the middle C key. Since both pipes are already sounding, pressing middle C can add nothing more (except sound from the pipe still an octave higher, which we are not considering). Or to say it another way, it makes no difference whether we cause both pipes to sound by pressing only one key or by pressing two. The sound is the same.

The 4′ coupler rank of springs is not carried all the way to the top of the keyboard, since the generating system runs out of pitches. In the top octave nothing happens when the 4′ coupler is on. Obviously the 4′ coupler does not work all the time. A tablet is provided. It switches 4′ output bus tones into the same filters as are fed by the 8′ output bus.

With a keying system of this kind, any given staircased generator output can be used for only one pitch register (aside from this sort of coupling) and on one manual. If, as on some Lowrey models, a given generator output is to be directly keyed at more than one pitch register or on both manuals, a separate staircased output must be provided for each use.

Upper Manual Keying

All 16′, 8′, 4′, and 2′ keying on the upper manual is done through diode keyers so that sustain, repeat, and attack control will be possible. (There is none of this for the lower manual.) There are 11 sets of springs and busses for the upper manual, of which 10 are for Automatic Orchestra Control (AOC) and the $2\frac{2}{3}$′ register. We will ignore these 10 for the moment and concentrate on how the basic 16′, 8′, 4′, and 2′ tones are keyed.

The basic Lowrey diode keyer arrangement appears in Fig. 12-8. The generator outputs used are taken directly from one collector of each flipflop (or from the oscillator buffer) and thus contain something under +15 volts d.c. as well as the square-wave signal. The signal is permanently connected to the cathode of diode D1. The output path, when the diodes are rendered conductive, will be through D1, through D2 and its following 22,000-ohm resistor, to an output bus which is common to several notes of the particular pitch register. This bus terminates in a resistance to ground which is in the area of 1,000 ohms. The termin-

ating resistor has been shown dashed to indicate it is not physically a part of the keying circuit.

Five tablets control the keying. All of them are shown in Fig. 12-8, all in the off position. That is, all sustain, repeat, and reverberation are *off*, and the attack switch is shown in the *fast* position. Keying is normal, therefore, as shown in the diagram, with no special effects. The keying spring, which is operated by an upper-manual key, is also shown off; the note is unkeyed.

There are two lines labeled *sustain* and *bias* which carry fixed d.c., depending on tablet settings. As things stand at the moment, −15 volts is applied to the keying spring and −20 to the upper end of the 40-μf capacitor C1 through 56,000 ohms. Both diodes are therefore very definitely biased off.

When the key is pressed, the keying spring leaves the upper contact and strikes the lower one. This immediately applies +15 volts to resistor R1. At the point where R1 and R2 join the positive voltage takes over and the −20-volt bias is ineffective because R2 is relatively large. The positive keying voltage

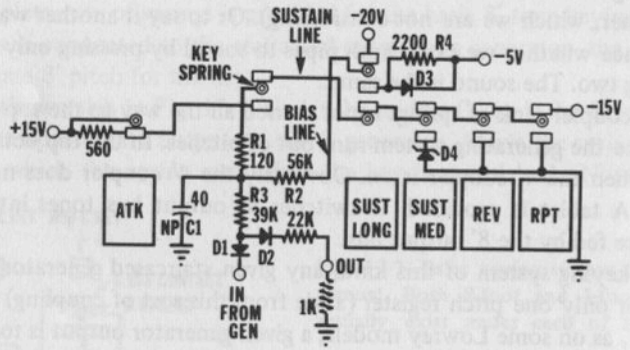


Fig. 12-8—Diode keying system used for all upper-manual pitches but the mutation.

charges C1 through R1 in a very short time (not more than 30 milliseconds), and the +15 volts is applied through R3 to the anodes of both diodes.

Since the cathode of D2 is at ground potential, the +15 volts causes it to conduct. As for D1, the voltage on its cathode from the generator changes, at the repetition rate of the generated square wave, from something approaching the generator B+, 15 volts, to a lower value representing the negative excursions of the square wave. Since the cathode is therefore always more negative than +15 volts, the keying voltage applied to the anode keeps it conducting. The generator signal thus passes through both diodes to the output. R3 is present simply for isolation. For d.c. operation both diodes could be connected directly to C1, but insertion of R3 prevents the control circuit from shorting out the audio signal.

When the key is released and the keying spring returns to the upper contact, −15 volts is again applied to C1 through R1. C1 thus very quickly discharges and recharges negative, and the diodes are cut off.

Attack is varied simply by inserting a 560-ohm resistor into the keying-voltage line; when this tablet is on the switch opens. The net voltage reaching C1 is still almost the same, so that the diodes turn fully on. However, the time required to charge C1 to full keying voltage is multiplied over 5 times, since the R of the time constant (C1 and the resistance in series with the +15 volts) has risen from 120 to 560 + 120 or 680 ohms. The resulting tone buildup time with slow attack is in the neighborhood of 1/5 second, which is appreciable to the ear.

The time required for the tone to decay after the key is released depends entirely on how fast the positive charge on C1 is dissipated and a negative charge is substituted. By controlling this the design provides a sustain effect of three different durations, so that the note dies away relatively slowly when the key is released rather than immediately. Sustain is controlled by the three tablets we have not yet discussed. (The Repeat tab is not related to any of this and its function will be discussed when we get to the repeat circuit.)

When the Sustain Long tablet goes on, its lower contacts open, so that C1 can no longer discharge through R1 and the upper contact of the keying spring. The upper set of contacts of the tablet also causes −5 volts to be applied to C1 through a few components.

The purpose of R2 is to change the positive charge on C1 to a negative one at a slower rate than the keying voltage changed the C1 charge to positive—that is, to cause the tone to decay more slowly than its attack. The large value of R2 does this slowing job because it makes for a large R-C time constant, and the actual decay time also depends on whether −20 volts or −5 volts is applied to R2. D3 is provided to prevent the −20 volts present on all the C1's from getting to the −5-volt source when the Sustain Long tablet is turned on.

If the Sustain Long tablet is down, obviously none of the remaining ones has any effect, since the bias is set to −5 volts and the sustain line is disconnected. Let us now assume that the Sustain Long tablet is off and the Sustain Medium tablet goes on. In the diagram, this means that its spring is in the down position, and that the bias line is again connected to −20 volts.

When the key is released, C1 has a positive charge, placed there during the on time. C1 is now connected through the off contact to the top of zener diode D4.

This zener diode is chosen for a zener voltage considerably less than 15. Therefore, when the key is first released, C1 discharges through D4 because its positive charge is higher than the zener voltage. The only limit on the discharge rate is R1, which is not much, so that the discharge and the consequent decay of tone is rather rapid. However, when the charge on C1 has decayed to the point of the zener voltage, D4 ceases to conduct and there is no further way for C1 to discharge via the sustain-line route. At this time the −20-volt bias takes over and slowly causes the rest of the turnoff cycle. Even in this second portion of the cycle, the rate of turnoff is faster than it was in the long-sustain mode, since the bias voltage is now −20 instead of −5. But again, though in a different way, the same idea of fairly rapid initial turnoff followed by extension of the decay at the tail end has been attained.

The so called reverberation introduced by the Reverb tablet in this organ is simply a short sustain, as is not uncommon in the industry. The effect does not

simulate a genuine reverberation, but it does help to mitigate the effects of a dead room.

The short "reverb" sustain is obtained here simply by grounding the discharge path for C1 when the Reverb tab is on with the preceding two off. This makes a fairly rapid decay of tone, but not as fast as with all the tablets off, when −15 volts is fed to C1.

When the strumming "repeat" effect is on (description later), it causes notes to go on and off at a rapid rate as long as a key is held down. When the key is released, the final decay is the same as for "reverb" since the Repeat tab grounds the sustain line just as the Reverb tab does.

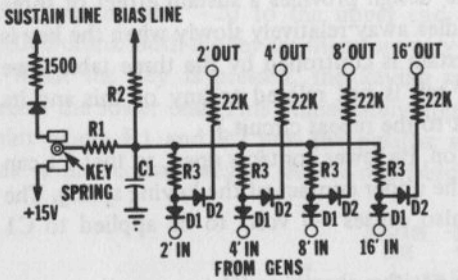


Fig. 12-9—This is the actual group of keyers associated with one key to control the notes of four pitch registers.

What we have shown and explained is the diode keying for a single pitch. Actually, of course, pressing almost any key causes keying of several pitches since there are four registers.

Fig. 12-9 shows what happens when a single key switches notes of several different registers. There is only one keying spring, R1, R2, and C1 for each key. These are connected to as many individual diode keyers as are to be operated simultaneously, each with its own D1, D2, R3, and output resistor. Where a single pitch is used in several places—for instance as the 8′ pitch for one key and the 16′ pitch for the key an octave higher—the appropriate generator output is simply connected to more than one diode keyer.

Note that the actual connection of each off contact to the sustain line is through a diode and 1500-ohm resistor. This prevents any positive voltage from any one of the charged C1's from getting to the off contact of another key and turning on diodes associated with that key. Any such positive voltage will get into the sustain line through that key's own diode but cannot get to another key through the other key's diode.

To dispose of a second one of the 11 rows of springs for the upper manual, a $2\frac{2}{3}'$ pitch register is directly keyed by springs and contacts just as the lower manual is keyed. Generator tones are taken from the flipflop and buffer collectors (without any staircasing) through the same d.c. blocking capacitors as are used for the lower-manual staircased outputs.

Intermanual Coupler

The organ has an intermanual coupler, a Solo To Accompaniment tablet. When this tablet is turned on and the lower manual is played, any upper-manual

stops and pitch registers which are on will be heard, along with the regular lower-manual stops.

The coupler has a special rank of contact springs operated by the lower-manual keys. Fig. 12-10 shows how the middle C works. Note first the same upper-manual keying spring and associated on and off contacts shown in Figs. 12-8 and 12-9. As we showed in Fig. 12-9, the sustain bus is connected to the off contact through a 1500-ohm resistor and a diode, and the on contact carries +15 volts keying potential. In Fig. 12-10 we add two more connections to the off contact. The first is the AOC, which we have put there just for completeness and will explain separately in the next section. The second is the intermanual coupler.

The lower-manual keying spring shown in Fig. 12-10 is one of a complete rank of these special springs. When the Solo To Accompaniment tablet is turned on, +15 volts is placed on the bus. When a lower-manual key is pressed, its coupler spring comes in contact with the bus and places the +15 volts, through a diode, on the off contact of the corresponding upper-manual keyer spring. This overcomes the negative voltage, if any, present on the off contact through the sustain line, and the diode keyer is turned on just as though the upper-manual key had been pressed. When the lower-manual key is released, the sustain provisions take over just as though the upper-manual key had been pressed and released. Thus the coupling is complete, including whatever sustain setting may be on the tablets at the moment.

The diode through which each coupler spring is connected to the upper-manual contact is necessary to prevent interaction when the Solo To Accompaniment tablet is off and the line to +15 volts is open. If then two lower-manual keys were pressed, both of which have coupler springs contacting the bus, the coupler connections to two upper-manual keying circuits would be simply shorted. Any positive voltage appearing on one, as the result, for example, of the AOC, which works by passing positive voltage to a keyer through the diode shown, would be passed on to the other note. The diodes prevent this, since no positive voltage at an upper off contact can pass through the AOC or coupler diodes because of their polarity.

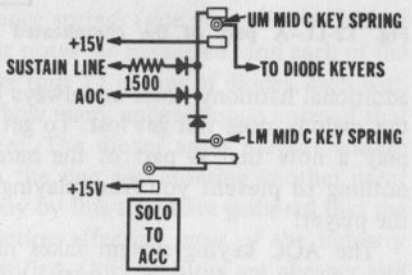


Fig. 12-10—Diagram shows how the Solo To Accompaniment coupler tablet functions.

Automatic Orchestra Control

One of the major differences between a beginning player and one of some

accomplishment is the way the upper manual is played. The beginner plays the melody "one-finger style"—one note at a time—while he plays chords on the lower manual. The better organist, however, uses his whole hand on the upper manual, playing harmonies on it as well as the lower. The result is a much fuller sound.

Automatic Orchestra Control is a rather misleading, not to say grandiose, name for the circuitry which allows you to play with one finger on the upper manual and have the harmonies you are playing on the lower duplicated on the upper along with the melody notes. Forgetting about the name, the effect is musically good and the idea and execution highly ingenious. I can testify from experience that AOC, while hardly necessary or even very useful for the experienced player (meaning me—notice that I said experienced, not accomplished), is an intriguing toy.

Musically it works like this. You play one note at a time on the upper manual and hold chords on the lower. For example, suppose you are holding a C major chord C-E-G on the lower. Now you play a C on the upper. You will also hear an E and a G on the upper manual, just as though you had pressed the keys. The

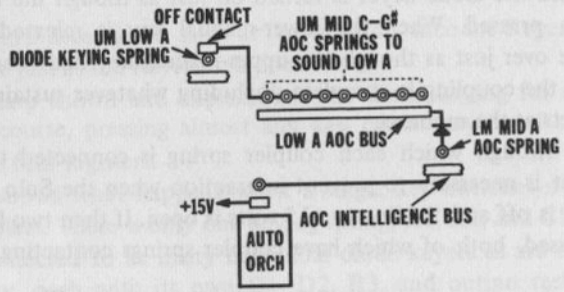


Fig. 12-11—A part of the complicated Automatic Orchestra Control keying system.

additional harmony notes are always lower than the one you actually played so the melody does not get lost. To get the AOC effect you need not necessarily play a note that is part of the harmony—any note will do; in fact, there is nothing to prevent you from playing discords, but something has to be left to the player!

The AOC keying system takes nine ranks of keying springs in the upper manual and one in the lower. The lower-manual springs are aptly labeled "AOC Intelligence," for they tell the system what notes you are holding on the lower manual. The nine ranks on the upper act on that intelligence.

Fig. 12-11 makes AOC as simple as it can be. The AOC intelligence bus has a spring for each lower-manual key. Remember that this spring and the bus are entirely independent electrically of the tone keying and coupling systems we have spoken of before. The lower-manual middle A key is unpressed as shown in Fig. 12-11. With the Automatic Orchestra tab on, +15 volts is placed on the bus. When the middle A key is pressed and held, the +15 volts is transferred to a short bus of the upper manual. This bus has nine associated springs, which

contact the bus when any one of nine keys—middle C through middle G#—is pressed. All nine springs are connected together (though each one can move individually), and all are connected to the off contact of the low A key on the upper manual. The connection and diode labeled AOC in Fig. 12-10 is a representation of the system of Fig. 12-11.

Suppose the Automatic Orchestra tab is on, its switch closed. Now, when the lower-manual middle A key is pressed, +15 volts is placed on the upper-manual low A bus. So far nothing has happened that the ear can detect. But now suppose you press one of the upper-manual keys from middle C to G#. First, you will hear whatever note you pressed, through the normal action of the diode keying circuits of Figs. 12-8 and 12-9, not shown in Fig. 12-11. In addition, you will now hear the upper-manual low A, just as though the low A key had been pressed. This is the essence of AOC.

To cover the bare bones of the idea with some flesh you must now imagine several things which I cannot diagram in full because it would take too much space. First, the springs for *all* the A keys on the lower manual are electrically connected, so that when *any* lower-manual A is pressed +15 volts goes to four diodes which control AOC for A in its four pitches.

Next, you must imagine a network of short busses for the upper manual similar to the low A bus shown. All are 9 notes long except those at the treble end, which are all different. They are end to end and also stacked and staggered vertically. For instance, the first complete row has a low A# bus with 9 springs, a middle G bus with 9 springs, and a high C# bus with 8 springs. There are nine such rows and 41 busses and spring groups. I will leave it to your imagination or future perusal of Lowrey service diagrams how they are all arranged for the desired effect. Obviously it can be done, and the details add nothing to an understanding of the basic principle. The only thing to note is that between each of the 12 groups of lower-manual intelligence springs (one group for each note of the chromatic scale, in which all similar notes are paralleled) and each of the 41 upper busses there is a diode. There are thus 12 groups of diodes, each with either three or four diodes, depending on how many upper-manual notes of that name are to sound as part of the system. The diodes again prevent voltage applied to one point from backing down the line and entering another point where it has no business appearing. You may by this time have gathered that the Lowrey people are diode-happy, an affliction affecting most of the industry today, and one which is very strongly justified. Only resistors are cheaper and more useful than diodes. In good quantities you can buy diodes for three cents or so. Somebody up there likes us!

Arpeggio

The Arpeggio effect is closely related to the AOC and it is obtained by adding nothing more than a tablet and its switching contacts.

The Arpeggio tablet does two simple things. First, it turns on the AOC even though the AOC tablet may be off, by effectively shorting its contacts. Second, it removes the keying potential from the upper-manual diode keyer busses.

The total effect is to have AOC but no normally operated melody notes. That is, when the upper-manual keys are pressed, you will hear notes corresponding to any you are holding on the lower manual, but not necessarily the note of the key you are actually holding down.

Suppose you hold a C chord on the lower manual. Now you just run a finger up the keys of the upper in the manner of a long glissando. You will hear the C's, E's, and G's of the C chord following you up the scale, and nothing else, just as if you had manually played a long arpeggio. As with normal AOC, the voices and pitch registers heard will correspond to whatever upper-manual tablets are on, since the normal diode keyers are being keyed and feeding their normal voicing networks. (For some reason, the Owner's Manual says the normal voices are cancelled; this is an error.)

Repeat Percussion

One effect remains to be discussed before we go on to the unique Lowrey voicing system. This is the repeat percussion, a tablet which causes the notes to

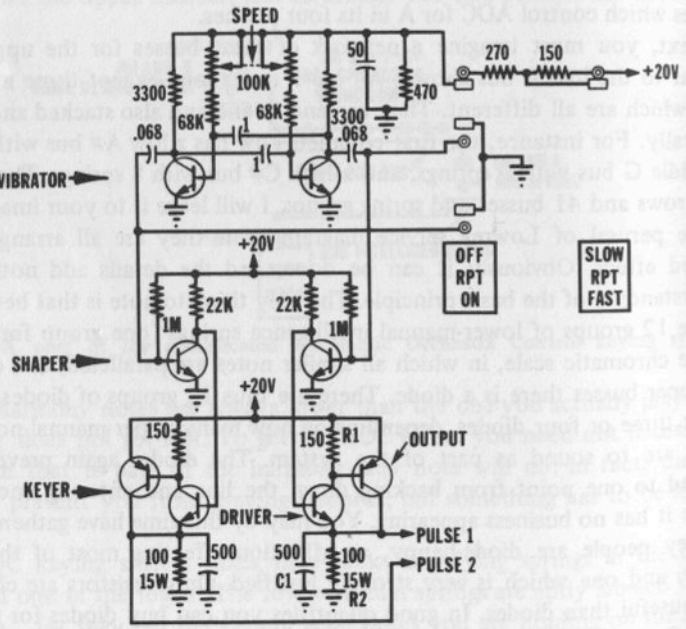


Fig. 12-12—Repeat generator consists of multivibrator, shaper, and keyer, all symmetrical.

go on and off alternately at a rapid rate, like a marimba or xylophone. The repeat is effective on all upper-manual diode-keyed pitches and also works with AOC and the Solo To Accompaniment coupler.

The basic idea of the repeat is very simple. When the repeat is off, the +15-volt keying lines shown in Figs. 12-8, 12-9, and 12-10 are just as indicated, lines carrying a constant +15 volts which, when contacted by key-operated springs,

turn on the diode keying circuits. However, when the repeat generator is turned on, these same +15-volt lines carry a voltage which changes from +15 volts to zero and back several times per second—in other words, a square wave with 15-volt positive peaks and zero-volt negative peaks. If, when this is happening, a key is held down, the associated diode keyers, instead of receiving a steady 15 volts, receive a pulsating voltage. The notes are not heard continuously but go on and off at a regular rate giving an effect like strumming.

Obviously, the only significant circuitry required for the repeat effect is that which causes the +15-volt lines to pulsate. This is the repeat generator, diagrammed in Fig. 12-12. All sections of the repeat generator are symmetrical, so to simplify things we will speak only of one side of each section, allowing you to assume reciprocal action on the other side.

You will be less than amazed to find that the basis of it all is the multivibrator composed of the top two transistors and their components group. Oscillation rate could be controlled by varying almost any component. In this case, the base-B+ resistance is varied by a two-gang potentiometer which varies both the time constant of the coupling between each collector and the other base and the bias.

The multivibrator is controlled externally by two stop tabs. The off-on tab, whose switches in the diagram are shown in the on position, removes the B+ and shorts the two outputs to ground when the tab is moved to off. The B+ contact alone would be quite sufficient to turn the multivibrator on and off. However, the remainder of the system is direct-coupled and grounding the bases of the shaper transistors is necessary to put 15 volts on the output lines (pulse 1 and pulse 2) when the repeat is off. This tablet also has another switch, omitted here for economy but shown in Fig. 12-8. It sets the sustain system for a small amount of decay delay when the repeat is on. Notice from Fig. 12-8 that that switch only affects the decay after the key is released because it is connected to the keyboard off contact. It does not delay the tonal decay between repetitions, since the key is held down to get the repeat effect. A small delay between repeats is achieved otherwise, as we shall see.

Two multivibrator repetition rates are available with use of the Fast-Slow tablet, which is shown in Fig. 12-12 in the Fast position. In the Slow position the contact opens and additional resistance is inserted in the B+ line—150 ohms. At the resulting lower B-voltage, the multivibrator rate decreases somewhat.

The shaper stage is simply a clipper fed through a differentiating capacitor, and gives a pulse output.

The keying section at bottom is composed of two stages, a driver, which is a pair of small-signal NPN transistors about like the others used in the organ, and an output stage containing a pair of 2N555 power transistors. Power transistors must be used to provide the keying voltage, not principally because of heavy current drawn by the diode keyers, but to maintain a low source impedance and avoid voltage drops proportional to the number of keys in operation at any moment. In a large sense, the output transistors are voltage regulators which obey the orders of the drivers at all times despite load alterations.

To understand the keying section more easily, let us assume that the repeat tablet is off. The multivibrator is stopped and both shaper bases are grounded. Since bias has thus been removed from the shaper bases, the shaper transistors are practically cut off, and nearly the entire B-supply of +20 volts appears at their collectors.

This high positive voltage is placed on the driver bases, causing the drivers to conduct heavily. A considerable drop therefore appears across R1 and across R2 as well as some across the transistor.

If we consider R1, the transistor, and R2 as a voltage-divider system between B+ and ground, we can see that the output transistor, which is a PNP device, is correctly connected for electrode potentials. Its emitter is attached to +20 volts and its collector is connected to that point in our voltage divider which gives the voltage most negative with respect to the +20 line other than ground itself. The output base is connected to a point which is also negative with respect to the B- line (and the output emitter), but not so negative as the output collector connection. The output transistor therefore has the right polarities and has considerable on bias, so that it conducts nicely.

The collector load for the output transistor is R2, which happens to be the same resistor as the driver emitter resistor. The actual current through it is mostly due to the output transistor because the latter is a power transistor and conducts much more heavily than the driver. The drop across R2 as the result of output transistor conduction is 15 volts, and the 15 volts goes out through the line labeled Pulse 1, which connects it to the keying busses for use in keying the diode systems.

The current drawn by the keying circuits in the keyboard varies with the number of keys being pressed and with the use of AOC and the intermanual coupler. If this were allowed to cause a variation in the 15-volt keying line, the result would be variations in note volumes. The 15 volts must therefore be kept constant regardless of load.

An increased load draws more current. It therefore has the same effect as if R2 were made smaller. If R2 became smaller, a given current would cause less voltage drop across it. Therefore, it would become less positive (or more negative) with respect to ground. This would be equivalent to making the driver base go more positive, increasing driver conduction. Increased driver conduction would cause greater drop across R1, making the output base more negative than before with respect to its emitter and increasing output transistor conduction. The resulting additional output-transistor current passing through the lowered value of R2 would tend to restore the voltage drop across R2 to its former value. The system is therefore a regulator.

C1 has little effect when the repeat is off and the system is simply furnishing +15 volts. The time constant of R2-C1 is in the order of 1/3 second, and load changes are unlikely to happen often at this rate. C1 therefore does little to inhibit the normal fast action of the regulator. What little effect it has is to make any sudden changes in the 15 volts unnoticeable, since C1 takes time to charge and discharge and there can be no sudden alterations.

If we now turn the repeat tablet back on again, we can see what happens with very little further discussion. Instead of a steady voltage, pulses are applied to the driver base, their positive peaks the same potential as was the steady potential when the system was turned off, the negative excursions near ground potential. The Pulse 1 and Pulse 2 lines, therefore, carry intermittent on voltages to whatever diode circuits are being energized.

When a pulse comes through causing the output transistor to conduct, the transistor charges C1 quickly because it is a low-impedance source. This causes a fast attack of the note, with the desired percussive effect. However, when the pulse is removed and the output transistor becomes nonconducting, C1 is left to discharge through R2, which takes almost 1/3 second. The potential appearing on the Pulse 1 and Pulse 2 lines, therefore, is not a square wave, with the +15 volts going cleanly on and off. It goes on quickly but subsides more slowly. The tone follows suit. This is more realistic, since a plucked string or even a struck wood resonator does not stop sounding the instant the pluck stops or the mallet is taken away.

One more detail. Why two pulse lines? Obviously, just one would suffice to apply to the entire +15-volt keying system. Actually, the +15-volt bus throughout the keying system is divided; there are two of them, in effect. The diode keyers operated by three keys obtain +15 volts from one bus. Then the next three keys use the second bus. The next group of three uses the first bus again, and so on. If you are holding a group of notes—or if several upper-manual keyers are being energized by AOC or the coupler—it is probable that notes in both groups will be included. Thus you do not hear all the notes go on and off together with the repeat. They alternate and produce a result more like the actual xylophonist who holds a hammer in each hand and strikes notes alternately.

Flute Voicing Scheme

The tone-coloring scheme of Lowrey organs is unique in that a single set of eight cutoff filters, used in various ways, does nearly all the voicing for both manuals and for all pitch registers. Each of the filters is a packaged circuit, adding further to manufacturing economies.

In organs for the home, particularly those intended for a predominantly nonprofessional and untrained group, the most important voices of the organ are the flutes. This is not to say that good flutes are not important to highly skilled players as well—or that more interesting voices such as good reeds and strings are not useful to amateurs—but untrained people tend, at the start, to be a bit afraid of obtrusive tone colors. Bland tones are therefore commercially most desirable; they sell organs, even though if nothing else is available they become monotonous after a while.

The Lowrey voicing system (like the Thomas) concentrates mainly on obtaining flute tones in every register with maximum economy.

The frequency-divider system of tone generation, itself the most economical, poses certain difficulties in obtaining thick, bland flute tones. The waveform is basically complex, and the task is to get rid of the harmonics, leaving not much

more than the fundamentals, without the scaling problems which were discussed in previous chapters and would be aggravated by such strong filtering as the designers of these organs want. The Thomas organ of the last chapter employs a compound L-C cutoff filter to get around the problem. The Lowrey solution is different but also uses a philosophy basically much the same.

We can quickly illustrate the essence of the Lowrey system by looking at the Tibia 8′ voicing for the lower manual. You will recall that the main register of the lower manual, which is at 8′ pitch, is directly keyed by springs and contacts. Up to this point we have left the impression that the 8′ output bus shown in Fig. 12-7 acts as a collector for all the keyed lower-manual tones. In actual fact, the output bus is broken into five sections.

Fig. 12-13 illustrates the basic scheme in block form. The five busses shown are equivalent to the output bus of Fig. 12-7, but each of them picks up tones only from the notes indicated. Each bus feeds the input of a filter which has a

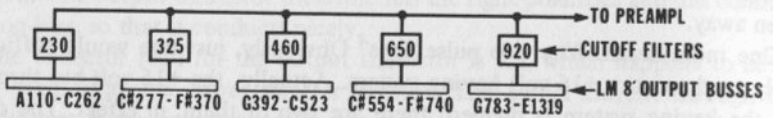


Fig. 12-13—How lower-manual unison output bus is split to deliver range sections to filters.

peak of 3 db or so and attenuation of around 20 db an octave higher. The cps figure in each filter box indicates the approximate frequency where the slope begins. You can see that each group of tones goes into a filter which leaves the fundamental but attenuates the harmonics severely. There is some fudging—the

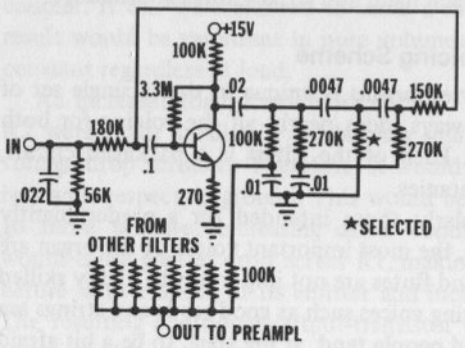


Fig. 12-14—Schematic of a filter shows similarities to a phase-shift oscillator.

first filter passes the second harmonic of the first couple of notes without attenuation, and in all the filters the upper few notes have their fundamentals slightly into the cutoff region. However, the cutoff slope is much more gradual around the turnover point and these little conveniences of design are not noticeable in the tonal results.

The same scheme is used in the upper manual. There, of course, the keyed tones come from the diode keyers, but in this same way the keyer outputs are divided into groups and fed into separate filters to get rid of harmonics. In the

upper manual the effect is more complete because the keyed tones are square waves, rather than the staircased ones of the lower manual. Unlike the staircased tones, the square waves' lowest harmonic is the third, which in every case is well into the attenuation band of its filter.

All the filter outputs are mixed together and fed to the preamplifier and without further ado we have flute tones. There is no attenuation at any point from generators to filters, since no preemphasis is necessary and the filters themselves have some gain, so the signal-to-noise ratio situation, always a problem in more conventional designs, is eliminated by the simple expedient of dealing with high level throughout.

The Filters

Before we go into the switching system which controls the voicing, and the provisions for reed and string voicing, a look at the actual filters shown in Fig. 12-13 is necessary, for they are used in all the voices and they are unusual.

Fig. 12-14 is a schematic diagram of a typical filter, to which all of them are similar except for certain values which determine the turnover frequency. Most of the components are contained in a packaged circuit. The starred resistor is factory-selected to trim the filter so that the cutoff frequency is just what is desired—that is, to compensate for tolerances of remaining components.

A circuit of this kind is actually an electronic filter, utilizing the transistor not merely as an amplifier but as an integral part of the filter. An idea of this basic type was used by Minshall (an organ manufacturer now defunct) back in the early fifties and is explained in the first edition of this book. Briefly, it was simply a phase-shift oscillator with the characteristic three sections of R-C phase shift between plate and grid of a tube. It was adjusted so that loop gain was insufficient for oscillation and it was used as a shunt component, the grid being the connection point to the resistive circuits carrying the keyed tones. As in a phase-shift oscillator, tones going through the plate-grid loop were in phase at just one frequency, at which point they added and produced a peak, and out of phase at one frequency, where a node was obtained. The characteristic was much like a resonant circuit using inductors and it was capable of producing formants in much the same way. Lowrey, however, was the first with this kind of filter in 1948.

This Lowrey filter operates along the same general lines, though it is somewhat modified. For one thing, output is at a point different from the input; and the output is taken through a series resistor and shunt capacitor to aid the low-pass action. For the same purpose the two 270,000-ohm resistors are returned to ground through a .01-μf capacitor. Essentially, however, there is a phase-shift circuit through which collector signal is returned to base, and the filtering action is the result of phase agreement and disagreement at different frequencies.

I briefly experimented with this circuit myself (which I much preferred to the revolting task of analyzing the thing mathematically) and was unsurprised to find that its transmission characteristic much resembles that of a single-section constant-k filter. Below "resonance," transmission is flat. As the nominal turn-

over frequency is approached, response begins to rise, reaching a well defined peak and then sloping off sharply.

The height of the response peak depends entirely on the gain of the transistor as an amplifier, which depends in turn on the transistor and on the value of some convenient component such as the emitter resistor. The transistor I used had, I am sure, a very much higher h_{fe} than those used by Lowrey, and I used a 30-volt supply. With a 470-ohm emitter resistor my peak was 8 db above the 0 db established by the flat section of the transmission characteristic defined by frequencies well below turnover. With a 15,000-ohm emitter resistor, there was no peak at all. Response was almost flat up to the former peak frequency, at which response was now down about 0.5 db. Above this frequency the downward slope increased very quickly. Such a result would not be easy to obtain with simple R-C filtering, and you would take a very large loss in level in the attempt. It could, of course, be done with one or more inductors, but inductors cost money—at least distinctly more than a transistor—and they are bulky as

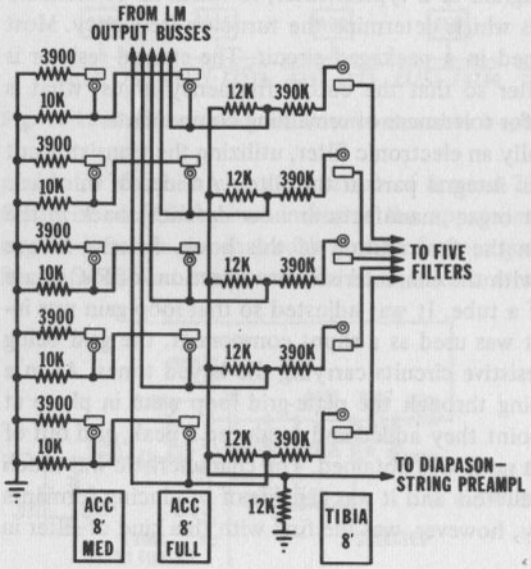


Fig. 12-15—Lower-manual 8-foot Tibia switching, interposed between busses shown in Fig. 12-13 and filters.

well. Lowrey actually does adjust the filters for a slight peak (3 db) even on the lower-frequency circuits, and sets the higher ones for greater peaks so as to give more pronounced formants for reed voices which, as you will see, also utilize these filters. In other organ models separate filters of the same kind are used for the reeds and they can, of course, be adjusted ideally for their single purpose.

Lower Manual Voicing Control

Fig. 12-15 shows how the lower-manual Tibia 8′ stop is controlled. (It is the only flute stop on the lower manual.) As shown in the diagram all tablets are in their off positions.

Each of the five sections of the keying output bus is connected to the center-

point of a pair of resistors. Of these, the 390,000-ohm resistor in each case conducts the tone through a normally open switch to a specific filter, as we described (without the switching) in Fig. 12-13. When the Tibia 8′ tablet is on, all the switches shown above it close, and the filters receive all the keyed tones. From each keying bus line there is also a 12,000-ohm resistor, all of which join and go through another 12,000 ohms to ground. At this single junction there appear all the keyed tones, mixed, for use in forming diapason, string, and reed tones, as we shall see later.

Two tablets at the left in the diagram control the loudness of the lower manual. With both of them off, as shown, each of the keying output busses is terminated through two switches by 10,000 ohms and 3900 ohms in parallel to ground. If you will take a quick look at Fig. 12-4, you will see that this net resistance is the shunt leg of a voltage divider of which the series legs are the staircasing resistors for each generator tone. The resistance between each keying bus and ground therefore determines the signal level. With both switches off as shown (contacts are normally closed), the level may be called soft.

If the Acc 8′ Med tab is turned on, all the contacts above it in the diagram open. This removes the 3900-ohm resistors from the circuit and leaves only the 10,000's. The increased resistance increases tonal level to "medium." Now if the Acc 8′ Full tab goes on, the contacts above it open and remove *both* resistors from shunting the busses, bringing the level up to "full," at which point the significant load on each bus is just its 12,000-ohm resistor going to ground through the bottom 12,000-ohm resistor.

A glance at the diagram will show that while the tones from each section of the lower-manual pickup bus go into a separate filter, *all* of the tones appear across the 12,000-ohm resistor which goes to ground, since it is a common load. This is the point from which tone is taken for nonflute voices, for each of which all the tones are treated together, not in groups as for the flutes.

Fig. 12-16 shows the switching for the nonflute voices. The 12,000-ohm resistor is the same one as appears in Fig. 12-15. All keyed lower-manual tones are taken from it and fed to a voltage amplifier, which has the grounded-base configuration so that the phase of the tones when they finally reach the output of the organ will not interfere with and cancel those of the flutes. Output of this amplifier is fed to the switching system.

The diagram shows all springs in their off positions—stops not sounding. The switching system is in two parts. The first part involves the topmost stationary contacts and a spring for each stop. Note that the top of the 12,000-ohm resistor is grounded through all the springs and contacts in series. This ground, which is removed whenever a stop tablet is switched on and the series line to ground is broken, serves to keep things quiet when nonflute voices are off and there would be no tones to mask any noise or leakage. The main voice switching concerns the lower two sets of contacts and the lower sets of springs, each of which strikes the center contact when the stop is off (as shown) and comes down to strike the lower contact when the stop is on.

When the Diapason is switched on, the top spring comes down, leaving the

contact above it and breaking the ground shunt away from the 12,000-ohm source resistor. Tone then goes through the amplifier and into the lowest contact.

The lower Diapason spring is now touching the lowest contact, carrying amplifier output (which, you will recall, is staircased square wave from the generator through the key switches) to both the 920-cps and 650-cps filters, through isolating resistors. The result at the filter outputs is a spectrum characteristic in which response is uniform to 650 cps, comes down to a lower value at that point, then continues out to 1300 cps and slopes off.

The same kind of switching action operates with the Cornet, which is also fed to two of the filters. It is fed to the lowest one, however, through a small capacitor, so that response is attenuated in the low-frequency region. The result is something of a peak in the range which, while very broad, gives something of a formant.

In the voice-switching section of the Violin tablet, amplifier output is fed to the spring through a capacitor which slightly attenuates the lower harmonics. The contact then feeds tone to a string amplifier which is in the "quality

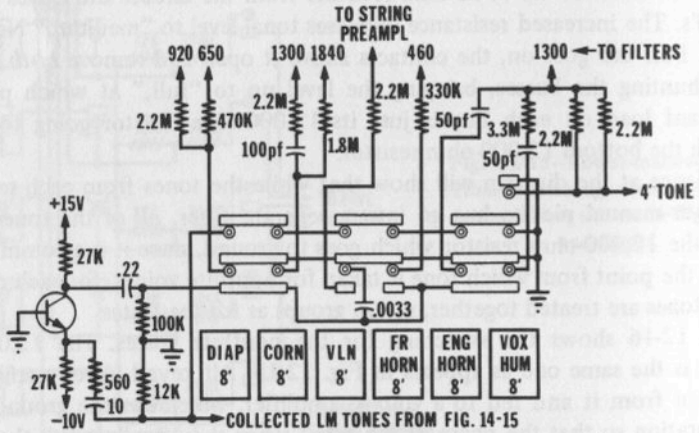


Fig. 12-16—The diapason-string preamplifier for the lower manual is a grounded-base circuit with emitter input from the lower right of Fig. 12-15. The 12,000-ohm resistor here is the same one that appears in the previous figure.

control" section of the organ, as that part is called which contains the filters, preamplifier, and certain other components. This is a simple amplifier which brings tone to the output preamplifier without further filtering.

The French Horn spring brings tone to the 460-cps filter, and the English Horn spring brings it, through a capacitor, to attenuate lower harmonics and produce the necessary whine, to the 1300-cps filter.

The Vox Humana is a tone which in pipe organs has at least two formants and can never be formed in as simple a manner electronically as other voices. Here the regular 8′ tone is fed through a capacitor and resistor to the 1300-cps filter. In addition, 4′ tone is taken from a collector (which is akin to the 12,000-ohm

resistor for the 8′ register) and fed to the same filter through a net resistance of about a megohm, thus adding an extra component to the total tone. An additional contact serves to ground this 4′ tone when it is not in use, just as the 8′ complex tone is grounded by the top series of contacts. (The 4′ collected complex tone is not used for any other purpose.)

The 4′ tone is, of course, collected from the busses of the row of keyboard springs associated with the 4′ coupler, as in Fig. 12-7. There are four such short busses (as 4′ tone exists only for the lower 32 keys of the manual, due to generated tone shortages) and each one goes through resistors and one of four contacts of the Lower To Lower 4′ coupler tablet to join the input wires from the 8′ output busses shown in Fig. 11-15. Thus when this coupler is on both 8′ and 4′ keyed tones go through everything we have discussed.

The voicing system in this organ yields excellent flutes for obvious reasons. But for nonflute voices with strong differences and authenticity this common-filter system cannot give ideal results. I count its chief virtue to be economy, as it makes practical the use of packaged circuits for most of the voicing. Individual R-L-C voicing filters cost more, both in materials and labor, so that an organ built in this way can give the customer more "features" for a given amount of money. But do not make the mistake of thinking, if you are new to the organ field, that the method gives the best voicing.

Upper Manual Voicing

On the whole, the upper-manual voicing is done in about the same way as on the lower manual, though with more elaboration in the nonflute voices and more multiplication in the flute voices because there are tibias at all five pitch registers. The space to show these similarities would not be justified. Component values are different and, of course, the tones are taken, not from a keying bus, but from collected outputs of diode keying circuits.

A few special points merit mention, however. It may occur to you to question the waveforms used for nonflute voices. You will recall that square waves right from the generator flipflops are fed to the diode keyers. If square waves were used for nonflute filters, all of them would have the hollow sound of tones lacking even harmonics.

The solution to this problem illustrates again the Lowrey design ingenuity—as well as the Lowrey trend toward economy and tonal compromise to achieve it.

Each of the tibias for the upper manual has a resistor and switching arrangement like that of the sole tibia in the lower manual (Fig. 12-15), though with different resistor values and provision for only two rather than three volume levels for each. And in the same way, each tibia system produces a common output containing all keyed tones of that pitch register before filtering, much like the common tone appearing across the 12,000-ohm grounded resistor in Figs. 12-15 and 12-16.

These commoned tones are brought together from all the tibia arrangements except the 16′ (and 2⅔′). This means that the final composite contains, when any key is pressed, square waves of all the octaves of 8′ and up available in the

organ. This mixing of octaves on the reed-string collector line is tantamount to staircasing at the generators, and the resultant tone contains both odd and even harmonics.

There are two of these reed-string lines, one as just described and the other from the 16′ tibia switching alone. Each of them goes through an amplifier like that in Fig. 12-16. The output of the 8′ amplifier, containing the mixed tones of all but 16′ pitch, is used to form nonflute voices much as in Fig. 12-16. The trick is that all these voices are of 8′ pitch, none higher! So obviously there is nothing wrong with using up the combined outputs of the 8′-and-up pitch registers in this way. (There are three, to be exact—Diapason, Violin, and Trumpet.)

The same collected tone could not, of course, be used directly for the three 16′ stops—Cello, Trombone, and Clarinet. Instead, the tone obtained from the 16′ tibia collector and amplified is used for 16′ stops; and to avoid missing the even harmonics tone is also mixed into the 16′ stops (except for the Clarinet) from the amplified 8′ line, right in the switching process through isolation resistors which prevent 16′ tone from getting into 8′ stops.

One more provision is unique in organology—believe it or not, a "frequency doubler!" When I first saw this term in the diagrams I was astounded since, while frequency doublers are certainly known to the electronic art, they, like frequency dividers, are invariably nonlinear and cannot be used for more than one tone at a time. The answer is very interesting.

The compass of the generated tones is just enough to give an 8′ tone for the top key of the upper manual. Were it not for the doubler, it would be laughable to pretend to have pitch registers as high as 2′, since true 2′ tones would be available only for the lowest octave and a half of the manual. Use of a doubler effectively adds an octave to the range of the organ, so that at least two and a half octaves of the upper manual actually have 2′ tone and all three and a half have 4′ tone.

The doubler system consists of two transformers which are tuned so that transmission from one winding to the other is good over the span of about one-half octave per transformer. One transformer is centered around 2600 cps and the other at 3800. Tones from appropriate sections of tibia switching arrangements, instead of going to filters like the rest, go to the doublers.

The tones fed to the doublers are in the top octave provided by the generators—that is, tones from the master oscillators (through the buffers). While the tones from the frequency dividers are square and thus have no harmonics below the third, the oscillator output waveform contains over 50% second harmonic. Since the doublers are tuned to these second harmonics, the doubler outputs contain *only* those harmonics and consist entirely, for all intents and purposes, of tones an octave higher than the master oscillators. The tones are, of course, sine waves (very nearly) and need no further filtering for tibia use.

Tone Output

The main preamplifier is a one-transistor amplifier of the garden variety whose input is fed by (a) all the filter outputs, (b) outputs of the doublers, and

(c) the string amplifier. There is a swell-shoe control consisting of a light-dependent resistor (LDR) in series with the signal and a lamp whose distance is varied from the LDR by the shoe. There is also a brilliance control and a loud-soft arrangement controlled by two tablets.

Pedal Switching

Pedal tone production in home-size organs is classically switching generator tones to a frequency divider and thence to filters or volume controls. As I said before, there is nothing simple about this organ, and the pedal system continues to bear me out.

What makes it complicated is the completely electronic sustain system. My comments in the Thomas chapter (page 215) outline the requirements for pedal sustain, which I will not repeat here.

In Fig. 12-17 we have shown the switching system with just three keys, since all 13 have the same repeated circuitry within the dashed lines. Parts outside the dashed lines suffice for the entire system.

Fig. 12-4 indicates that pedal tones are fed to the pedal system directly from flipflop transistor collectors, so that each contains positive direct potential always less than 15 volts. This is applied to the pedal inputs shown at the top of Fig. 12-17.

Let us confine the discussion first to the C pedal. The positive voltage from the C generator is connected to the cathode of diode D1, but there is no effect because the cathode is the negative terminal and because the anode of D1 does not go any place where d.c. conduction could take place, as long as the C pedal

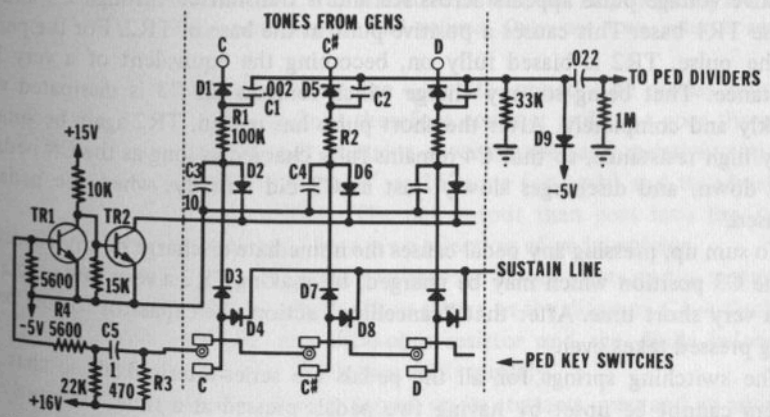


Fig. 12-17—Three typical pedal switches. Items to right and left of the dashed lines are common to the entire system; circuits within lines are repeated for each of the 13 pedals.

is unpressed and the spring is up in the position shown in the diagram.

When the C pedal is pressed and held, +16 volts is applied through R3 to the anodes of D3 and D4. Through D4 it goes to the sustain gate which controls output from the pedal divider; ignore this for now and we will speak of it later.

As the 16 volts exceeds the potential from the generator, current passes through both D3 and D1 so that they conduct, and generator tone passes through D1 and C1 to the pedal dividers.

At the same time the +16 volts quickly charges C3 so that its upper plate is positive. It also places a positive potential on the collector of TR2, so that TR2 is in operating condition. The normal d.c. at the base of TR2, however, is such that TR2 is cut off. Ignoring all the rest, the main purpose of it all—getting C-generator tone through to the pedal dividers—has been accomplished.

Now suppose the C pedal is released. The positive charge at the top of C3 then leaks off through R1 and D1. The decay of this charge is slow because TR2, not being fully conductive, represents a rather high resistance; the time required for the charge on C3 to decay, and thus for gate diode D1 to shut off and stop C tone from going to the dividers, is longer than the sustain decay occurring through circuitry we shall discuss later in the outputs of the pedal dividers. In other words, C tone continues to go to the pedal dividers longer than necessary so the divider will not conk out during the sustain decay.

Suppose next that the C# key is pressed before all charge has leaked off C3, while the C note is still being fed through D1 to the dividers. Two things must happen: (a) D5 must begin to conduct so as to send C# tone to the dividers; and (b) D1 must instantly be shut off to stop the C tone, since the dividers cannot handle more than one tone at a time.

When the C# pedal is pressed, the +16 volts is applied through R3 (and D7) to C4. C4 charges very quickly because R3 is such a low resistance. In so doing, it causes a pulse of current to pass downward through R3, as a result of which a negative voltage pulse appears across R3 and is transmitted through C5 and R4 to the TR1 base. This causes a positive pulse at the base of TR2. For the period of the pulse, TR2 is biased fully on, becoming the equivalent of a very low resistance. That being so, any charge which remains on C3 is dissipated very quickly and completely. After the short pulse has passed, TR2 again becomes a fairly high resistance, so that C4 remains fully charged as long as the C# pedal is held down, and discharges slowly, just as C3 did initially, when the pedal is released.

To sum up, pressing any pedal causes the immediate discharge of any capacitor in the C3 position which may be charged, by making TR2 a very low resistance for a very short time. After that "cancelling" action, the capacitor for the pedal being pressed takes over.

The switching springs for all the pedals are series-interlocked, so that the system cannot be upset by having two pedals pressed at a time. Obviously, in such a case the lowest one pressed would take over.

Diodes D3, D7, etc., are present to prevent positive d.c. remaining on C3, C4, etc., from feeding through D4, D8, etc., after a pedal is released. As you will see, positive voltage is desired through D4, D8, etc., only while a pedal is being held down. Similarly, D4, D8, etc., are provided to prevent positive d.c. charge on the sustain capacitor to which they are connected from dissipating through the other keying components.

Pedal Sustain

The lowest frequencies fed to the pedal system from the generators are two octaves higher than those needed for the 16′ pedal tones. Two dividers are necessary, therefore, to bring them down to the correct pitches. Each is more or less normal and they will not be diagrammed.

The output of one of them is shown in Fig. 12-18. Since the voltage on the collector is positive, the diode directly connected to it is normally shut off, and since the anode of the one leading to the pedal tone output line is connected through 39,000 and 82,000 ohms to a source of −5 volts, it, too, is shut off. The output of the other flipflop is connected in just the same way. One flipflop, of course, furnishes 8′ tones and the other 16′.

In Fig. 12-17, when a pedal switch is pressed +16 volts goes through R3 and D4 or a diode in a similar position to the sustain line and immediately charges C1 in Fig. 12-18 positive. The positive voltage is greater than that on the flipflop

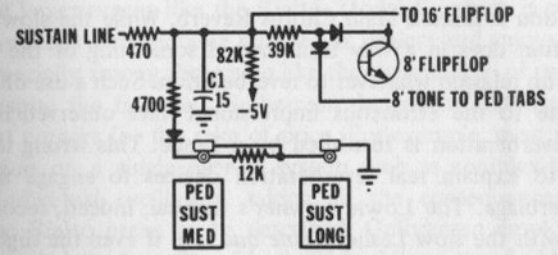


Fig. 12-18—The sustain line from Fig. 12-17 operates a gate for 8-foot pedal tone (shown) and a similar one for 16-foot tone. The remainder of the pedal stop switching is a routine arrangement for selecting pitch register and volume. Only one tone color is available.

collector and exceeds and swamps the −5 volts applied. The diodes connected to the collectors of both flipflops therefore conduct and tone goes through to the output. The output circuit contains a waveshaper and resistive stop-tablet switch to allow selection of either 8′ or 16′ tone (or both) and to adjust total pedal loudness with stop tablets. The net output then goes into the 140-cps filter used for manual voicing. There is no selection of pedal voicing.

When a pedal is released, positive voltage no longer exists on the sustain line. With the two pedal sustain tablets off, as shown in the diagram, C1 is discharged relatively quickly through the 4700-ohm resistor and the diode below it to ground, and the signal diodes block tone fairly rapidly.

If the medium sustain tablet is turned on its contacts open and an additional 12,000 ohms is interposed in the discharge path, so the decay takes longer. And if the long sustain tablet goes on and its switch opens, the path to ground is removed entirely. All the discharge goes through the 82,000-ohm resistor (with the help of the −5-volt source) and discharge and tone decay take longest.

If another pedal is pressed (or, of course, the same one is pressed again) a new positive charge comes in on the sustain line and gates open completely again. When it is released the decay begins anew.

Final Section

The amplifier and power supply for this organ are on a common chassis, and both of them, plus the speaker switching provisions, are fairly normal for Lowrey organs. A multitap power transformer is used with two rectifier-filter systems to furnish unregulated voltages, positive and negative, of the various values required in the organ. There is nothing especially out of the way about the power amplifier, which utilizes four transistors and a driver transformer and furnishes a single transformerless output through a 1,000-μf capacitor.

The speaker complement includes two 12-inch woofers and a 6 X 9-inch midrange unit plus L-C crossover network for the main speaker system, plus a built-in single-speaker Leslie unit. There is a headphone jack, a pleasant convenience; it disconnects the speakers when in use.

The output of the entire organ may be fed either to the Leslie or to the main speakers or both, and there is a connection for an optional external Leslie cabinet. The Leslie has, as usual, two speeds. The tablet which calls the slow Leslie effect into action is labeled Main Chorus Reverb. While the slow speed on any Leslie installation does in a way lean toward something of the function of a chorus, it has no relation whatever to reverberation. Such a use of the words can only contribute to the erroneous impression I have observed in many organ buyers that reverberation is furnished by a Leslie. This wrong idea has forced those trying to explain real reverberation devices to engage in much extra explanatory verbiage. The Lowrey owner's manual, indeed, recommends using some sustain with the slow Leslie, a *sine qua non* if even the slightest suspicion of anything distantly related to reverberation is to be had.*

*Stop-press news flash: Lowrey organs now almost all have Hammond spring-type reverberators in them, and the tablet discussed now reads Main Chorus, minus the Reverb. Hooray!

Chapter 13

Gulbransen Organs

The Gulbransen Company, 2050 North Ruby Street, Melrose Park, Ill., is an old-line piano manufacturer which turned some years ago to the manufacture of electronic organs in addition to its pianos. The company was acquired in 1966 by Seeburg, best known as a maker of juke boxes, which had already bought out Kinsman organs.

Gulbransen was among the earliest organ manufacturers to go all out for transistors. Its present line includes both spinet types, such as the Model H of Fig. 13-1, and larger organs like the Rialto, Model K, which is pictured in Fig. 13-2. Gulbransen has a respectable number of dealers and enjoys a good deal of popularity, especially among those who like the theatre type of organ in which it specializes, despite the fact that Gulbransen designs are not cheap; there is no attempt to cut corners for the sake of price. For example, most organ designers make a decision on a single over-all system such as complex-tone frequency dividers or individual oscillators, taking all the consequences the decision implies. In the Rialto organ to be described, Gulbransen employs two sets of generators, to enjoy the benefits of both. This philosophy of adding whatever is necessary to accomplish an additional result seems to pervade Gulbransen design, in contrast to the more usual process of trying the designer's ingenuity to the limit to squeeze additional effects from a single integrated system.

Main Tone Generators

There are two separate and complete tone generation systems in the Rialto organ. One is composed of seven octaves of individual sine-wave oscillators, known as the *tibia generator.* The other is the *main generator,* which produces sawtooth waveforms for tone coloring by formant filters. They are handled entirely separately up to a rather late point in the organ. We can therefore discuss them quite separately.

The main generator is actually 12 individual generators, as usual, each consisting of a master oscillator and five frequency dividers. The master oscillator is diagrammed in Fig. 13-3. It is a fairly standard grounded-collector-type Hartley, with the collector not quite "grounded." It feeds a clipper stage which furnishes complex-wave output for the top octave and synchronization signal for the first frequency divider. The top octave produced by the main generator is Octave 6, Table 1-1.

The vibrato circuitry is interesting. C3 is the main tuning capacitor across the oscillator coil, with about 85% of the total capacitance. However, there are two auxiliary tuning capacitors, C1 and C2, each across the coil through a diode.

With the diode biased on, either capacitor is effective; with its diode not biased on, a capacitor is effectively not in the circuit. C1, C2, and C3 are factory-selected. To illustrate how the vibrato works, let us assume C1 and C2 are equal.

When vibrato is off, a negative voltage is applied to terminal V2, so that D2 conducts and C2 is effectively across the coil. At the same time, V1 is grounded, so that D1 does not conduct appreciably and C1 is not in the circuit. Thus, with vibrato off, one-half of the auxiliary capacitance is effective, and in this condition the organ is initially tuned by rotating the top of the cup core in which the coil is built.

When vibrato is turned on, terminals V1 and V2 are removed from ground and fixed negative. They are shorted together and a pulsating negative voltage is applied to both. At the instant when the pulsating voltage is at maximum, both

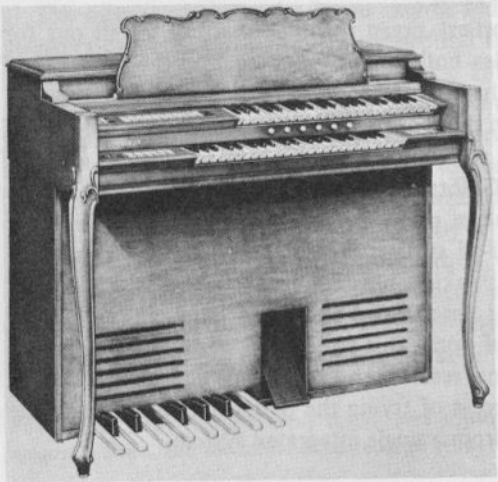


Fig. 13-1–One of Gulbransen's spinets, Model H. Several manufacturers offer "French Provincial" consoles of this kind.

D1 and D2 conduct and the full auxiliary capacitance is effective. When the voltage is zero, both diodes are off and neither capacitor works. The auxiliary capacitance varies during vibrato, therefore, from zero to 100%. With no vibrato it is fixed at 50%. As a result, the no-vibrato frequency is the same as the average frequency during vibrato so that the ear hears no apparent change in note pitch when vibrato is turned on and off.

The frequency dividers in this organ are not the usual flipflops, but blocking oscillators instead. Blocking oscillators were discussed beginning on page 54, so there is no point in dissecting Gulbransen's. One of them is diagrammed, however, in Fig. 13-4, so you can see what it looks like. The circuit is almost exactly like that shown in Fig. 3-14, with a transistor substituted for the triode tube. Synchronization signal for the first divider emanates from the upper right of Fig. 13-3, and output, marked 5 on the assumption that the divider shown is the one immediately following the master oscillator, is taken from the top of the R-C timing network rather than through the capacitive divider of Fig. 3-14. Signal to synchronize the next divider comes from a potentiometer across the collector winding of the transformer.

While a transistor blocking oscillator is likely to be inherently more stable than a tube one, simply because transistor characteristics do not change much, the circuit is definitely periodic and is sensitive to tolerances in all components and voltages. Its only advantage is that it provides sawtooth tone directly. Designers today seem unanimous in feeling that flipflops are preferable (and I concur) because of their stability and the fact that they need not be designed for a particular frequency of operation; staircasing is easy enough to eliminate the



Fig. 13-2–The Model K Rialto described in this chapter is a theatre-type instrument.

advantage of the direct sawtooth waves. The blocking oscillators exist in this organ, I am sure, simply because these generators were designed quite some time ago, even though the organ is still a current model as of 1967. What I am trying to say is that there is no point in using blocking oscillators in future designs, so let's get on to the next subject.

Main Generator Keying

All keying of the main generators is electronic, with keyers for each note like

the one in Fig. 13-5. Main generator output for one note is connected at the point marked GEN. The grounded bus shown runs under all key switches of one manual, and there is a switch finger as shown for each note.

This is a shunt-type diode keyer. The tone passes through R1, R2, R3, and either R4 or R5 to a preamplifier. At each junction there is a diode to ground,

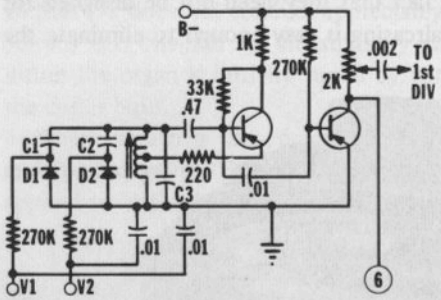


Fig. 13-3–Master oscillator and shaper.

with a large resistor from each diode to a common point. Negative voltage is applied to this common point and thus to all diodes permanently; since the cathodes are away from ground, the negative voltage causes the diodes to conduct, shorting the signal to ground. Three of these shorts are required, since a conducting diode is not a perfect short and with only one section there would be some tone leakage. Simpler organs do, indeed, use simple shunt or series diode keyers, but they invariably have a higher leakage level, which is usually ameliorated by chopping off highs, with acceptance of the loss of voicing brilliance which that brings.

When the key is pressed and the switching finger comes down and touches the ground bus, the 6-μf capacitor, whose charge is the voltage which keeps the diodes conducting and the tone turned off, is discharged to ground through R7,

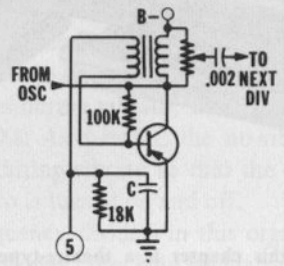


Fig. 13-4–Gulbransen dividers are blocking oscillators.

which is just enough resistance to round off the tone attack and keep it from being instant. With the diodes nonconducting, tone goes to the two outputs. When the key is released, −34 volts is again applied to charge up the 6-μf capacitor and energize the diodes. However, the capacitor must charge through a total of 48,000 ohms, so the decay of tone is also somewhat gradual.

All the manual keyers are similar to the one in Fig. 13-5, though values starred differ somewhat to provide what the designer believes to be the appropriate attack and the correct relative volume of the note. There are 60 such keyers for each manual, handling generator octaves 2 through 6. (Octave 1 is used only

for the pedals.) The lower-manual keyers have only one output, so R4 is missing.

Fig. 13-6 gives some idea of how the key switching works. The drawing is an end view of the switches behind the keys of the upper manual. There are 12 contact finger wires lined up as shown, their lower ends supported by a fixed

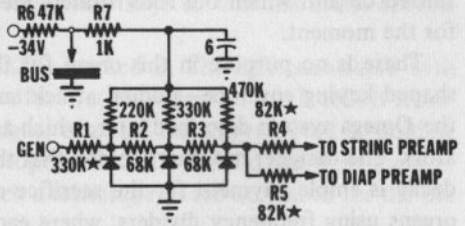


Fig. 13-5–Diode keyer for one note.

insulator. The wires run through a movable actuator. At the positions shown near the top are 12 bus rods which run the length of the manual. When the actuator is pushed toward the rear of the organ (to the right in the drawing), all the contact finger tops move in the same direction, and each comes in contact with the bus at its right in the drawing.

These busses have a peculiar character. Each is an insulated rod (the semicircular portion) from which a metal bus extends. In the diagram, all the busses but the one at the far right (main 4′) are facing up, so that when the switches are actuated each finger but the last will come in contact with the insulated part of its bus and nothing electrical can happen. The 4′ main bus, however, has been rotated 90° counterclockwise by the electromagnet energized by the Solo 4′ coupler, so that the main 4′ finger will contact the metal bus when the key is pressed and a linkage associated with it pushes the actuator toward the rear of the organ.

The 4′ main switch finger under each key is connected to a keyer (like that of Fig. 13-5) handling tone of the appropriate pitch. With the Solo To Solo 4′ tablet on, therefore, pressing any key causes a 4′ tone to go through its keyer to the following voicing circuits.

If the 8′ tablet were on and the 8′ main bus therefore rotated to the same position shown for the 4′ bus, the 8′ spring under any key would pick up keying voltage and pass it to the keyer of the correct 8′ pitch. And of course, the same is true for the 16′ bus and springs.

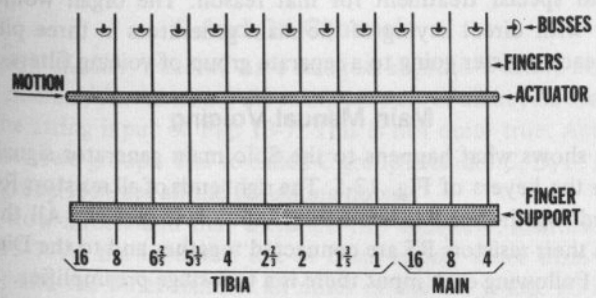


Fig. 13-6–The upper-manual key-switch assembly has 12 contact fingers for each key.

These three fingers and busses therefore control the pitch registers of the tones sent to the common voicing circuits. Since the voicing filters collect all tones keyed, obviously each voice cannot have its own pitch register; it sounds in one, two, or three pitch registers which depend on which couplers have been turned on and which bus rods rotated. Please ignore the nine tibia switch fingers for the moment.

There is no purpose in this organ for the diode keyers other than to obtain a shaped keying envelope—gradual attack and decay—for no sustain exists except the Omega system described later, which applies only to the other rank of generators. The designer has thus decided that the somewhat more pipelike attack and decay is ample payment for the sacrifice of tonal versatility attainable in other organs using frequency dividers, where each stop can sound at its own pitch and ensembles of stops can therefore be made up with greater variety of effects. In

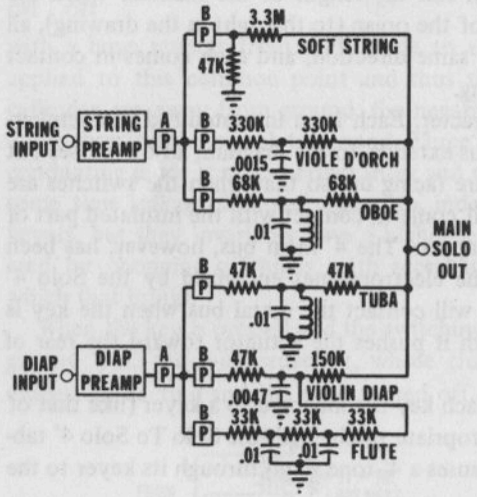


Fig. 13-7—Solo manual voicing for nontibia stops.

a theatre-type instrument where snappy response is often valuable and where the kind of insistence on fine points like the keying envelope sometimes heard from serious organists is rarely encountered, I count this a waste, especially since the sawtooth waveform being keyed is not a source of unpleasant keying transients and needs no special treatment for that reason. The organ would have been much better with direct keying of the main generators in three pitch registers, the tones of each register going to a separate group of voicing filters.

Main Manual Voicing

Fig. 13-7 shows what happens to the Solo main generator signals after they emerge from the keyers of Fig. 13-5. The right ends of all resistors R4 (Fig. 13-5) are connected together and to the String Input of Fig. 13-7. All the keyer outputs through their resistors R5 are connected together and to the Diapason input of Fig. 13-7. Following each input there is a two-stage preamplifier.

The output of each preamplifier goes to three formant-type voicing filters.

There is a light-sensitive resistor, indicated by P in a box following each preamplifier. Both of those which are marked A are illuminated only when any stop is on. When no stops are switched on, LDR's A have very high resistance, the purpose being to kill any leakage that may come through the keyers unless a stop is on and, presumably, the player is pressing some keys so that the music drowns out the leakage.

Each filter is also preceded by an LDR marked B. The light associated with each of these is turned on by the tablet for that stop; each of the B cells is therefore just a stop switch. Unless the designer assumed the player would be holding down keys at the moment of turning stops on and off, and did not want them to come on abruptly (no organist uses such a technique), I see no reason for the use of LDR stop switching.

The reason for the separate preamplifier preceding each group of three stops is a difference in their frequency responses; the one labeled Diapason has more highs knocked off. Possibly the ability to preset the levels for the two groups

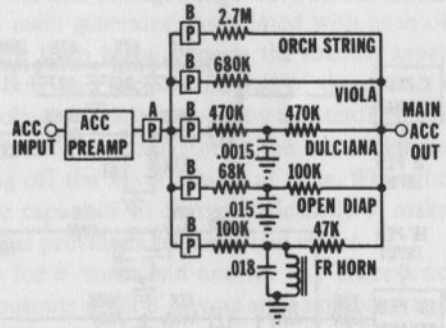


Fig. 13-8—Complex-tone stops for the Accompaniment manual.

also was a factor. From where I sit, however, providing two preamplifiers seems (like other things in this organ) elaboration without adequate benefit.

At any rate, all the filter outputs are collected together and form the total Solo manual signal due to the main generator—all the nonflute stops. Accompaniment tones are handled in the same way, except that each Accompaniment keyer has only one output (no R4, Fig. 13-5). They are all gathered and fed to the Accompaniment preamplifier shown in Fig. 13-8, then through the same LDR-type switching system to the five Accompaniment stop filters.

C And C# Channels

We must now modify what we have said just slightly. We have explained that for the Solo manual, for instance, the string outputs of all keyers are commoned and fed to the string input of Fig. 13-7. This is not quite true. Actually, tones are divided into two groups. The so called C group includes C, D, E, F#, G#, and A#. The C# group includes all the remaining notes.

You may now understand that there are two complete, identical setups like that in Fig. 13-7. At the inputs of one of them appear tones from all keyers in the C group; and the other handles all notes of the C# group. At the terminal labeled Main Solo Output, therefore, appear keyed and voiced tones from the C

group. At another exactly similar terminal appear tones from the remaining six notes. The same is true of the Accompaniment signals of Fig. 13-8. At this point, therefore, we have four signal outputs from the manuals: (1) Solo C group, (2) Solo C# group, (3) Accompaniment C group, and (4) Accompaniment C# group. You will find that this separation into C and C# groups is carried on through the rest of the organ, right up to the loudspeakers. The stated reason is the desire to avoid electronic mixing of pitches separated by a musical fifth or fourth. Such tones have harmonics quite close together (so much so that they are used for tuning) and any intermodulation would cause very unpleasant product tones. I would personally have thought that the solution found by other quality manufacturers would be adequate—seeing that none of the audio circuits operate nonlinearly. However, at Gulbransen elaboration is the name of the game!

Line Amplifiers

The C-channel output of the Solo manual and the C output of the Accompaniment are now brought together and fed through a so called line amplifier. The

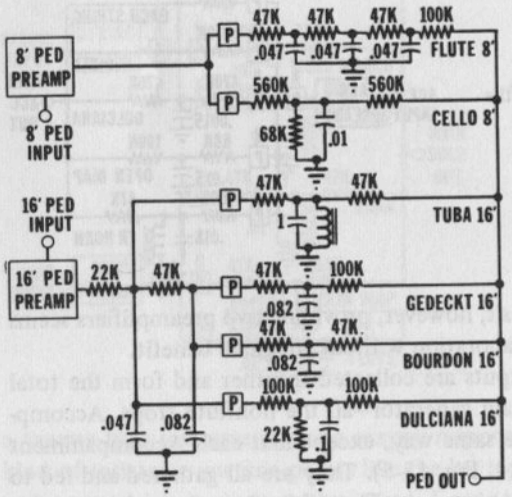


Fig. 13-9—Pedal voicing system.

C# outputs from the two manuals are also brought together and fed to a C# line amplifier. The two amplifiers are identical.

Between the first two stages of each amplifier is an LDR controlled by the swell shoe, for expression control. The final stage of each amplifier actually includes two output transistors, not in push-pull, but with their inputs in parallel to provide two identical outputs. Between the input of each of these output transistors and ground is a simple network including an LDR. These LDR's are lighted alternately by a rotating shutter at a vibrato rate when the tremolo (not vibrato) is turned on, imparting to each output some amplitude modulation. The LDR's are illuminated alternately, so that when one output is at maximum amplitude the other is at minimum. The two outputs are provided in case the owner has two identical tone cabinets. Each cabinet is then fed one of the C outputs and one of the C# outputs. With both cabinets working and receiving identical signals, this tremulant system amplitude modulates them out of phase, creating an interesting result. The lamps are controlled by the same rotating shutter which controls the vibrato signal applied to the master oscillators by alternately lighting and darkening an LDR in series with a fixed d.c.

To sum up, we now have two identical (if we forget about the little tremulant idea) outputs for all manual tones of six alternate generators beginning with C, and two more identical ones for all the other six generators. If we assume just one tone cabinet, we have simply two outputs—manual C and manual C#. These two tone groups are connected to separate amplifiers and speaker systems in the rather gigantic tone cabinet made by Leslie especially for use with this organ. Gulbransen makes no amplifier-speaker equipment for the Rialto. The speakers used for these two channels are stationary, though rotating ones in the customary Leslie fashion also exist in the cabinet for another purpose.

The Pedal System

Pedal tones are all taken from the main generator. Associated with each of the 25 pedal keys are two diode keyers. Each is the same as the manual keyer shown in Fig. 13-5 except that it has only two sections instead of three, and a single output. The negative blocking voltage is made variable by a potentiometer in the power supply. When it is maximum, the capacitor in the keyer charges quickly when a key is released, turning off the keyer almost at once. When the voltage is lower, it takes longer for the capacitor to charge sufficiently to make the diodes conduct and kill the tone. This provides a pedal sustain effect.

There are two ranks of keyers, one for 8′ tones and one for 16′. There is no division into C and C# channels. The outputs of all 8′ keyers are commoned and fed to the 8′ pedal input of Fig. 13-9; those of the 16′ keyers go to the 16′ input. In each case, the signals go through a preamplifier, thence to stop filters, each preceded by an LDR controlled by the stop tablet. Notice that these stops are the only ones for the main generators which have their own pitch registers and do not require use of couplers to determine pitch register. All filter outputs come together to form a pedal output. This is fed through a line amplifier containing a swell-shoe-operated LDR and a level preset potentiometer, and then to the pedal power amplifier and speaker system in the tone cabinet.

Tibia System

It might almost be said that the tibia system of the Rialto constitutes a separate organ housed in the same console and operated by the same keys as that we have described so far. Its heart is a completely separate set of tone generators consisting of 84 (seven octaves of) individual sine-wave oscillators, the outputs of which are used without tonal change to furnish the bland tibias which are the smooth, heavy flutes of the theatre organ.

Fig. 13-10 is a diagram of one of these oscillators. Capacitor C is selected for each note for correct pitch, and the coil core may be varied for exact tuning. The seven octaves of this generator are Nos. 2 through 8 in Table 1-1, covering the

lowest 8′ manual notes to the highest 2′ manual octave. The only repetition necessary is for the 1′ register, the highest octave. The lowest 16′ octave is not present; with only a 16′ stop on, the lowest octave of keys simply does not sound. (For some reason, the same is true of the main generators, even though Octave 1 is available from them. It is just not used for the manuals. Perhaps the designer believed these low 16′ tones have no use and would only thicken the sound unnecessarily. This view has some merit, but then why not at least have the lowest 16′ octave repeat the lowest 8′ instead of being completely blank?)

All the 84 oscillator outputs are connected through 100,000-ohm resistors to the common output line shown; and we will see in a moment where that leads. For now, please disregard the Omega connection as well.

The organ does not contain any different kinds of tibia stops, but it can provide tibia tones in seven pitch registers on the upper manual and six on the lower. Looking once again at Fig. 13-6, notice that among the switch fingers operated by each key are nine for the tibias. The 6⅔′ can be disregarded, as it is only used for chimes.

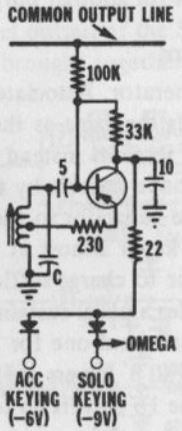


Fig. 13-10—One tibia oscillator.

In the key-switch assembly, the Solo Keying terminal of Fig. 13-10 is connected in parallel to all switch fingers which are to operate a particular oscillator. For example, the oscillator furnishing 440 cps is connected to the 8′ finger under the middle A key, the 4′ finger under the A key an octave lower, the 16′ finger under the key an octave higher, and so on. If the Solo Tibia 8′ stop is turned on, the electromagnet rotates the Tibia 8′ bus in Fig. 13-6 90° counterclockwise, so that the exposed metallic surface will come in contact with the 8′ switch finger when the key is pressed. This switch finger, if the key is middle A, is connected to the Solo Keying terminal (Fig. 13-10) of oscillator 440; the bus is connected to a source of −9 volts. When the key is pressed, therefore, −9 volts is applied through the 22-ohm resistor to the oscillator collector, turning on the oscillator and sending its tone to the common output line.

At the same time, the −9 volts charges the 10-μf capacitor. When the key is released, therefore, the oscillator does not stop instantly but does so over the

space of a little time required for the capacitor to discharge through the transistor circuit. There is just enough time for a pleasant decay without transients, but it is not a sustain.

The same arrangement holds true of the Accompaniment manual, which supplies keying voltage through the other diode. The Accompaniment keying voltage, however, is only −6 volts, so that playing the Accompaniment manual provides tibias at lower volume than the Solo. Naturally, if a particular oscillator is being keyed at the same time by both manuals, it will put out maximum volume. But the two diodes, under normal circumstances, are "one-way valves" to assure that if both manuals are keying an oscillator at the same time, the higher Solo voltage will not get back into the Accompaniment circuits. A tablet labeled Tibia Minor FF (the Accompaniment tibias are called Minor and Solo ones Major) can, at the organist's will, apply the full 9 volts to the Accompaniment keying if he wants them as loud as the Solo ones.

There is no provision for vibrato in the tibia oscillators, since all tibias are fed to the rotating Leslie speaker system, their only source of vibrato.

The common output line carrying all tibia tones is connected to the Tibias In point in the network of Fig. 13-11. In series near the input is a light-sensitive resistor labeled VOL. The lamp for this LDR is controlled by two tablets, Tibias MF and Tibias FF. When the FF (*fortissimo* in musical parlance, meaning *very loud*) tab is down, maximum voltage is supplied to the lamp and the VOL LDR has minimum resistance, passing maximum signal. With the MF (*mezzo forte* or *middling loud*) tablet, the lamp illumination is less and the LDR resistance greater, reducing signal level.

The remainder of the network controls the brilliance or brightness of the tibias by determining treble rolloff. Note, of course, that this sort of control does not change the tone color or waveforms of the tibias, which are sine waves in the first place. With a treble rolloff, volume simply decreases at the higher notes. With the FF brightness control on, the lamp for the LDR labeled BRIL has maximum brightness. The signal therefore tends to bypass the 2-section low-pass network below, and uptilting of the treble end is helped by the .0047-μf series capacitor. With the light low or off, the bypass network includes the 1-megohm resistor, so does little bypassing and the low-pass network takes down the higher notes.

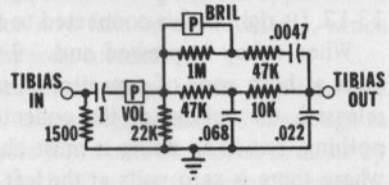


Fig. 13-11—All tibia oscillator outputs go to this volume and brightness control network.

From the Tibias Out point of Fig. 13-11, the signal goes through a line amplifier much like those used for the C and C# manual signals and the pedal signals. It includes an LDR operated by the swell shoe, and its final stage sends the tibia signals to the amplifier which feeds a rotating or fixed speaker, depending on the

position of a selector switch which is on the console available to the organist.

Omega

The last letter of the Greek alphabet, *Omega* (Ω), is used to indicate what Gulbransen calls a percussion system but is actually a sustain for the upper manual. It is simply a method of prolonging the decay of collector voltage applied by the upper manual to each of the oscillators of Fig. 13-10. The Omega circuit is shown in Fig. 13-12.

The points marked Omega in Figs. 13-10 and 13-12 are connected together. That is, for each tibia oscillator there is a circuit like that within the dashed lines

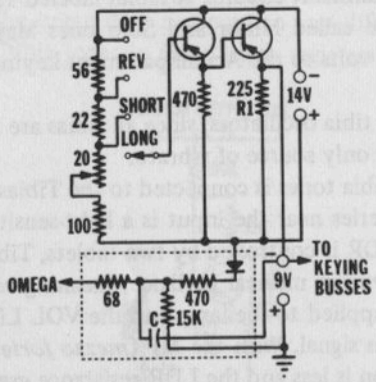


Fig. 13-12—The Solo sustain system (Omega Unit). Parts in dashed box are repeated for every note.

of Fig. 13-12. The 9-volt source shown is that which is placed on all the tibia keying busses to energize oscillators when keys are pressed. In each of the dashed-box circuits there is, disregarding other components, a capacitor C, Fig. 13-12. Its right end is connected to the −9-volt keying supply.

When a key is pressed and −9 volts is applied to the oscillator, there is −9 volts at both ends of capacitor C, meaning that C is discharged. When the key is released, the voltage at the collector of the oscillator wants to recede back to nothing, but in so doing it must charge C. That is, it wants to reach a condition where there is zero volts at the left end of C where it is connected to the Omega point in Fig. 13-10, while the right end of C remains at −9. The difference between the two ends of C can only be reached by charging C. To charge C, electrons must be taken off its left plate, passed through the relatively high resistance of the oscillator transistor to ground and up through the 9-volt supply to the right C plate. This takes time, and until the charge is complete, the oscillator still has negative voltage on its collector and is sounding. The negative voltage is constantly being reduced, however, so that the note fades away gradually over a period of around two seconds.

To reduce the time taken to charge C when the decay is not meant to last so long, an auxiliary circuit is provided whose function is to help charge C through an alternate path. That is, it applies to the left end of C a voltage which is positive with respect to the right end of C. The more charging voltage so applied, the less of the work of charging must be done by the oscillator circuitry and the

faster will the oscillator note fade. This is how sustain duration is controlled.

The two-transistor circuit with the separate 14-volt supply is the controller. Notice that neither end of the 14-volt supply is grounded. The two transistors are a pair of emitter followers in cascade. For a reason you will see in a moment, the whole purpose is to place a controlled voltage across resistor R1. The four-position switch does this. With the switch at Off, the base of the first transistor is just as negative as its collector, with respect to the positive end of the 14-volt supply. There is thus almost 14 volts across the 470-ohm emitter resistor, and only slightly less than that across R1. The voltage across R1 is almost 14 volts, the negative end at the transistor emitter.

The negative end of R1 is connected to the right plate of C and the positive end, through the diode and voltage divider in the enclosed area, to the left plate. As far as the control circuit is concerned, then, the left of C is made quite positive with respect to its right end. When a key is pressed and the Omega point is connected directly to the −9-volt keying supply, the resistances of the control circuit we have been discussing are great enough so this applied voltage to C is swamped out, and the −9 volts is well and truly placed on the oscillator to key it. But when the key is released and the oscillator collector voltage wants to go back to zero—that is, in a positive direction—it does not have to charge C in the process. The voltage applied across C by the two-transistor circuit steps in and charges C, taking the burden entirely away from the oscillator, so that the oscillator stops working quickly, as it would without any of the Omega connected to it.

If the control switch is in the Long position, the first base is unconnected and the transistor is cut off. This means practically zero voltage at both emitters. There is no voltage drop across R1 and no helpful charging current applied to C when the key switch opens. The whole burden now falls on the oscillator, which takes its maximum time to decay and gives the long sustain.

The Short switch position places an intermediate voltage on the first base and thus across R1. Now when the key is lifted, the voltage across R1 is enough to help charge C but not enough to do the whole job. Some of it must be done by the oscillator, so the sustain is of medium length. The Rev (meaning "reverb"—see my comments on pages 128 and 235) position places more voltage across R1 to reduce the sustain effect to its minimum length.

In Fig. 13-12 all circuitry outside the dashed lines serves the entire organ. All the Omega circuits, one for each oscillator, are like that shown within the dashed lines, and all of them connect at the same points to the +14-volt line, the top of R1, and the −9-volt point.

The reason for using an emitter-follower arrangement to supply the helpful charging voltage for C, rather than a simple voltage divider or some other supply method, is that the current needed at any time for charging C's depends on how many notes require help. The two transistors are simply a regulating circuit which maintains a constant selected voltage across R1 regardless of the current drawn. This use of emitter followers as regulators may be seen in the power supply of the Schober Organ (page 205) and in other places.

The Omega unit is not actually controlled by a rotary switch, but instead by five of the seven "piston" buttons beneath the front of the upper manual. These are white buttons which look like combination action pistons, but they are not. The five controlling the Omega are marked Reverberation, Percussion Short, Percussion Long, Percussion Preset, and Ω Cancel. They are interlocked, so that pressing any one releases any other that may be in. The Percussion Preset button sets the Omega for long sustain, kills all nontibia Solo voices, and stops the Leslie rotating speaker.

A sixth piston in this group is for chimes. It stops the Leslie, sets sustain for long, kills all nontibia Solo voices, and also sets up a preselected group of Solo tibia pitch registers which includes the 6⅖′, 2′, 2⅔′, and 4′. The seventh piston is a Sforzando (full organ), which automatically turns on certain stops for both manuals and pedals.

Electro-Mute

One of the weird and interesting features of the Gulbransen box of tricks is the Electro-Mute, which is now on all Rialto models. It is a Solo stop with a tuba sound, and when a knee lever which hangs down from the board supporting the lower manual is moved back and forth by the organist's knee, it makes a *wah-wah* sound like a jazz trumpet when the trumpeter moves the brass derby back and forth over the bell.

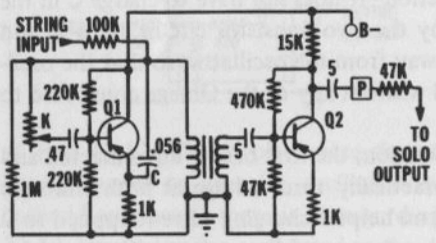


Fig. 13-13—The Electro-Mute produces a "wah-wah" horn sound.

Fig. 13-13 is the Electro-Mute circuit. In the organs where it appears, the Tuba filter is removed from the Solo main voicing, Fig. 13-7, and the Electro-Mute is the substitute. Tone is taken from the string input, Fug. 13-7.

In Fig. 13-13, K is a variable resistor operated by the knee lever. Start by supposing the resistance of K is maximum, so that practically no signal enters emitter follower Q1. In that case, string signal enters through the 100,000-ohm resistor and is placed across an L-C formant filter composed of the transformer primary as the inductance and C as the capacitance. The bottom of C is not connected to ground, but as it is connected to the output of an emitter follower, it is within about 50 ohms of ground, so may be considered to be about as effective as though the 50 ohms did not exist. The signal, with its tuba formant, goes through the transformer, is amplified by Q2, and, through the stop-switching LDR, joins the main Solo output.

Now suppose the knee lever is pushed so that the resistance of K becomes smaller. Signal goes into the Q1 base and emerges from its emitter in the same

phase and almost at the same level. Since signal of the same phase is at both ends of the capacitor, the effect is about the same as though the capacitor were shunted by a resistor, the size of the resistor depending on the relative amplitudes of the signals at both ends of the capacitor. If, for example, the emitter follower output were of the same amplitude as the signal at the top of the transformer, the effect would be like a dead short across C. With emitter-follower output anything less than this, the "resistor" has a finite value, the smaller the emitter signal the larger its resistance.

The result of shunting a resistor across a capacitor is, of course, to change its reactive effect—to, in effect, alter its capacitance; and in this case, that changes the resonant frequency of the tuned circuit and the formant. So as the player moves his knee back and forth, the character of the horn slides from an *oo* to an *ah* and back again. The effect is a surprising *wah-wah* and I have yet to see the listener who didn't smile when he heard it. Bach might not like it, but a dyed in the wool theatre organ fan will almost invariably eat it up—if it isn't used to death.

Electronic Piano

One of the highly distinguishing features of the Rialto organ is the electronic piano accessory which can be factory installed as an option in any new instrument or in any Model K Rialto manufactured after September 30, 1964. As of this writing—early 1967—the piano will not be installed in the field, largely because case work is required to accommodate the controls.

Gulbransen's piano effect is not only unexpectedly realistic but also extremely useful musically. I say "unexpectedly" because most prior attempts at electronic simulation of pianos have been fairly miserable at best. Gulbransen has not succeeded in imitating a piano to the very life, and I am not sure anyone really ever will (economically) because of the extreme complexity of real piano tone and transient effects. But any pop organist will enormously enjoy using this "piano," and the only other one in the industry I know of which compares with it is Schober's (whose designers, by the way, never laid eyes on the Gulbransen circuit until well after the Schober percussion was well into production).

The electronic piano is available on 60 of the 61 upper-manual keys. The system consists of 60 special key switches actuated by the ends of the keys, above which the assembly is mounted, 60 gate circuits, a preamplifier and filter for each octave, a collector preamplifier, and a brilliance and volume control circuit, plus some d.c. proportioning and distribution circuitry operating from the organ's main power supply.

Piano Gates

The important part of the circuitry is the 60 gate circuits, one for each key. When a key is pressed, its gate transmits tone from the appropriate main generator stage to the preamplifiers, control assembly, and the special Leslie speaker system through which piano tone is heard.

The system operates the way a piano would. When a key is pressed and held,

the "hammer" strikes the "string" and the tone begins a long decay. But if the key is released before the decay is completed, the "damper" falls on the "string" and the tone stops. The gate does all this, and attends also to such details as the correct decay time and slope when the key is held and the correct damping time if it is released.

Fig. 13-14 is a diagram of one of the 60 gates. The missing values are those which differ for various groups of notes so that the higher tones have gradually shorter decay times than the lower ones as is true in a real piano.

The transistor is used as a grounded-base amplifier. That is, its base is grounded for audio by C3, and signal from a generator is applied to the emitter through blocking capacitor C4 and isolating resistor R7. R6 is the emitter connection to ground, necessary for transistor operation, but is several times as high as the actual load on the input path, which is the emitter-ground impedance. Output is taken from across collector load R9 through isolating resistor R10.

A stage of this kind has no normal transmission of tone from input to output and very little leakage, unless the base is given very definite on bias, positive in this case of an NPN transistor. (Keyers of this type are also used by Conn,

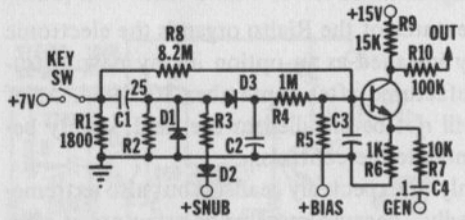


Fig. 13-14—One of the 60 piano gates.

Chapter 19.) When the key switch is open, no positive voltage reaches the base through it or the Snub connection. A small positive bias is applied through the Bias connection. However, this is adjustable by octaves and is set so that just a trifle less bias is applied than would be sufficient to cause the stage to conduct. Normally, then, there is no output.

When the key switch closes, +7 volts is applied to the left plate of C1, causing C1 to charge. The electron path is from the left C1 plate to the positive power supply terminal, and from the negative supply terminal through R2 to the right plate. As C1 begins its charge, the electron flow is maximum, causing a maximum voltage drop across R2, with the top of R2 positive with respect to ground. This positive potential is applied to the transistor base, immediately turning it on to the maximum extent and causing the initial percussive tone strike. As the charging of C1 continues, the electron flow through R2, and thus the positive potential at the top of R2, decreases at an exponential rate. As a result, the transistor conduction decreases similarly, and the tone gradually fades. When C1 has been fully charged, there is no further electron flow through R2, no further positive potential at the transistor base, and no further tone output.

If the key is then released, C1 discharges. To discharge C1, the excess electrons now collected on the right plate must pass to the left plate. They do this not only through R2 but also through D1, which is in the correct polarity for

electrons to pass down through it. Since the conducting D1 is a very low-resistance path compared to R2, the discharge of C1 is a very great deal faster than its charge. R1 is also part of the discharge path. During charge R1 was only a static load on the +7-volt input, having no effect on gate operation, and D1 was in the wrong polarity for the charging electrons to pass through it. Both R1 and D1, therefore, operate during discharge only.

As described thus far, the decay of the tone was exponential, in accordance with the normal charging pattern of a capacitor, C1. Piano tone, however, does not decay exponentially. Immediately after striking, it decays very fast, descending to perhaps half of its peak amplitude in a very short time. Thereafter the decay is much slower and smoother; and the tail end of the decay flares out and is extended still further in time.

D2 and R3 produce the fast portion of the decay immediately following the strike. A so called "snubbing" voltage—either +2 or +3 volts, depending on the note—is fed to the Snub terminal. With the key open, this blocks D2 and produces no effect. When the key is first closed and the initial high charging current through R2 produces a maximum positive potential at the top of R2 (and of R3) the potential is high enough to overcome the snubbing voltage and cause D2 to conduct. This opens up a second path—one in addition to R2—through which C1 may charge. The additional charging path allows C1 to charge more quickly and thus the positive potential applied to the transistor base to decay faster, producing the rather fast initial part of the tone decay. However, when the charge has proceeded long enough (about half the total C1 charge) so that the positive potential at the top of R2 is adequately reduced, there is no longer enough positive potential to overcome the snubbing voltage at the cathode of D2. D2 then blocks and the remainder of the C1 charge is at a slower rate, as is the tonal decay.

R8 takes care of the flared-out tail of the decay by providing a residual small positive potential on the transistor base as long as the key is held down. All by itself, this potential is not enough to hold the transistor in conduction. However, it is enough to augment the last dying positive potential at the base as C1 nears the end of its charge, flattening out the very end of the potential decay.

Thus far we have said nothing of D3, R4, and C2. R4's first purpose is to isolate the d.c. timing circuits from the relatively low transistor base impedance. C3 is solely to keep the base at ground for audio and, being shunted directly by the relatively low base-ground resistance, it has no effect on gate operation.

D3 and C2 are for damping. That is, they only have effect in case the key is released before C1 has reached its full charge and the key-down decay has been completed. If the organist does lift the key before the tone is all gone, he expects to hear it stop relatively quickly, though not instantly, as though the felt damper had fallen on the string. If it were permissible to have the tone stop almost instantly, C2 would not be needed. Lifting the key would cause C1 to discharge immediately and that would remove all positive potential from the transistor base at once. It would also sound quite wrong.

When the key is pressed and C1 begins to charge, a part of the initial charging

current also comes from the negative end of the power supply (ground) through C2 and D3. C2 is very much smaller than C1. Because of this, it takes very little time to charge and does so almost at once, so that after the first few instants it is fully charged and does nothing further. Since it is so small, its upper plate remains, during the rest of the C1 charging cycle, charged to whatever positive potential exists at the top of R2. If the key stays down, C2's charge merely follows the R2 potential and contributes nothing.

If, however, the key is released while there is still positive potential across R2, C1 is immediately discharged and no positive voltage remains at the left of D3. The positive charge on C2, however, remains at whatever value it had at the moment the key was released, and that charge is still applied to the transistor base, causing the transistor to remain in conduction. The positive charge at the top of C2 cannot dissipate through anything to the left of D3, as the D3 anode is now at ground potential and a positive C2 charge on its cathode causes it to block. The charge on C2 must now dissipate through R4 and the transistor base-ground resistance, a rather high-resistance path. It thus takes an appreciable time for the C2 charge to dissipate and for the sound to decay to nothing. And C2 is so proportioned that this time is about the same as it takes a string to stop vibrating when a thick piece of felt is dropped on one part of it.

The values of R2, R3, and C2 get smaller as pitches get higher, since all decay times must be shorter for authenticity. An adjustment is provided for the +7-volt keying potential, which is derived from a 9.1-volt zener diode through a transistor regulator, since this must be exactly right. The bias can also be adjusted for each octave of gates to a trifle more than that which will cause complete cessation of tone when the keys are up.

Each octave of gate outputs is fed to a simple one-stage voltage amplifier, which is followed by a tone filter consisting of some high-pass and some low-pass action. Since the manufacturer does not furnish values, it seems useless to show the diagram. All five octave amplifiers feed a single two-stage preamplifier through an LDR operated by the swell shoe so that the piano volume follows general organ volume.

The control assembly is inserted after the preamplifier to provide two kinds of control. Organs equipped with the electronic piano have two drawbars not unlike the familiar Hammond ones, which operate variable resistances. One controls the volume and the other the brightness of the piano effect.

The piano stop tablet is unusual in that it has a center position between the up and down positions. When it is up, the output of the control assembly is shorted to ground and the piano is not heard. This short is removed in center and down positions. In center position the piano is heard in addition to any normal organ stops turned on. When the tablet is all the way down, Solo tibia keying voltage is interrupted, as is also voltage to the magnets which turn all the other Solo bus rods to their on positions, so that all Solo tone except piano is cut off. The idea is very useful to the player, but none of the circuitry is remarkable enough to warrant diagramming.

Chapter 14

Wurlitzer Organs

Where in the ranks of theatre organ enthusiasts breathes there a soul whose heart does not skip a beat at mention of the "mighty Wurlitzer?" The phrase does not, of course, refer to the Wurlitzer organs of today, but to the great* pipe instruments of yesteryear that graced so many silent movie theatres across the land and helped in such an important way to found and popularize the tradition of theatre organ music.

Perhaps no other company made so many theatre organs—many of which are in private hands now that the theatres have been almost entirely cleaned out—and few if any other manufacturers presently in the electronic organ business have such a long history in the world of music. Nicholas Wurlitzer, born in 1659, is said to have started the musical history of his family as a maker of lutes and later violins. In 1853 Rudolf Wurlitzer came to the United States and three years later founded the company which until recently bore his name and is now called simply The Wurlitzer Company.

Wurlitzer has for many years manufactured musical instruments of various kinds, in addition to pipe organs (a thing of the past) and electronics and maintains its own retail stores in a number of cities, though in other places its products are retailed by noncompany stores. The company does not handle string instruments, though another member of the family who died not long ago, Rembert Wurlitzer, was one of the world's best known dealers in and collectors of fine violins, violas, and cellos.

Wurlitzer entered the electronic organ field shortly after the war with a series of instruments based on wind-blown reeds with electronic pickups, a technique which it took over from Everett Piano Company, which had manufactured the Everett Orgatron prior to the war under patents of Benjamin F. Miessner. (Everett Piano is now owned by Hammond, such are the involutions of business and finance.) An example of the Wurlitzer reed-type organ is described in the last edition of this book, and the reed pickups are discussed briefly on pages 41 and 42 of the present opus.

All Wurlitzer product engineering is done in North Tonawanda, N. Y. Sales headquarters is DeKalb, Ill., and organ manufacture is in Corinth, Miss. Not many years ago, management decided to go into purely electronic organs. For a time both reed-electronic and pure electronic instruments were made, but the reeds have now been phased out, as the saying goes, and Wurlitzer is solidly in

*Great by theatre standards, that is, where a 30-rank organ was a monster. Church organs are often, of course, much larger.

pure electronics. I am sure the company will be delighted to learn that I heartily approve of this change.

The Wurlitzer organs of today are immersed in the fiercely competitive market of today. They are better than some and not so good as others, and so has the "mighty" fallen. But a good deal of interesting engineering has gone into them. Fig. 14-1 shows the largest organ Wurlitzer has marketed since the pipe days, the Model 4700 Concert Organ retailing for around $4600, from which you may correctly infer that the company is more interested in moderately



Fig. 14-1—The 4700, a concert-type organ, is today's largest and most expensive Wurlitzer.

priced instruments for the home than in the Allen-Rodgers end of the scale. Its smallest instrument is about $700 list, and the 4140 spinet, which we shall describe in this chapter and is pictured in Fig. 14-2, runs about halfway between one and two thousand.

The 4140 has the usual two 44-note keyboards, with overhanging organ-type keys, and 13 pedals. It has voices in four pitch registers on the upper manual, one (with a coupler to add a second) on the lower, and two for the pedals. In addition to a pair of regular 12-inch speakers plus a 4-inch unit for treble, it has a pair of 4-inch speakers which rotate, spraying sound out the left end and top of the console. This system, almost the only rotating speaker arrangement in the industry which is not a Leslie, is called Spectra-Tone. Unlike the Leslie idea, it is the treble which is switched between rotating and fixed speakers.

There is a sustain for 8′ voices of the upper manual, which can also be played with the lower manual, and there is also a repeat percussion plus two attack percussion effects. The Wurlitzer Ssh-Boom is included, crash and sizzle cymbal and bass drum sounds activated by playing keys and pedals, as well as a full manual of tuned bongo drums. The organ has 18 speaking stops, plus an arrangement for emphasizing any pitch registers of either manual.

In advertising for Wurlitzer organs two terms appear which, as a technical reader familiar with organs, I found puzzling and which you may sometime be



Fig. 14-2—The 4140 spinet described in this chapter, in contemporary console. Four other console styles are available.

asked to explain. One is "automatic tone controls." These simply refer to the stop tablets, which are, of course, no more automatic than anyone else's stop tablets. The theory is that all conventional stop tablets are "automatic" by comparison with Hammond's drawbars. Pressing a stop tab "automatically" produces the tone lettered on the tab, while with drawbars combinations of several must be set to obtain any tone color.

The second term is "Total Tone." This means in essence that since all Wurlitzer organs make available both square waves and quasi-sawtooth waves (and some also genuine sawtooth) for presentation to the tone coloring circuitry, the organs can produce tone colors of all the types needed for any sort of music.

Both terms are somewhat strained if you ask me, but Wurlitzer didn't ask me.

Tone Generators

The Wurlitzer 4140, like all modern instruments of this company, has frequency-divider generators. The generator system provides a total of 67 notes. The highest octave generated is No. 7 in Table 1-1, making the 4′ register of the upper manual complete except for the highest C. The lowest note for generators F through B is in Octave 2, Table 1-1, but since the lowest key of the lower manual is the second F below middle C, the C through E of Octave 2, Table 1-1, is not needed and these generators end at Octave 3. The seven generators for F through B, therefore, have oscillator and five dividers, furnishing a total of six octaves, 2 through 7; the C through E generators have only four dividers, furnishing five notes, Octaves 3 through 7. As we have done throughout the book,

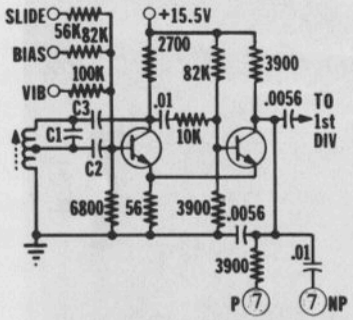


Fig. 14-3—Oscillator section of one Wurlitzer tone generator.

we shall identify any pitch by its octave number in Table 1-1 rather than by its ordinal number in the actual generator. In this organ, seven of the generators produce no notes in Octave 1 and the other five produce nothing in Octaves 1 or 2. This represents a good range of generated tones for a spinet organ, in contrast to many which get by with an octave less at the high end—or even more skimping than that. It completes the 16′ register at the low end except for the lowest half-octave of Accompaniment and the 4′ register for the top end except for the top C. The instrument as a whole has a 7-octave range, since its pedal division employs two cascaded frequency dividers triggered by main generator Octave 3, and so covers pedal tones right down to the 32.7-cps C.

The first transistor of Fig. 14-3 is one of the 12 master oscillators. C1, C2, and C3, with the position of the movable coil core, determine the tuning, but the exact pitch is also sensitive to the base voltage. A fixed, regulated bias is applied at the Bias terminal to keep this steady.

A low-frequency vibrato signal is applied when desired to the Vib terminal; since it varies the voltage on the base at a vibrato rate, it modulates the oscillator frequency. A "slide" effect like Lowrey's "glide" (page 231) is obtained when a positive voltage normally on the Slide terminal is lowered; the result is to flat the pitch by a semitone. Vibrato automatically stops during the flatting. These three terminals are common to all 12 oscillators, the resistors between terminals and base individual to each oscillator.

Fig. 14-4 shows where these voltages come from. A power-supply source of +15.5 volts is used as generator B+ and is also applied through a 1,000-ohm resistor to the emitter of a diode whose base is grounded and collector unused. Employed in this way, the emitter-base section of the transistor connected to the voltage source in reverse acts as a zener diode, so that the voltage at the emitter is regulated. The regulated voltage, about +10, is applied directly to the bias terminal to stabilize the oscillator.

The +15.5-volt source is also the source for a voltage divider to ground, consisting of R1 as the series leg and the 56,000-ohm slide resistor (Fig. 14-3) and base-ground impedance as the shunt leg. At the Slide terminal, which is the tap on this divider, the normal voltage (with the Slide switch open as shown) is about +15. The Slide switch is a toe-operated device fastened to the swell shoe. When it is used, the two sections shown in Fig. 14-4 close. The lower section simply grounds the output of the vibrato oscillator-amplifier (which we are not showing because it is conventional), thus removing vibrato. The upper section shunts R2 from the Slide terminal to ground, reducing the voltage at the Slide terminal and flatting all the oscillators.

The voltage at the Slide terminal is also the charge on C. When the switch closes, C must discharge somewhat to lower the terminal voltage, and this takes a small amount of time since the discharge is through resistor R2, factory selected for correct average frequency change. When the switch opens again, C must recharge through R1, again giving a little delay. The aural result is an oscillator

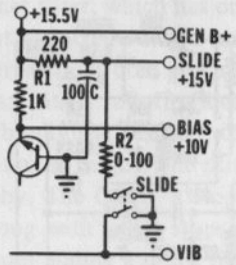


Fig. 14-4—Central control for slide, bias, and vibrato.

pitch change which is fairly smooth rather than instant. As normally used—to imitate a Hawaiian guitar effect or something of the kind—the switch is closed just as a key is pressed and then immediately opened when the key is down.

Returning to Fig. 14-3, the second transistor is a squarer, a symmetrically overloaded amplifier. Its collector furnishes the top octave of tone for the organ. You will notice two outputs. One of them, taken through a capacitor which is solely for d.c. blocking and has no filtering effect, is for the *nonpercussible* keying system; the other, taken through a resistor since the positive potential of the generator is needed, is for the diode or *percussible* keying system. What these two terms mean will, I hope, be clear shortly. Trigger signal for the first divider is also taken from the squarer collector.

The frequency dividers are flipflops, as is the fashion today, though, as Fig. 14-5 indicates, the triggering method here is a little unusual. Most organs introduce trigger pulses into the flipflops through a common resistor to the B-supply.

Wurlitzer uses the more classic and practically defunct method of feeding sync directly to the transistor bases, through small individual capacitors except for the first divider, which has resistors. I know of no reason why one method is

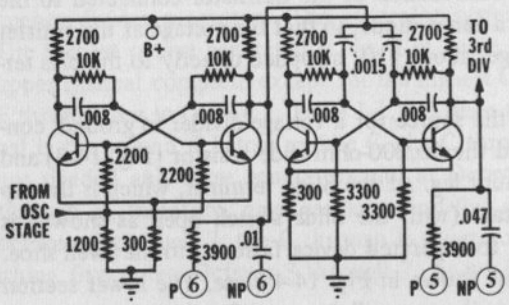


Fig. 14-5—First two dividers of a tone generator.

better than the other, though my personal experience has been better on stability grounds with the other.

As at the top-frequency squarer, each of the dividers has two outputs, one for the percussible (P) and the other for the nonpercussible (NP) keying systems.

Nonpercussible Keying

There are actually three separate keying systems in this organ, which we shall discuss separately. The first concerns five stops of the upper manual, all stops

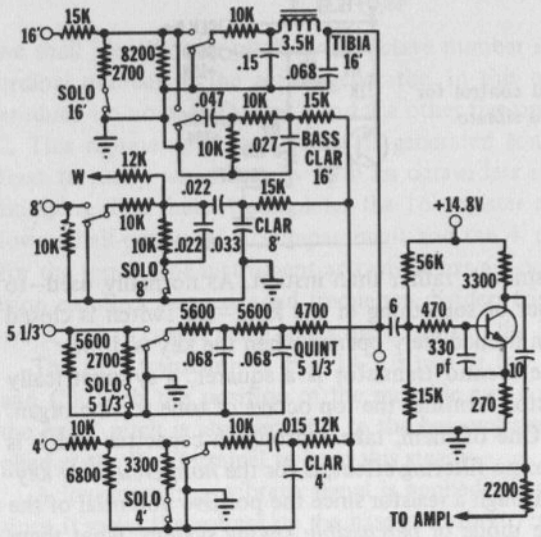


Fig. 14-6—Voicing filters for the upper-manual "nonpercussible" voices.

other than certain of the 8′ voices. This is a "normal" keying system, for which each playing key is provided with a schematically ordinary quadruple switch. When a key is pressed, four contact fingers touch four busses running the length of the keyboard. The four fingers and busses are for 16′, 8′, 5⅔′, and 4′ pitch registers. Each finger is connected to a generator output of appropriate pitch

through a series isolating resistor. All the isolating resistors are 100,000 ohms, except that for the 8′ switch fingers they are scaled and run from 150,000 ohms to a megohm. The tones handled by this part of the keying system are the "nonpercussible" ones taken from the NP outputs in Figs. 14-3 and 14-5. They are so called because they are straight organ voices which are not subject to sustain or percussion.

Fig. 14-6 shows what happens to the outputs of these four keying busses. The 16′ bus, for example, is terminated to ground (after a 15,000-ohm series resistor) by 2700 and 8200 ohms in parallel.* When the Solo 16′ switch is opened, part of the resistive termination to ground is removed, and the amplitude of tone increases, making the 16′ stops louder than they normally would be, compared to other stops.

The 16′ tone goes to two voicing filters. That for the Tibia 16′ is an L-C low-pass filter with a cutoff at about 325 cps. As we see in other chapters, this is one way of lopping off harmonics to get a tibia tone, and it should not be confused with a reed filter which usually employs a tuned circuit to create an emphasis or formant over a certain part of the spectrum. On the upper manual, 16′ tone ranges from 87.3 to 1046 cps, so that a cutoff filter is a distinct compromise. It does a fairly good job of removing harmonics in the first half of the keyboard, where the tonal character is most important, without resulting in fundamental attenuation. However, above cutoff it starts removing fundamental as severely as harmonics, especially as there is no counterscaling in the key-switch resistors.

The 16′ tone also goes to the Bass Clarinet filter, which has one low-pass section and one high-pass. There is no formant here created by a tuned circuit, but the combination of low- and high-pass filter sections does give a region between around 300 and 600 cps which is fairly flat with a rolloff above and below it, and thus provides a broad formant. Undoubtedly the designer wanted a clarinet with very little bite, and undoubtedly that is what he got. The other two clarinet filters operate on much the same philosophy. The Quint, a stop at a mutation pitch which is always intended for use along with other stops, purely for the distinctive tone coloring effect one gets from adding a quint pitch, is a simple treble rolloff beginning at around 500 cps to obtain the normal character of a rather dull diapason such a stop should have.

All these nonpercussible stops go to the amplifier (along with other items to be shown later), two of them through the one-stage amplifier shown. Point X is another input to this amplifier whose purpose we shall get to farther on. Point W is a takeoff point for some nonpercussible 8′ tone, for a purpose to be discussed.

Electronic Keyers

All remaining keying is done by diode keyers for the 8′ upper-manual stops (except Clarinet) and the lower-manual stops, all of which are at 8′ pitch. The keyers serve not only to turn the tones on and off, both normally and with the

*Please note carefully that in this chapter—and in others unless specifically mentioned—all switches are shown in their off or normal positions. Thus the Solo 16′ switch is shown in its closed condition, which is how it is when the Solo 16′ tablet is off.

slow decay needed for sustain. They also act as waveshapers, transforming the square-wave generator outputs to signals which have the even harmonics needed for the normal tone-color filters. Fig, 14-7 shows upper- and lower-manual keyers for one generator output. Let us talk first about the lower-manual keyer since it is the simpler, lacking provision for sustain.

The generator signal must get through both D1 and D4 to reach the keyer outputs labeled Flute and Complex through capacitor C. The signal goes through two diodes instead of one simply because when the diodes are blocked (the key not played) two diodes permit less leakage than one, especially since the .001-μf capacitor following D1 tends to kill the higher harmonics which leak the most. When the diodes are made to conduct, D1 and D4 in series act just as a single diode would; and at that time D1 is no longer a high resistance so the .001-μf capacitor does not act as the shunt leg of a low-pass filter and does not bypass highs significantly. We can thus, for our explanation, ignore D1 and its following capacitor, assuming that the generator signal is connected directly to the cathode of D4.

The square-wave output of each generator stage is really two values of positive direct voltage, the frequency divider transistors switching this voltage abruptly from one value to the other at a constant repetition rate. For one half of each

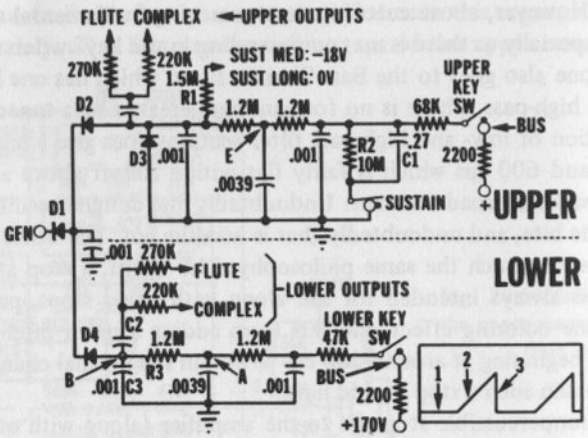


Fig. 14-7—Upper- and lower-manual keyers for one tone. Output waveform is in box.

cycle the generator is at +1.5 volts and during the next half it is at +13. When the key switch is open, D4 is blocked because the positive generator signal is connected to D4's cathode. For D4 to conduct significantly, a potential must be applied to its anode which is at least 0.5 volt more positive than the voltage at its cathode.

Let us assume now that the key switch has been closed long enough for the +170 volts to charge the .0039-μf capacitor and appear full strength at point A. Let us say that at this instant the generator potential is at its +1.5-volt value. Since the positive voltage applied from point A through R3 to the diode exceeds

this 1.5 volts by well over 0.5 volt, the diode immediately conducts. Its resistance is very small, and the resistance in the generator is likewise very small with respect to R3. Thus, considering point A as the potential source, the 170 volts is placed across a voltage divider consisting of R3 as the series leg and D4 plus the generator internal resistance as the shunt leg. The shunt leg (D4) is variable; it can conduct more or less, depending on the voltage across it.

If its resistance is high enough so that the voltage across it tends to be more than 0.5 volt, enough current will flow through it to reduce its resistance.

The diode will, in fact, adjust its resistance so that just 0.5 volt drop is maintained across it. A higher voltage drop would cause more current to pass through it; and this causes a diode to act as a lower resistance, bringing down the voltage drop to the 0.5-volt figure. A smaller voltage drop would cause it to stop conducting so that it would no longer act as a shunt in the voltage divider and the voltage would rise. A diode operating in this way is said to have a *clamping* action. It clamps the voltage across it at a set point. Thus, during the half signal cycle when the generator voltage is at its more negative (actually less positive) value, the potential at point B is 2 volts with respect to ground—0.5 volt more positive than the +1.5 volts applied by the generator to the diode cathode.

At the beginning of the second half-cycle, the generator voltage abruptly becomes +13 with respect to ground. If capacitors C2 and C3 were not present, the potential applied from source point A would also immediately rise until it reached a point again 0.5 volt higher than the generator voltage, at which point it would again clamp. The waveform at B would be a square wave just like the generator signal, varying between +2 and +13.5 volts.

However, C2 and C3 provide a time-constant effect. When the generator signal rises to +13 and the diode shuts off, the rise of the source-A potential at point B is slowed by the R-C time constant—R3 as the resistance and C2 and C3 as the capacitance. The potential at B therefore rises slowly, not instantly. In the box you can see the waveshape at B. Part 1 is the period during which generator potential is at +1.5 volts and point B potential is at +2. Part 2 is the gradual rise of potential during the positive portion of the generator signal.

The C of the time constant (actually capacitor C2) has been proportioned for any note so that part 2 of the point-B wave reaches almost exactly its maximum possible value of +13.5 volts at which it would clamp, at the time the positive portion of the generator cycle ends. At that instant, generator voltage snaps back again to +1.5 volts. The diode is again a low resistance which instantly readjusts itself (line 3) to clamp the point-B potential back again at +2 volts. (The capacitors quickly discharge through the low-resistance diode.) The procedure continues as long as the key is held down, producing the waveform in the box.

The exact slope of the rise of voltage during part 2 of the wave is determined by the keying voltage and the R-C values. In the organ R3 and C3 are the same for all notes, so C2 is the component adjusted for each one. If the C were too small, part 2 of the wave would rise too fast and reach its possible maximum with the generator cycle still at its positive extreme. This would cause a flat top on a portion of the wave, as though it had been run through a positive clipper. If the

C were too large, and the rise of part 2 too slow, the voltage reached by part 2 at the time the generator signal went back to its low-voltage half-cycle would be less than the possible maximum. The amplitude of the wave would therefore be smaller, though the shape would be unchanged.

This latter fact can be used to control the over-all keying envelope. When the key is actually closed, the full +170 volts does not reach point A immediately. Instead, it builds up slowly because of two cascaded time-constant circuits. The first—47,000 ohms and .001 μf—has little delay and just kills the sharp initial attack to prevent a click or thump in the output. The second portion, 1.2 megohms and .0039 μf, slows the entire buildup of keying potential at A to the attack the designers desired. While the voltage at A is in the course of building up and is lower than its final value, the slope of part 2 of the waveform at B is more nearly horizontal, making the signal amplitude smaller than its eventual value *without changing the waveshape.* A lower voltage at A thus has the same effect as a larger capacitor at C—lowering amplitude and leaving waveshape unchanged.

The same possibility is used for the sustain action which can be had in the keyer for the upper manual. With the Sustain switch open, the upper keyer is the same as the lower one and works the same way. To turn it into a sustain system, it is only necessary to arrange for the potential at point E (the equivalent of point A) to recede slowly when the key is released rather than almost immediately. Ignoring R1 for the moment, we do this by closing the Sustain switch, grounding C1. With C1 grounded, it charges to the full keying voltage while the key switch is closed. When the switch opens, it is still charged and still supplies potential to point E so that signal continues to pass. C1 slowly discharges through R2, and equally slowly the signal fades away.

A medium sustain is provided by R1, an external source of −18 volts, and D3. D3 has no effect unless this medium-sustain voltage is applied to R1, since its cathode is up and none of the positive voltages can pass through it.

When the −18 volts is applied, it has no effect while the key switch is closed, as the high positive keying potential swamps it out. The net potential at the cathode of D3 is still positive, so D3 is blocked.

However, when the switch opens the −18 volts passes through R1 and the 1.2-megohm resistors to speed the discharge of C1 and hasten the decay of signal. When the net potential at the anode of D2 is less than +2 volts, signal stops. D3 begins to conduct and form a short to ground after this point, simply to prevent C1 from being charged negative and thus slowing up the tone attack when the key is next closed.

A point of some interest about this whole scheme is that in actual fact it is not the generator signals which appear in the outputs at all. Instead, it is energy from the source of keying voltage. The only function served by the generators is to block and unblock the diodes when certain values of keying voltage are present. Another point of interest is the circuit's ability to maintain a constant waveform during the attack and decay periods, while keying voltage is less than maximum. In more usual diode and transistor keyers, where the keying voltage merely sets the effective conductive threshold of the active element and there is

no clamping action, only a square wave can be varied in amplitude without changing shape.

Keying Details

So far we have simply shown how the keyers themselves operate. Now let us descend to the nuts and bolts of what is keyed. In the following discussion, all key switches mentioned are in addition to the four under each upper-manual key which took care of the stops in Fig. 14-6, and we shall simply not mention those.

Under each upper key are three contact fingers, each contacting one of three busses running the length of the manual. Each of these takes care of a pitch register. For example, the bus shown at upper right in Fig. 14-7 may be the 8′ upper bus. If so, the finger under each key which touches that bus when the key is pressed is connected to the keyer carrying tone of 8′ pitch for that key. If the drawing happens to be of the keyer for 440 cps, the switch is under the middle A key.

There is also a 16′ bus, with its own finger under each key, these fingers connected to keyers an octave lower. And there is a rank of 4′ fingers and a 4′ bus. Most keyers, therefore, are connected to three fingers—an 8′ finger under one key, a 16′ finger under the key an octave higher, and a 4′ finger under the key an octave lower.

Each of the three busses is constructed on the same principle as the Gulbransen busses in Fig. 13-6. That is, it can be rotated. The outer shell is insulated, while the conducting bus itself protrudes through only one side of it. In Fig. 14-7, for instance, the conducting part of each of the key switch busses (represented by the black dot on the perimeter of the circle) is up; when the switch finger comes down it will make contact with the conducting metal of the bus. However, if the bus were rotated clockwise, as is that for the Sustain switch, the switch finger touches only the insulation. In the illustration, therefore, the Sustain switch is electrically open, even though the finger is down against the bus.

Of the three upper-manual busses, only the 8′ bus is normally in such a position that the switch fingers contact its metal. Thus the normal effect of upper-manual keying is to key 8′ keyers. The tone filters fed by the combined outputs of the keyers therefore produce what may be and are called 8′ stops.

When the Upper To Upper 16′ coupler tablet is turned on, the 16′ bus rotates (through a mechanical coupling to the tablet, not with an electromagnet) so that the 16′ fingers will now touch the conducting metal of the 16′ bus. When a key is pressed, positive on bias then goes not only to keyers carrying 8′ pitches but to those for 16′ pitches as well. The Upper To Upper 4′ coupler works the same way. This all amounts to normal intramanual coupler action, though only for these 8′ percussible stops. It is abnormal in that it does not do anything to the nonpercussible stops discussed before. The 8′ bus can also be made to turn when the Upper 8′ Off coupler is used; in that case the bus turns its conductive metal *away from* the fingers, so that the stops no longer sound at the normal 8′ pitch but only at 4′ or 16′ pitch if one of those couplers is on. A more usual

name for the Upper 8′ Off coupler would be Upper or Solo Unison Off.

Lower-Manual Keying

The lower manual has only three fingers under each key, with a corresponding trio of rotatable busses. The 8′ bus is normally turned so its fingers make contact. Each of these fingers leads to a keyer handling an 8′ pitch, as shown in Fig. 14-7, and the keyer outputs are combined into 8′ stop filters as for the upper manual.

There is also a 4′ bus for the lower manual, operated by an intramanual coupler labeled Lower To Lower 4′. It operates just like the corresponding upper coupler. And there is a Lower 8′ Off tablet, too.

The third lower-manual bus is an intermanual coupler. Each of its switch fingers is connected to an *upper-manual* keyer at 8′ pitch. So when this bus is turned to the on position by the Upper To Lower 8′ tablet, upper-manual stops sound from lower-manual keys—and the sustain effect is obtained as well, at no extra charge, if it has been turned on.

One last point of clarity about the Sustain switch. As you can see, there has to be a separate Sustain switch finger for each keyer. To turn sustain on and off all these switches must be opened or closed at the same time, but it has nothing

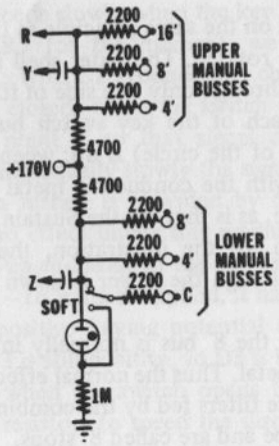


Fig. 14-8—How keying voltage is supplied to rotatable busses.

to do with operation of the playing keys. To handle this multipole switching action, there is a rotatable bus like the others, with its metallic strip grounded. As a part of the regular key-switch assembly there is also a finger for each keyer, connected to the junction of the R2 and C1. Unlike the rest of the switch fingers, however, the sustain fingers are permanently touching the sustain bus; they do not go up and down with key motion. To turn on the Sustain, then, the tab need only rotate the bus, and as many fingers as there are keyers immediately come in contact with the metal of the bus and are grounded. In other words, this is just a 44-pole, single-throw switch integrated with the key-switch assembly for convenience but functionally unrelated to it.

Fig. 14-8 shows the source for the keying voltage carried by the rotatable

keying busses. The 170-volt supply is connected through a 4700-ohm resistor and individual 2200-ohm resistors to the three upper busses. (Ignore connections R, Y, and Z for now.) The 170 volts is also connected to a voltage divider consisting of a 4700-ohm resistor and a neon lamp and 1-megohm resistor to ground. Voltage for the three lower busses is taken from this divider at a point where it is almost the full 170 volts.

The neon lamp is actually a voltage regulator, since such a lamp tends to maintain a constant drop across it. When a tablet called Upper To Lower Coupler Volume Soft is turned on, it not only rotates the coupler bus C to its working position but also changes the position of the switch shown. This places a fixed lower voltage on the coupler bus so coupled keyers sound at lower volume.

We now have keyers for all the notes of each manual, the outputs of which must be combined for the voicing circuits. The manner of combination would make a boring schematic, so a few words will suffice.

You will note in Fig. 14-7 that each keyer has two outputs, labeled Flute and Complex. The two signals are, of course, identical, and both are the quasi-sawtooth waves shown.

The "complex" outputs, those intended for use in producing nonflute voices, are simply connected, through their isolating resistors, such as the 220,000 ohms shown in Fig. 14-7, to four common busses for the upper-manual keyers. Each pair of busses is joined resistively and an output taken from the junction, resulting in a total of two complex outputs for the upper manual. The individual keyer output resistors are graded, and this, with the resistors in the busses, results in scaling correction at the outputs.

The individual flute output resistors of the upper keyers are connected to four different output busses. One bus picks up tone from the lowest 20 keyers, the last three from 12 each. Each bus has inserted into it at intervals series resistors and shunt capacitors to ground. Thus we end up with four flute outputs for the upper.

The lower manual is treated in the same way, with the flute outputs divided into three groups which are the same as those of the upper but lacking the highest 12 notes. And two complex outputs are also derived in the same way as for the upper.

Upper-Manual Percussible Voicing

There is nothing simple about the voicing in this organ. The advertising brochure shows color photographs of engineers recording pipe sounds, making three-dimensional models of them for analysis, then listening in an anechoic room to the electronic reproductions from the organ design, and watching the waveforms on oscilloscopes to determine the accuracy of the imitations. While I have never personally found that all these fancy goings-on produce any better electronic sounds than a canny engineer with a highly trained musical ear, I can well believe the story of the photos—not so much in terms of the final aural results, which I do not find distinguished, but on account of the intricacies of the voicing circuitry, into which I am about to lead you on safari. In the usual organ the cir-

cuitry is fairly simply organized, with roughly the same treatment accorded all voices, or at least groups of them. Here it is apparent that whatever the engineers felt had to be done by way of special treatment for each voice was done.

Let us examine Fig. 14-9, a diagram of the voicing for the five upper-manual stops controlled by the diode keying system. All of them are at 8′ pitch, with 4′ and 16′ available with coupling.

The four flute outputs of the keying system are fed to the four L-C low-pass filters shown, so that at their common junction, point F, all tones appear, each group having upper harmonics lopped off and therefore sounding fluty.

The two complex outputs are fed to an amplifier at the output of which, point C, all the neo-sawtooth waves appear.

The Flute 8′ stop has a double-pole switch. When it is on, two things happen. The upper section takes some tone from point W, Fig. 14-6 (the ordinary, non-percussible, keying), and feeds it to the input of the complex-wave amplifier, along with the two complex inputs from the diode system. The lower switch section passes collected flute signal from point F through 5600 ohms to point OP, which is the output amplifier for the Solo percussible division.

The Diapason 8′ also has a dual switch. The lower, principal switch section picks up complex tone from point C and passes it through a normal low-pass voicing filter to the output amplifier, first passing it through a subsidiary amplifier stage which is used for nothing else. The upper switch section picks up some tone from the No. 4 (highest-octave) flute input and passes it on to point OP

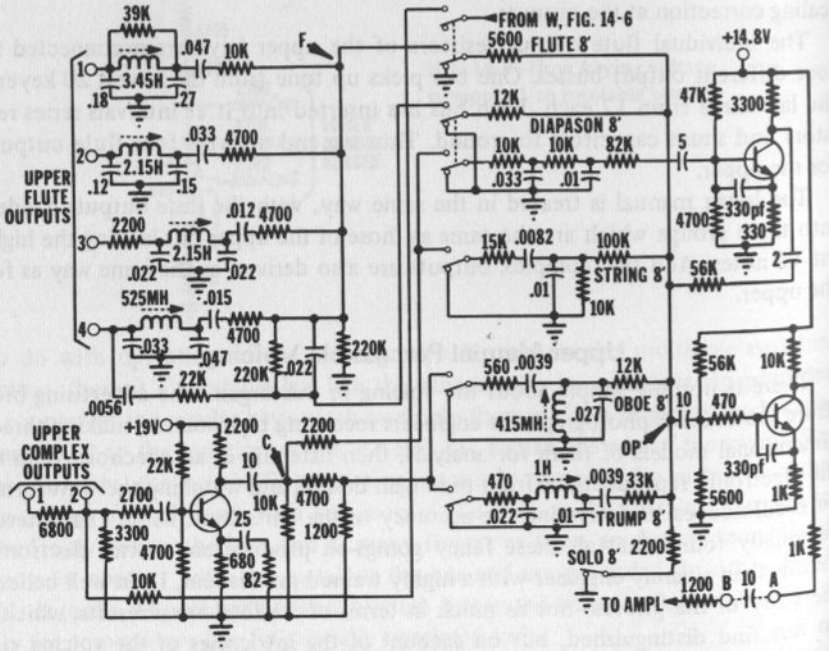


Fig. 14-9—The voicing circuitry for the "percussible" stops of the upper manual.

through a 12,000-ohm resistor to brighten the top end of the Diapason stop.

The String stop is refreshingly normal. It takes complex tone from point C, passes it through some high-pass filtering, and donates it to the output amplifier at point OP.

The Oboe and Trumpet stops are also relatively uncomplicated, deriving complex tone from an additional voltage divider connected to point C. The Oboe filter is the customary shunt L-C formant, while the Trumpet filter has a sharp L-C cutoff which imparts a horny quality but leaves lots of body.

You will recall in Fig. 14-6 and its description that each nonpercussible pitch register of the upper manual is equipped with a Solo tablet which increases the volume of stops of that register. Since the nonpercussible stop group included one 8′ stop, a Clarinet, this Solo 8′ switch was shown as a part of Fig. 14-6. To take care of the stops in the percussible group of Fig. 14-9, however, another section is necessary on the switch operated by this tablet. That section is shown. Normally closed, it loads the stop filter outputs going to point OP with 2200 ohms to ground. When the tablet is used, this switch opens, removing the 2200-ohm load and causing signal amplitude to increase.

Output of the upper-manual percussible-signal amplifier normally goes to the organ amplifier through a couple of resistors and a capacitor. As we shall find later, however, sometimes these signals do not go directly to the amplifier, but instead through the attack and repeat percussion system. That is why we have shown terminals A and B and the dashed capacitor connection in Fig. 14-9.

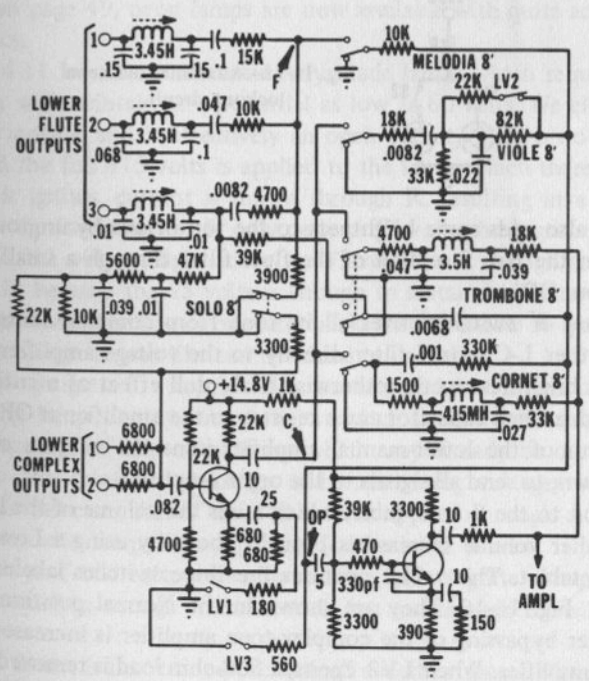


Fig. 14-10—Schematic shows how tones from lower-manual diode keyers are obtained.

Lower-Manual Voicing

The lower-manual voicing system shown in Fig. 14-10 is actually somewhat more complex than the other. The three flute outputs from the keying system go into the three tuned low-pass filters and all notes meet at point F. The two complex voices again go through an amplifier, the output of which, point C, carries all square-wave tones. The input to this amplifier also carries some flute tone derived from the No. 3 flute filter to brighten the top ends of the stops.

A Solo 8′ switch is provided for the lower manual. One of its two sections loads the combined flute filter outputs with 3900 ohms to ground; the other loads the complex amplifier input with 3300. When the tab is used and the switches open, these loads are removed and both flute and complex tone amplitudes rise.

Again, let us analyze the four lower-manual stops all this is in aid of. (This, by the way, is the entire organ voice complement of the lower manual, a surprisingly small number of stops for an organ of this price and elaboration.)

The Melodia 8′ tablet switches total flute tone from point F through 10,000 ohms to point OP, which is the input of the output amplifier for this manual. The Viole 8′ switch sends complex tone from point C through a high-pass filter (ignore the LV2 switch) to point OP.

The two-section Trombone 8′ switch takes complex tone from point C and passes it through an L-C cutoff filter directly to the output of the lower-manual

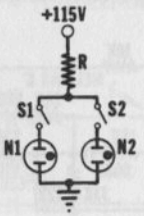


Fig. 14-11—A familiar old neon lockout circuit.

amplifier; it also adds some brightness to the top octave by ungrounding a connection from the No. 3 section of the flute filter through a small capacitor to amplifier input OP.

The Cornet 8′ switch derives all its tone from complex source C. It goes through another L-C cutoff filter directly to the voltage amplifier output, and also, to add brightness to the otherwise fairly dull effect of a cutoff-type reed filter, through a small capacitor and a resistor to the amplifier at OP.

The output of the lower-manual amplifier joins the outputs of two of the stops, as shown, to send all signals to the organ amplifier.

In addition to the Solo 8′ tablet which raises the volume of the lower-manual stops, a further volume increase is brought about by using a Lower Keyboard Normal-Full tablet. This tablet operates the three switches labeled LV1, LV2, and LV3 in Fig. 14-10. They are shown in the Normal position. When LV1 closes, emitter bypassing of the complex-tone amplifier is increased, raising the gain of the amplifier. When LV3 opens, a 560-ohm load is removed from the input OP of the output amplifier. And when LV2 opens, the output resistor of the

Viole 8′ stop is increased in net value to prevent that stop from overbalancing the others.

Pedal Switching Principle

The pedal tones of the Wurlitzer 4140 are furnished by a pair of cascaded flipflop frequency dividers working from switched main generator tones in Octave 3. Output of the first divider furnishes 8′ pedal pitches (Octave 2) and output of the second flipflop furnishes the 16′ tones of Octave 1. This setup is common.

There is, however, a pedal sustain system which, as usual, complicates matters since it requires a latch arrangement. That is, if a pedal is pressed and released, the tone must still come through to the flipflops at least until the sustain gating has allowed final output to fade away. But if another pedal is pressed before the first tone fades entirely, the first tone must immediately be disconnected from the flipflops and the second substituted. This is the same problem which occurs in the Thomas and Lowrey organs described earlier and in many others.

Wurlitzer solves the problem with neon lamps. The initial basis of the idea is the old trick diagrammed in Fig. 14-11. Any neon lamp requires a certain minimum *striking voltage* to ignite it, and after it is lighted the voltage may be reduced to a lower value, the *sustaining voltage,* at which it will remain lit and conducting. The sustaining voltage is not enough to ignite a lamp which is initially off; it is only enough to keep it on if it has ionized in the first place. As we mentioned on page 49, neon lamps are now available with quite accurately held characteristics.

In Fig. 14-11 let us assume accurately made lamps which require 110 volts to strike but will maintain at a potential as low as 60 volts. We close S1. Since N1 has not ionized and is effectively an open circuit, there is no voltage drop across R and the full 115 volts is applied to the lamp, which thereupon ignites. As soon as it ignites, current is drawn through R, resulting in a voltage drop which leaves the lower end of R only perhaps 75 volts above ground. If we now also close S2, N2 will not light, since the 75 volts is not enough to strike it. But N1 remains lit because the 75 volts is enough to sustain it. We have here a *lockout* circuit, in which ignition of one lamp makes it impossible to ignite any other —and as many lamps as you like can be added to the setup.

Let us now progress to a combination *latching* and *transfer* circuit, one which will cause any lamp to stay on when its switch is closed momentarily, at the same time turning off any other lamp which may already be on. Gaze at Fig. 14-12. Note the minimum ignition and maintaining voltages of the lamps. Assume, as is the case, that if the 170 volts is applied through a 270,000-ohm resistor to any one lamp alone, igniting it, the lamp current is just sufficient to cause a drop of 100 volts across the resistor, reducing the voltage at the lamp to 70 volts, which is enough to maintain it but not enough to strike any other, unlit lamp.

When the device is first turned on, with all switches open, 170 volts is applied through R2 and all diodes to all lamps. Some one lamp, N3 we shall assume, is

slightly more sensitive than the others and strikes. The current drawn by N3 through R2 reduces the potential at point B to 70 volts, enough to maintain N3 but not enough to allow any other lamps to strike. The +70 volts from point B does, of course, reach all lamps, as the diodes are in the right polarity. Since point B is at +70 volts and point A is at the original +170 (all switches are open and no current is being drawn through R1 to cause a voltage drop), the top plate of capacitor C is, as shown, positive with respect to its lower plate.

Now let us close S2. This applies 170 volts from point A directly across N2, which promptly ignites. Notice that the 170 volts applied to N2 does not reach point B and affect any other lamps because diode D2 does not allow it.

As soon as N2 strikes, it draws current through R1, resulting in 100 volts drop across R1, immediately reducing the voltage at point A to 70—still enough to keep N2 lit.

Point A has suddenly assumed the same potential as point B. This means that the voltage difference across C no longer exists. The excess electrons which were on the lower plate of C must go back to the upper plate. They cannot pass through any diodes, which are in the wrong polarity for this. They must there-

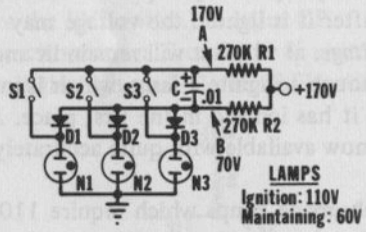


Fig. 14-12—Basis of Wurlitzer latching-type pedal transfer keying.

fore pass through R2 from left to right. In so doing, they create a pulse of voltage drop across R2 which is negative at point B with respect to the 170-volt supply. This pulse subtracts from the +70 volts at point B, bringing the potential at B momentarily below 60 volts. The pulse, and the reduction of voltage at B, lasts only a very short time—long enough for C to discharge—but it is quite long enough to rob N3, the lamp originally lit, of its maintaining voltage, and N3 goes out. We thus have N2 on (switch S2 is still closed) and N3 (and, of course, all other lamps) off. In other words, closing S2 has *transferred* the illumination from N3 to N2.

Now we open S2. By this time, C has long since discharged and the 70 volts at B has resumed. It maintains the illumination of N2 and will continue to do so until another switch is closed. Obviously, as many lamps as we like, each with its switch and diode, can be added to this lineup. And any time we close one switch, its lamp will go on and any one of the others which was on will go off. Since the selected lamp remains on even when we allow the switch to open, this is also a *latching* circuit. In fact, assuming that the conduction of the lamps can be used for some switching purpose, it is the electronic equivalent of the familiar group of mechanical pushbutton switches wherein pushing one releases all others while the pushed one remains in.

From here it is but a step to deciding on a quantity of 13 lamps, one for each pedal, and employing the current flow through the selected lamp to open a diode gate and pass the associated tone to the pedal flipflops.

Pedal Gate Circuits

Fig. 14-13 shows three of the 13 actual pedal gate circuits used in the organ, based on the idea of Fig. 14-12. The terminals at bottom indicate input points for signals from the C, D, and E generators. Since all the 13 circuits are alike, you will see at once what the entire lineup would look like. These signals are taken from Octave 3 of each generator (and the top C from Octave 4) through a 3900-ohm resistor each. The input terminals thus carry a positive potential.

When the organ is first turned on, one neon lamp, and only one, goes on. Let us assume that this is the C lamp, N1. The other lamps are off and thus are open circuits. The first negative-going signal alternation applied by the D generator to C2 has no effect on C2. For the capacitor to charge or pass signal, electrons would have to go from its top plate through D5 (and the 22,000-ohm resistor)

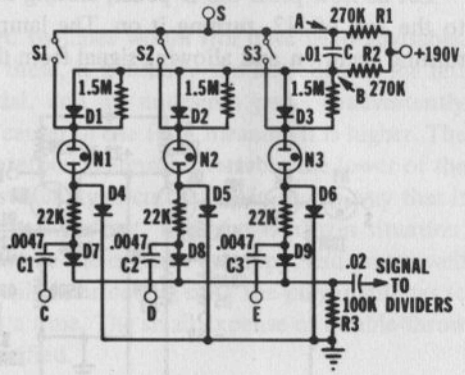


Fig. 14-13—Three keyers of the actual pedal transfer and latching circuit.

and through D8 (and R3) to ground. This they cannot do, as the polarization of the diodes will not allow it. The first positive-going alternation, however, which applies positive voltage to the bottom capacitor plate, does cause C2 to charge. Electrons are drawn from the bottom plate by the power supply and pass upward through D5 and D8 to the top plate of C2. When the next signal alternation places negative-going potential again on the bottom plate, the excess electrons now on the upper plate are unable to return to the lower plate through the diodes, and the capacitor thus remains charged. After a few cycles, the charge on C2 builds up to its maximum, which is about −5.5 volts. This negative potential on the top plate of C2 is applied to the anodes of D5 and D8, keeping them cut off. Signal from the D generator thus cannot pass through D8 to the output load resistor R3 and the frequency dividers. Each of the 12 keyer circuits whose neon lamp is out acts this way, so that 12 of the 13 generator signals are blocked from the output. This blocking idea is very similar to Baldwin's arrangement for sustain keying shown in Fig. 9-12, page 170.

We have assumed that the C neon is the one which initially went on and

stayed on. That being so, N1 conducts, its maintaining current passing from the supply, through R2, D1, N1, and D4. Like any semiconductor diode, D4 is not a perfect short circuit when on (when fully conducting such a diode is usually the equivalent of about 10 ohms), so there is a small positive potential on its anode with respect to ground. This potential, transferred to the anode of D7 through the 22,000-ohm resistor, is enough to turn on D7. With D7 conducting, the generator signal voltage applied to C1 charges and discharges C1 with each alternation through D7 and R3, so that the C signal appears across R3 and goes to the frequency dividers.

With all switches open, the actual voltage at point A is 170, there being a 20-volt drop across R1 due to some external circuitry attached to terminal S. The voltage at point B with no switches closed is +70, due to the drop across R2 caused by current through the one lamp which is on. The 1.5-megohm resistors across the D1, D2, and D3 diodes have no substantive effect on circuit operation; they are there just to provide a d.c. path for correct diode operation when the associated switch is open and the diode nonconducting.

Let us now press the D pedal, closing S2. The 170 volts at point A is applied to the top of N2, turning it on. The lamp current passes through D5 and D8, turning them on and allowing signal from the D generator to pass through D8 to

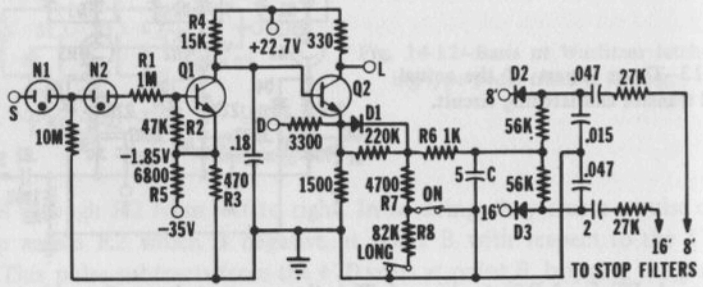


Fig. 14-14—The sustain circuitry for the pedals, controlling outputs of the 8′ and 16′ pedal frequency dividers.

the output. At the instant S2 closed, when N2 current passing through R1 pulled the voltage at A down to about the same value as at B, capacitor C was forced to discharge, which it did by passing excess electrons from its lower plate to its upper through R2. These electrons caused a pulse of reverse voltage across R2, momentarily lowering the potential at point B below the maintaining voltage of the lamps, and extinguishing N1, which, you will recall, was on to begin with. The *transfer* action of the circuit thus activated the D keyer in substitution for the C.

When the D pedal is released, now, and S2 opens, N2 remains alight—and D signal continues to go to the output—because it is maintained by the potential applied to it through D2 from point B, which returned to normal following the pulse of discharge of C. This is the *latching* action. (When the switch opens and point A returns to +170 volts, C, of course, recharges, this time briefly *adding to* the potential at B. However, the addition is not enough to reach the striking

voltage of lamps which are out, so its existence can simply be ignored.)

One more feature is apparent. One-note-at-a-time (monophonic) pedal systems must always have protection against the possibility of having more than one tone at a time fed to the flipflops. Usually this is provided by double-throw switches connected in a series transfer circuit. That could have been done here, but it is not necessary. It would be extremely rare for two switches to close at precisely the same instant, even if two pedals were pressed. The first switch to close will always light its lamp, even if the time difference is only microseconds; and since that will reduce the potential at A to below lamp striking voltage, it will not be enough to light the lamp of the other switch closed. Even if the two switches should somehow close at exactly the same instant, one lamp would almost certainly light first, locking out the other, because of minute differences in striking potential characteristics. It is, of course, theoretically possible for two lamps to light. However, in such a case the resulting voltage drop across R1 would reduce the potential at A to less than the maintaining voltage and both (if identical) or one of them (the one with the higher minimum maintaining voltage) would go out.

Despite all this theory, interlocked switches would *still* have been advisable. In actually playing short pedals like these, it is usually the left edge of the left foot which presses the desired pedal, and an undesired pedal inadvertently pushed at the same time is under the center of the foot, meaning it is higher. The pedal the organist means to play, therefore, is almost invariably the lower of the two, and almost all organs have the switching circuit set up in such a way that it is the lowest pedal pressed which will take effect. With the Wurlitzer situation, it might well be either, which would make the organist unhappy, and pretty well destroys the advantages of the electronic interlocking since the player still has to be careful to push only one pedal at a time. The small expense of double-throw switches would, I think, have been justified.

The keyer output signal goes to a waveshaper and two identical and conventional flipflop dividers, which we shall not show. Output of the first divider is 8′ pedal tone, and 16′ tone comes from the second.

Pedal Sustain

The pedal sustain circuitry operates, as usual, at the outputs of the two frequency dividers, before tone goes into the pedal stop filters. It is diagrammed in Fig. 14-14.

Terminal S is connected to terminal S in Fig. 14-13. Whenever no pedal switch is closed, the +170 volts appearing at point A in Fig. 14-13 is applied to neon lamps N1 and N2, Fig. 14-14, in series with R1, R2, and R3. These two lamps have different characteristics from those in Fig. 14-13; they will strike at 60 volts each and require 44 volts to stay alight once ignited. The 170 volts applied across the two of them in series is considerably more than the total of 120 the two in series require to ignite, and once ignited the voltage remaining across them after the voltage drop in R1, R2, and R3 combined is enough to keep them lit. As a result of the current through the series lamp circuit, the base

of the transistor is positive enough with respect to ground to be turned fully on, and because of transistor collector current there is a large drop across the collector resistor so that collector voltage is very low. At this time the −35-volt supply, voltage-divided by R5 and R3, results in 1.85 volts at the emitter, less than it would by simple voltage-divider action, because some of it is opposed by the positive voltage at the emitter resulting from current coming through the neons. This negative voltage has no effect, being swamped by the action of the neon current.

Since transistor Q1 is drawing considerable current, its collector is only slightly positive. The collector is connected directly to the Q2 base and consequently, Q2 being an emitter follower, its emitter is only slightly positive.

The circuitry at the right begins with the 8′ and 16′ terminals, which are the outputs taken directly from the transistor collectors of the two divider flipflops. The cathodes of D2 and D3 are therefore distinctly positive due to divider B+. Since the anode of D1 is less positive than the cathodes of D2 and D3, all the diodes are cut off and the potential applied across capacitor C is zero. And since D2 and D3 are cut off, no flipflop output can pass through them to the stop filters.

When a pedal switch is closed and held, Fig. 14-13, the voltage at terminal S immediately descends from 170 to 70. Since N1 and N2 in Fig. 14-14 require at least 44 volts each, or a total of 88 for the series pair, to remain ignited, they both go out and stay out as long as a pedal is pressed. In going out, they remove the positive on bias from the base of Q1, cutting off Q1, though a slight current still flows through Q1 because the base and emitter are now biased slightly negative (−2.5 volts). The voltage at the Q1 collector is now quite positive, since there is very little current through and voltage drop across R4, and this positive voltage is applied to the Q2 base. The negative voltage is applied to Q1 and the transistor is allowed to conduct a little so that the collector voltage will not reach full B-voltage. If it did, the base of Q2 would be more positive than its collector, which would give Q2 a fast stomache ache and might be fatal. (The 330 ohms in the Q2 collector is to supply a negative voltage for the Ssh-Boom, described later. Terminals D and L are also for Ssh-Boom.)

Since the Q2 base is now positive, so is its emitter and the anode of D1. The anode of D1 is much more positive than the cathodes of D2 and D3, so that all three diodes conduct, and +14 volts is applied across C. Since it is applied through only a small resistance—1,000 ohms, D1, and the Q2 emitter impedance —C charges quickly, fully opening diodes D2 and D3 and allowing signals to go to the stop filters.

Now the pedal is released. Point A in Fig. 14-13 returns to +170 volts. N1 and N2, Fig. 14-14, reignite. Q1 once again conducts, reducing its collector voltage so that insufficient positive appears at the D1 anode to keep it on and D1 opens. (We assume at this time that the Long and On switches are closed as shown.) As soon as the positive charge on C dissipates through R6 and R7, which are rather small resistors and therefore introduce little delay, D2 and D3 open and signal disappears. This, then, is the normal, nonsustain keying.

If the On switch is now opened, the discharge of C when a pedal is released must take place through R8 in addition to R6 and R7, and this considerably extends the time required for the tone to fade away, creating a sustain effect. And if the Long switch is also opened, the discharge path for C includes the 220,000-ohm resistor across D1, which is much larger still and affords a long sustain.

Pedal Voicing

The pedal voicing filters are shown in Fig. 14-15, all of them feeding a single-transistor voltage amplifier whose output goes to the organ amplifier.

There is a single flute filter, consisting of a tuned cutoff circuit. The Flute 8′ is fed to it directly (through attenuator resistance). The Bourdon 16′ is also fed right to the flute filter after an attenuator resistor which is bypassed by .022 μf

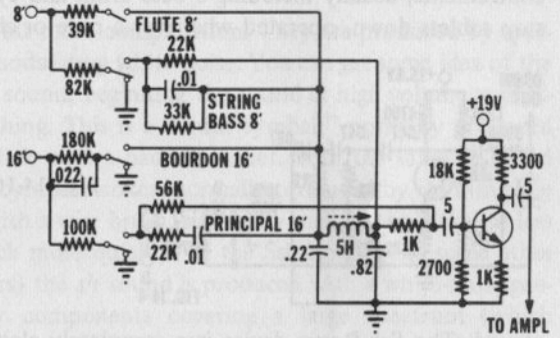


Fig. 14-15—Pedal voicing uses a single cutoff-type flute filter.

to give some emphasis to the higher frequencies. The Principal 16′ also goes through the flute filter, but it also goes, in lesser amount, around the flute filter so that higher harmonics are not eliminated entirely. The String Bass 8′ is connected directly to the amplifier through its 22,000-ohm resistor, and also through the flute filter after a large resistor having a high-frequency bypass.

Sound Production

The organ contains a single power amplifier which has two voltage amplifiers, a "pre-driver" stage, and a driver transformer-coupled to the power stage. The latter is direct-coupled to the speakers.

The console contains five speakers. Two are 12-inchers and are fed at all times (except when a phone jack for silent playing cuts out all speakers) right from the amplifier. One is a 4-inch tweeter, normally paralleled, through a 5-μf capacitor, with the bass speakers.

In addition, two rotating 4-inch speakers are provided, forming what is called Spectra-Tone. High frequencies can be switched away from the fixed 4-inch unit and to the rotating ones to put the Spectra-Tone on, or fixed and rotating tweeters can be energized simultaneously, at the command of a tablet labeled Spectra-Tone Chorus. A third tablet controls the speed of speaker rotation. The fast vibrato speed is labeled Theatre and the slower speed Cathedral. (If your church or home is smaller than a cathedral you're just out of luck.)

The swell shoe operates a 10,000-ohm potentiometer placed between the first voltage amplifier and the second. Manual tones go into the first amplifier, while pedal tones join at the potentiometer. The second stage also feeds a single stage which energizes the input transducer of a spring-type reverberator; the output transducer, through two more stages, joins reverberations to main signals at the input of the pre-driver stage. A switch and variable resistor in series with reverberation output supply on-off control and control of the amount of reverb.

The organ power supply, which also operates the amplifier, is conventional and unregulated.

Ssh-Boom

The old-time theatre organ was equipped with many odd and interesting accoutrements, usually including a bass drum and cymbal which, with the proper stop tablets down, operated whenever a note or chord of manuals or pedals was

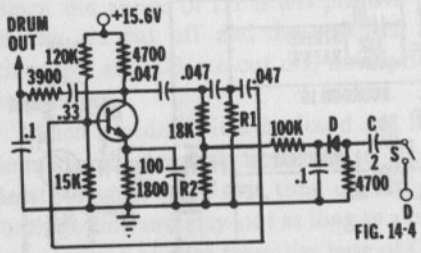


Fig. 14-16—This is a bass drum circuit.

played. The Ssh-Boom device is a completely electronic system of reproducing bass drum beats and cymbal crashes when pedals are pressed (along with the regular pedal notes) and sizzle cymbal sounds when manuals are played (again along with the regular notes).

Fig. 14-16 is a diagram of the bass drum section of the Ssh-Boom. It is basically simply an R-C phase-shift oscillator whose feedback is insufficient for oscillation.

Going back for a moment to Fig. 14-14, you will recall that when a pedal key is pressed a positive voltage appears at the emitter of Q2, which turns on gate

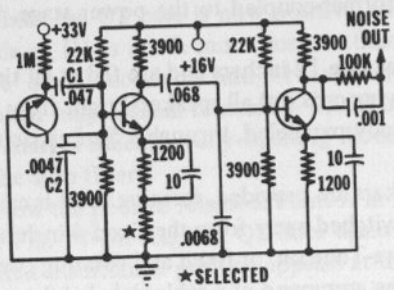


Fig. 14-17—Noise generator and cascaded amplifiers.

diodes D2 and D3 to pass pedal tones to the pedal stop filters. This same positive voltage appears at terminal D in Fig. 14-14 and at terminal D in Fig. 14-16. It is applied through diode D and capacitor C to the top of R2, where, because of C,

it appears as a positive pulse rather than a steady voltage. This pulse is enough to shock-excite the oscillator, which thereupon emits a single damped-wave train at its "resonant" frequency.

The frequency of the oscillator is fairly low, as controlled by the series capacitors (.047 μf) and shunt resistances, the latter including R1 and R2, values for which are factory selected for the exact desired frequency. R2 is also selected for the correct amount of shock to be provided by the input pulse so that the duration of the damped-wave train as well as its frequency are correct for simulating a bass drum thump. The diode is provided so that when a pedal is released and the voltage at terminal D goes negative, the negative pulse is not transmitted as a new shock to the oscillator.

Output is taken from the collector, and where it goes we shall see in due course.

Cymbal sounds require a bit more complication. They are produced by spectrum-shaping and envelope-modulating white noise. You can get some idea of the effect if you will make a *sh* sound, beginning the sound at high volume and letting it gradually fade to nothing. This is a "crash cymbal," normally produced by hitting the cymbal one lick with a padded mallet. With the same *sh* sound you can approximate brush cymbal strokes, normally produced by rhythmically tapping the cymbal lightly with a wire brush, simply by making the initial *sh* less loud and fading it away much more quickly. In the Ssh-Boom (and some other rhythm instrument simulators) the *sh* sound is produced with a white-noise generator containing frequency components covering a large spectrum (which sounds like a *ssss* hiss) and using a formant filter to get the exact kind of noise desired.

Fig. 14-17 shows the noise-producing section of the Ssh-Boom. Only the base-collector diode of the first transistor is used, and this diode is connected in reverse, with a series 1-megohm resistor. This is recognizable as a zener-diode arrangement, which it is, in effect, conduction occurring as the result of reverse breakdown and avalanche. Indeed, the semiconductor devices usually sold as noise generators or "noise diodes" are zeners, carefully calibrated to produce so much noise at so much voltage and at so many dollars apiece. A selected transistor is infinitely cheaper and, for this application, just as good.*

If you have used zener diodes only as voltage regulators in well bypassed d.c. circuits, it may come as a surprise that all zeners can generate noise. In this case, the noise appearing across the improvised zener diode is passed to a two-transistor amplifier. The unbypassed part of the emitter resistance of the first amplifier stage is factory selected for the desired output amplitude, which otherwise would require a control or one of those expensive calibrated noise diodes.

Fig. 14-18 shows the control circuits in which the noise is converted to cymbal brush strokes when keys are pressed or to crashes when pedals are used.

*Don't try this with an ordinary diode on the theory that Wurlitzer has wasted a perfectly good transistor when a cheap diode would do. Its reverse current will try to be infinite when inverse voltage rating is exceeded, and you will be out a diode. Also, see Fig. 14-4 for another use of a transistor as a zener diode.

Harking back to Fig. 14-8, notice that positive keying voltage for the diode manual keyers is placed on the rotatable busses of the upper manual through a single 4700-ohm resistor, followed by individual 2200-ohm resistors for each bus. Thus, when these busses, or any one of them, is on and a key is pressed, current is drawn through the 4700-ohm resistor to open the keyer diode(s), resulting in a drop in the voltage at the junction of the 4700- and 2200-ohm resistors. This change toward negative goes through the .047-μf capacitor to point Y as a negative pulse each time an upper-manual key is pressed. The same arrangement exists for the lower manual, providing a negative pulse at Z whenever a key is pressed. Points Y and Z are used in Fig. 14-18, where these pulses are transferred through tablet switches to the control circuitry when the organist desires the Ssh-Boom brush cymbal effect when playing the keys of either manual.

Let us assume he closes the switch for the lower manual, so that the negative pulse at point Z goes through the switch to the base of Q1. It appears at the Q1

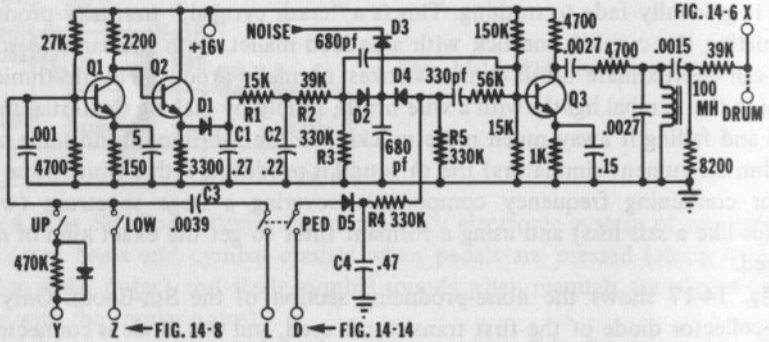


Fig. 14-18—This is the Ssh-Boom cymbal control circuitry for the manuals and pedals.

collector and Q2 base as a positive pulse; and it makes a positive pulse at the emitter of Q2, which is an emitter follower. The positive pulse passes through D1 and through envelope-shaping R's and C's (to slow the attack, which should not be instant) to the anode of D2.

D2 is connected in series with D3, cathode to anode, and the cathode of D3 goes to the output of the noise generating system in Fig. 14-17. This output comes from the collector of the last transistor without any d.c. blocking, making the cathode of D3 positive, and normally blocking both D3 and D2, Fig. 14-18. When the positive pulse, however, strikes the anode of D2, it overcomes this original diode off bias and causes both D2 and D3 to try to conduct. The noise signal then passes through D3, D2, and the 680-pf capacitor to the base of amplifier Q3.

As the negative input pulse from Z subsides, the base of Q1 goes back in a positive direction to reach its original quiescent point, causing a negative-going pulse at the Q2 emitter. Because of the polarity of D1, this cuts off D1. C1 and C2, which were quickly charged positive when the original pulse came along,

through the small resistance of the Q2 emitter and relatively small resistor R1, making for a fairly fast, but not instant, attack, must discharge before D2 and D3 can stop conducting and the signal can fade out. This discharge, since D1 is shut off, can only take place through the large R3 (as well as through the diodes and the 100,000-ohm output resistor in Fig. 14-17). This takes time, so the noise signal fades more slowly than it came on, though not very slowly.

Obviously, the same thing happens if the switch for the upper manual is closed and upper-manual keys are played, due to pulses from Y. So much for the manual Ssh-Boom action.

When the double-pole pedal Ssh-Boom switch is closed, the cymbal attack should occur, as with the manuals, and also the *sh* sound should continue as long as the pedal is held down and then fade slowly when the pedal is released.

The left section of the pedal switch makes the initial attack. When terminal L in Fig. 14-14 goes negative as a pedal is pressed (see explanation a few paragraphs back), this negative change goes through the left pedal switch, Fig. 14-18, and C3, producing the same action as when a manual key is pressed.

At the same time, positive signal, directly coupled without a capacitor, is taken from terminal D in the pedal sustain circuit of Fig. 14-14 and charges capacitor C4, Fig. 14-18. The positive charge, applied to diodes D5, D4, and D3 in series, causes all these diodes to conduct. The signal from the noise generator thus also passes through D3 and D4, then through the 330-pf capacitor and 56,000-ohm resistor to the base of amplifier Q3. Unlike the signal due to the pulse at L (or a manual-produced pulse at Y or Z) this signal keeps on coming as long as the pedal is down.

When the pedal is released, positive voltage no longer appears at terminal D, and the positive charge remaining on C4 blocks D5. C4 must now discharge through R4 and R5 and the two remaining diodes, quite a high-resistance path. This takes some time, so that the cymbal sound dies away very gradually, as a cymbal crash would.

The output of amplifier Q3 contains an L-C formant filter which, in conjunction with other R's and C's in the circuitry as a whole, shapes the total spectrum response as needed to produce what the designers like best as a cymbal sound. The output, terminal X, goes to the input of the nonpercussible-stop voltage amplifier, Fig. 14-6, and thence to the organ amplifier. An additional tablet does some simple impedance switching, both at the point where the drum signal of Fig. 14-16 joins the cymbal signal, Fig. 14-18, and in the cymbal output itself, to provide for greater than normal Ssh-Boom volume when desired. Another switch shorts cymbal and drum signals to ground when Ssh-Boom is to be turned off.

Percussion

One more effect in this organ is a percussion which is quite separate from the sustain and is an actual percussive effect. It is not used, however, for general percussive stops but only for very short-decay sounds typified by one of the tablet titles: Pizzicato Percussion.

Fig. 14-9, the voicing for the upper-manual percussible stops, shows a connection from its voltage amplifier output to the organ amplifier through a 10-μf capacitor connected to terminals A and B. When the percussion is turned off, these 8′ stops do indeed go through the capacitor to the amplifier. When percussion is turned on, however, signals from point A go instead to terminal A in the right section of Fig. 14-19, thence through an amplifier stage and the photocell section P of an LDR to terminal B and the organ amplifier. The resistance of the photocell controls the amplitude of transmission through the system.

The cell P in Fig. 14-19 is in the same light-tight enclosure as lamp L in the Q3 circuit. This lamp is a neon. As normally biased, Q3 conducts very little, so that the difference between the supply voltage of 170, to which one end of the lamp is connected, and the voltage at the Q3 collector, to which the other end connects through a 22,000-ohm resistor, is insufficient to light the neon lamp. However, when a positive pulse is applied to the Q3 base, the lamp blinks. It is characteristic of the CdS cells in most LDR's that when illuminated by a reasonable level of light the cell resistance is reduced very quickly; and when the light is removed it takes a much longer time for the cell resistance to return to its original, unilluminated, high value. Thus, each time the neon lamp blinks briefly, the cell P goes to low resistance, almost immediately passing signal from point A to output B at maximum design volume. Then, at a much slower rate, the cell resistance rises again until it is so high that substantially no signal passes through the system. This obviously produces a percussive amplitude envelope each time the lamp is made to blink. And if the lamp is caused to blink repeatedly at a fixed rate of speed, the sound will be a repeat percussion. For this repeat effect,

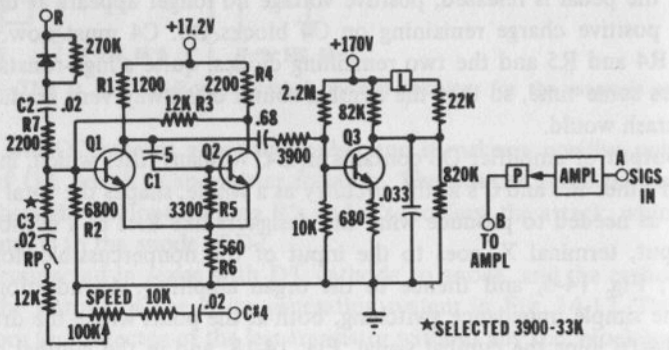


Fig. 14-19—Percussion circuit. Modulator is at right and pulse-producing system at left.

the decay time, though longer than the attack time, must be quite short. This is arranged both by selecting the photocell characteristic and by making it, as far as signal is concerned, the series leg of a voltage divider whose shunt leg is quite small. The decay time could be adjusted, but in this organ it is not; it always is as short as is needed for the repeat.

In a manner which we shall discuss in a moment, the Repeat Percussion tablet (a) causes the upper-manual regular percussible stops to go through the photocell

system, and (b) causes the neon lamp to blink on and off several times per second. When the Repeat tablet is off, these upper-manual regular stops go through right to the organ amplifier as in Fig. 14-9. Percussion affects them only when the repeat effect is used.

It is also possible to make the lamp blink once each time a key is pressed. This is done when the Pizzicato Percussion tablet is on. In addition, this tablet causes the same nonpercussible square wave 8′ and 4′ tones which go to the nonpercussible stops of Fig. 14-6 to pass also through the photocell-amplifier arrangement, which includes a filter which makes the short-percussion effect on these tones sound more or less like plucked fiddle strings. And finally, another tablet, labeled Bongo Percussion, passes 16′ tones from the same source through a suitable low-pass filter and the photocell system, so that when each key of the upper manual is played, a brief *bong* sounds at the 16′ pitch of the note. The bongo and pizzicato percussions are stops in their own right; that is, they do not "percuss" stops which ordinarily are regular organ voices. The repeat percussion, on the other hand, transforms most of the regular 8′ upper-manual voices to strumming sounds.

The important technical point of this system is the method of supplying positive pulses to the base of Q3, Fig. 14-19, either once each time a key is pressed or repeatedly at a controlled rate, so as to make the neon lamp blink once per key strike or repeatedly. For the repeat effect, the same problem exists as in the reiteration scheme of Schober, page 202. That is, you cannot simply have a multivibrator furnishing pulses. You must also make sure that at the instant a key is pressed the first pulse is produced, even though if left to itself the pulsing system might be in the off condition at that instant. The solution here is different from Schober's but it is equally interesting.*

Transistors Q1 and Q2 in Fig. 14-19 form a Schmidt trigger. In analyzing it, do not look at any components except the ones we shall mention. Since R6 is not part of the classic Schmidt trigger, forget about it, too, and assume R5 and both emitters are grounded.

When no external signals are applied, Q2 is nonconducting, since its base has no source of on bias and is maintained practically at ground potential by R5. Q1 is fully conducting, however, because positive bias is applied to its base through R4 and R3 as the series leg of a voltage divider from the power supply, with R2 the shunt leg. The collector of Q1 is therefore nearly at ground potential, so that practically no potential difference exists across C1 and C1 has almost no charge.

If an external negative pulse is now applied to the Q1 base, Q1's conduction is reduced and its collector voltage rises. This positive-going voltage is applied through C1 to the Q2 base, turning on Q2. The resulting negative-going potential at the Q2 collector is applied through R3 to the Q1 base, reinforcing the original applied negative pulse. The action snowballs, just as in a flipflop or multivibrat-

*It is also of some interest to note that not all makers see fit to attack the problem at all—Lowrey and Thomas, for example. Having been concerned with the design of a repeat percussion on two occasions, I think the point has to be considered unless you feel sure the organ buyer will never play anything fast.

or, and very quickly the original conditions are reversed; Q1 is fully off and Q2 is fully on. After the immediate change of state, C1 continues to charge, electrons from its upper plate flowing through the power supply to ground and up through R5 to its lower plate. This flow causes a voltage drop across R5 which maintains the positive voltage on the Q2 base which keeps Q2 conducting.

As C1 nears the end of its charge, charging current decreases and the voltage drop across R5 becomes less than that required to keep Q2 fully conducting. As its conduction begins to drop, Q2's collector begins to go positive. The positive-going potential is applied to the base of Q1 through R3, turning Q1 on, so that Q1's collector begins to go negative. The negative-going voltage goes to the base of Q2 through C1, and again the action snowballs so that very quickly the original conditions are reached again—Q1 on and Q2 off. The time during which the circuit remains in its unstable state is thus determined by the time required for C1 to charge. This is a simple time constant comprising C1, R1, and the net impedance between the Q2 base and ground.

When the circuit flips back to its original stable state, C1 is still charged, its bottom plate negative. The entire circuit does not, therefore, return to its original condition until C1 has had time to discharge (through the same time-constant circuit). Until C1 has discharged sufficiently, a new negative trigger pulse applied to Q1 and the resulting positive pulse sent to Q2 will not have any effect.

Summing up, we have a circuit which will produce a positive pulse at the Q1 collector when a negative pulse is applied to the Q1 base, but will not thereafter produce another positive output pulse in response to a negative input until after a certain time has passed. As you will have guessed, it is this positive output pulse which is applied to the Q3 circuit and causes the neon lamp to blink and produce a percussive note.

In the single-strike operation mode, the repeat switch (RP in the diagram) is open as shown. Terminal R is connected to point R in Fig. 14-8. When an upper-manual key is pressed, the voltage at R (Fig. 14-8) goes slightly negative. The negative step applied to R in Fig. 14-19 goes through the diode, C2, and R7 as a negative pulse to Q1. By the method we have described, this causes a single-note percussive strike.

When the repeat switch is closed, 277-cps square-wave tone from the C# tone generator is applied through the speed-control potentiometer and C3 (among other components) to Q1. C3 and the Q1 base impedance form a differentiator so that the square-wave signal becomes a series of pulses, both negative and positive, the pulse height being dependent on the setting of the speed control. Positive pulses have no effect on the circuit and can thus be ignored.

Assuming that the trigger circuit is in its resting, stable state, let us press and hold an upper-manual key. If one of the negative pulses from the C# generator happens to be present at that instant, it will trigger the trigger (the trigger the trigger). If not, the keying pulse at terminal R will do the job, so that in any event, the cycle of events always starts at exactly the instant the key is pressed.

After the trigger puts out its initial pulse and the neon lamp blinks and a first percussive tone strikes, nothing further can happen, even though the C# pulses

continue to parade along at the rate of one every 3.7 milliseconds, until C1 has discharged. How long we must wait until one of these pulses causes another trigger action depends solely on the height of the pulses, as adjusted by the speed control. If the pulse height is small, C1 must be practically fully discharged before a pulse can have effect and initiate a new cycle of events. The period between percussive tone repetitions is therefore long. If pulse height is greater, C1 need not discharge so fully before one of the pulses can initiate a trigger, so repetition rate is higher. The action continues as long as a key is held down—a constant series of triggers followed by waiting periods, resulting in a repeat percussion effect.

Chapter 15

Seeburg Organs

The history of Seeburg's participation in the organ industry is short, but that of its organ manufacturers much longer. Something like a dozen years ago, a "development group" was formed in Brattleboro, Vermont, consisting of former employees of the now defunct Minshall Organ organization, plus me. This group developed the first practical neon-lamp organ and sold it to a corporation formed for the purpose, headed by Earle V. Kinsman of Laconia, N. H., and including members of the group. The company manufactured various models of the organ under the Kinsman name. The neon generators proved entirely practical (the original idea was contained in an RCA patent—see page 51) and indeed the Schober Spinet Organ still uses them because of their economy.

After some years, certain corporate problems remained unresolved and the operation was sold to Seeburg, known for many years as the juke box giant. After some development, the neon organs were replaced with transistor models.

In 1966 Seeburg acquired Gulbransen, an old established name in the organ field by the standards of this comparatively young industry, and all organ manufacture is now concentrated in the Gulbransen facilities in a Chicago suburb. The Gulbransen designs and name have been retained, and probably the Seeburg name will shortly disappear entirely from organs, remaining only on the Gulbransen letterhead and perhaps on the Select-A-Rhythm. This chapter on one of the Seeburg organs, however, is worthwhile because the instrument does contain circuitry and gadgetry of interest to those who want to learn about electronic organs. Of special and continuing interest is the Select-A-Rhythm, a unique automatic rhythm device described in some detail a little later.

The Seeburg Carlyle Model S3 is a spinet-style instrument which also comes in the Deluxe version pictured in Fig. 15-1. Both versions are the same, except that the Deluxe has a two-channel amplifier with provisions for feeding either manual to the Leslie speaker and the other to the main, a chime stop, and the Select-A-Rhythm. The organ has nine upper-manual voices at 16′, 8′, and 4′ pitches, four lower-manual stops at 8′ pitch only, and three pedal stops at 16′ and 8′ registers. Upper-manual and pedal voices can be sustained. The remaining controls and functions will become apparent as we proceed.

Tone Generators

The main tone generators of the Carlyle provide 56 tones. The highest is the C at 2093 cps, which is the first note of Octave 7, Table 1-1; the generators are arranged with C as the highest note of the 12 rather than the lowest, which is the C#. The F through B generators have master oscillators running in Octave 6, and

have four dividers each. The C# through E generators also have their master oscillators in Octave 6, but they have only three dividers each. Since the keyboards are the standard 44-note F through C manuals common to spinet organs, the lowest 8′ C through E are not needed. The lowest unneeded C is eliminated by making the C generator the highest-pitched of the group; and the unneeded C# through E are left out by having one less frequency divider on each than for the

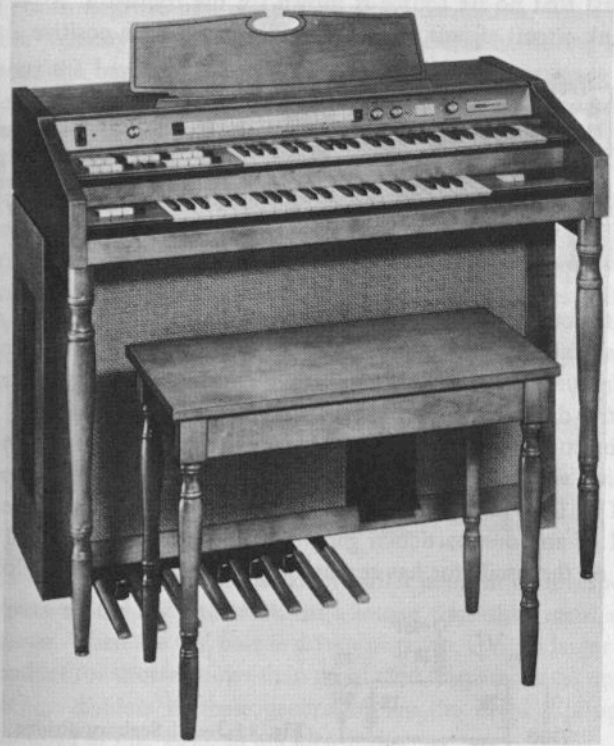


Fig. 15-1—This is the Seeburg Carlyle Deluxe organ which we shall describe in this chapter.

other notes. There are thus four each C#, D, D#, and E, and five each of the remaining notes, a total of 56. The lowest pitch is the 87.3-cps (Octave 2) F.

Fig. 15-2 is a schematic of one of the 12 master oscillators. The oscillator itself is transistor QO, in a grounded-collector Hartley circuit. The tank circuit is the coil and capacitor CT in parallel with it. The starred resistor, usually 470 or 680 ohms, is chosen to reduce the Q of the coil somewhat so that when vibrato is applied all oscillators will deviate by about the same percentage. QS is a simple overdriven amplifier which acts as a squarer to provide both tone output at the master oscillator frequency and trigger signal for the first frequency divider.

The vibrato system used in this organ is unusual. Basically, it employs diodes to add or subtract from the tank circuit the effects of an additional capacitor, an idea also used by Lowrey and Gulbransen. It is done in quite a different and more subtle way, however, though I have no reason to believe it is better.

Transistor QV in Fig. 15-2 is not part of one of the 12 generators; there is only one in the organ, and the D1 diode of each generator is connected to its collector. The base of QV is supplied sine-wave vibrato-frequency signal which originates at a multivibrator and is passed through a filter to remove harmonics, in a circuit which has the usual facilities for controlling depth and rate and is not of special interest.

Notice first that no B+ source is shown for the transistor. B+ is supplied by oscillator tank circuit signals of all 12 oscillators. At each positive alternation of

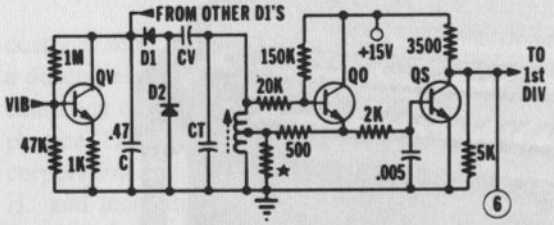


Fig. 15-2—A master oscillator QO and the output stage QV of the vibrato generator. QS is a squarer stage.

each oscillator current passes through that oscillator's D1 to charge capacitor C. Since the oscillator signals are not phase-locked in any way and are at comparatively high frequencies, this serves to keep a constant charge on C and provide a steady positive d.c. for the QV collector and for its base bias.

When vibrato is off and no a.c. signal is applied to the QV base, there is thus a fixed positive voltage at the collector. This positive potential appears on the cathodes of all D1's as a certain amount of reverse bias. The reverse bias is such that the D1 of any one particular generator does not conduct until the positive alternation of the oscillator has reached a point near, but still below, the peak

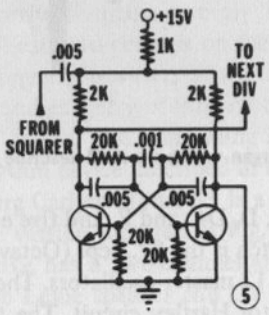


Fig. 15-3—All Seeburg divider flipflops are identical to this.

value. When that point is reached, that oscillator's D1 conducts, and continues to conduct until oscillator voltage reaches the positive peak and then recedes again below the D1 conduction value.

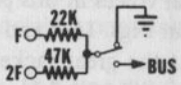
During the time an oscillator's D1 conducts two things happen. First, the oscillator is adding its energy to that from the other 11 oscillators to keep C charged and furnish B+ for QV. Second, the conduction of the D1 connects the left end of that oscillator's CV to the top of C. Since C is so much larger than any of the CV's (which range from .0022 to .0047 μf) this effectively means that the left end of CV is grounded and thus CV is shunted directly across the tank circuit.

The positive potential at the collector of QV is actually placed across both diodes D1 and D2 in series to ground, so that as an off bias it is equally effective for both. Thus, when the oscillator goes into a negative alternation, D2 conducts only after its off bias is overcome, meaning during a time near the peak of the negative cycle. And when D2 conducts, CV is grounded through it and again placed across the oscillator tank. In sum, then, CV becomes part of the tank circuit during both negative and positive peaks of oscillation. While the instantaneous angular velocity of the oscillator wave is different during peaks from what it is about the baseline, the result to the ear is a single average frequency, something between what it would be without any effect of CV and what would result if CV were across the tank all of the time.

It follows that the exact "average" frequency attained depends on the percentage of each cycle (or half-cycle) during which CV is active. It also follows that this percentage can be controlled by controlling the positive potential at the QV collector. And, getting finally to the point, if we vary the positive potential at the QV collector at a vibrato rate, all the oscillator frequencies will go up and down at the same rate.

This is exactly what happens when vibrato is turned on and sine-wave vibrato-frequency signal is fed to the QV base. When signal at the VIB input goes positive QV conducts more heavily and becomes effectively a lower resistance,

Fig. 15-4—Connections to each lower-manual key switch.



so that all the diodes begin to conduct at an earlier point in their respective half-cycles. This puts all the CV's to work for a longer time than usual and all frequencies go down. When the QV base is driven negative, QV is a larger resistance, the diodes conduct for shorter times than usual, and frequencies go up.

The frequency dividers in these generators are the usual flipflops. One of them is diagrammed in Fig. 15-3; all are identical.

Lower Manual

The lower manual is a simple direct keying system involving a single-pole double-throw switch for each key, as in Fig. 15-4. The normal 8′ tone from the appropriate generator output is connected to the F terminal and the generator an octave higher to the 2F terminal. The composite tone thus obtained on the switch arm is the simplest possible staircase with the correct amplitude relationship because of the resistor values. This setup suffices because the lower manual has only 8′ stops.

There are four collector busses for the outputs of the key switches, one for each of the top three octaves of the lower manual, and a fourth for the remaining eight notes from the low F to the first C. These four groups are kept separate so that there can be some scaling, since all the key switching resistors are the same.

Fig. 15-5 shows how the scaling is done. The four busses are fed to the four

inputs shown, No. 4 being the highest octave. Busses 1, 2, and 3 are each terminated to ground by a 12,000-ohm resistor (and 220 ohms, which is insignificant). The 1,000-ohm resistors are also part of the terminations. In addition, they attenuate the signals progressively so that bus-4 signals go to the voicing filters without attenuation, while signals from bus 3 are attenuated, those from

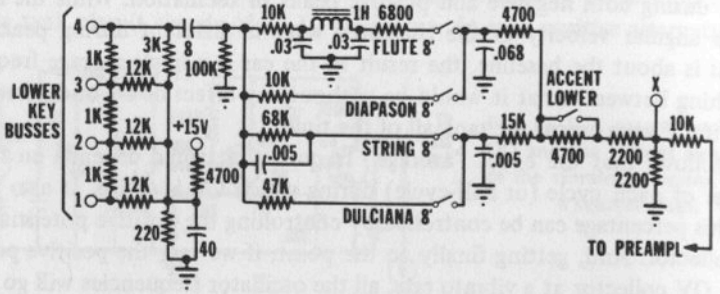


Fig. 15-5—The voicing of the lower manual includes stops at 8′ pitch register only.

2 are attenuated still more, and those from 1 the most. This follows the usual line of emphasizing the treble tones to offset the treble rolloff present in most filters (present for all voices in this particular organ).

You will notice in Figs. 15-2 and 15-3 that the generator outputs are not d.c.-blocked. To prevent the great racket this would cause in the key switching, the designer simply injects a 15-volt potential into the collector system of Fig. 15-5, so that at the point where all signals from the switches meet, a d.c. voltage appears which is the same as the voltage carried there from the generators. The idea is, of course, that as long as the d.c. potential at both sides of a switch is the same, it need not be zero to avoid transients. This dodge is cheaper than putting a blocking capacitor at each generator output or key-switch input. In addition, the d.c. at the generator outputs is needed for the diode keying system of the upper manual. Wurlitzer has this same problem, with both a direct and an electronic switching system, but solves it by providing two outputs for each generator stage, one with and one without a blocking capacitor.

Fig. 15-5 shows the voicing filters for the four lower-manual stops. The Flute is a cutoff filter, now not uncommon in organs but, I believe, first used by the chief designer of the original Kinsman organ, under whom the principal designer of this Seeburg organ worked. I disagreed with the idea then and I still do, because cutoff filters make a flute sound horny and because the relatively sharp cutoff makes the stop hard to scale, if not impossible. More elaborate cutoff filters with a number of sections, each taking care of a relatively small range of notes, do not suffer from the scaling problem at least.

The Diapason and Dulciana filters are not really just resistors; note the shunt capacitors following the switches. The Accent Lower switch, when closed, shorts a series resistor and increases lower-manual volume. Several Accent tablets appear on this organ. Output of this system goes through a selector switch to

one of the two preamplifiers, one of which feeds a power amplifier for the main speaker system and the other the Leslie.

Upper-Manual Keying

All keying for the upper manual is electronic, a combination transistor-diode system with an adjustable decay so that it can be used for sustain effects. Fig. 15-6 shows all the components required for a single key.

There is one transistor for each of the 44 keys, and a single switch under each key. In Fig. 15-6 the switch is shown in its off position, the key unplayed. The transistor is an emitter follower; the small bypassed collector resistor is, according to the service manual, for hash suppression, whatever that may mean. Let us assume for the moment that the Sustain Control terminal is grounded. Since there is no positive potential applied to the base and a definite negative voltage applied to the emitter, the transistor does not conduct.

Three appropriate tones from the generators are connected to the cathodes of the three gate diodes. Since there is no blocking capacitor at each generator output, the instantaneous values of the square waves applied to the diodes range from +1 to +9 volts. This keeps the cathodes always positive (and the transistor emitter is about 3 volts negative), so none of the diodes conduct and no tone is passed to the output.*

When the key is pressed and the switch arm moves down, +15 volts is applied to the transistor base through a time constant consisting of the 1,000-ohm

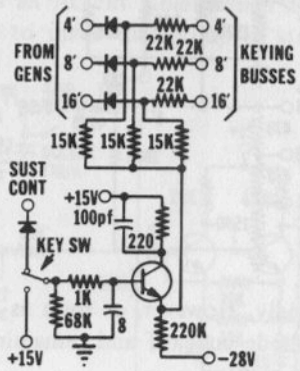


Fig. 15-6—Electronic keying components for one upper-manual key.

series resistor and the 8-μf capacitor. This causes a slight delay before the +15 volts appears at the emitter, tending to kill transients. The resulting positive voltage at the emitter is applied to the anodes of all three diodes, which then conduct and pass their tones to the three collector busses.

When the key is released, the 8-μf capacitor discharges almost immediately through the 1,000-ohm resistor and the diode in the sustain control leg (which

*Such a simple system is bound, of course, to have considerable leakage due to capacitance through the diode. If the organ is reasonably quiet with no keys played, it can only be due to sufficient treble rolloff in the voicing and other circuits. You will notice in Fig. 15-8 that no tones are allowed to go through without rolloff.

you will recall we assumed was grounded). This is normal, nonsustain keying.

When long sustain is wanted, the grounds are removed from the Sustain Control points of all 44 keyers and the points just left unconnected. Now when a key is released, the 8-μf capacitor must discharge through the 68,000-ohm resistor, which takes a comparatively long time, so that the decay of conduction in the three diodes is gradual and protracted.

For the medium sustain effect, +5 volts is applied to all the Sustain Control terminals. During the first period after the key is released, the charge on the capacitor is so high that the control diode conducts fully and the capacitor discharges rapidly. However, as soon as the charge decays to slightly less than 5 volts, the diode cuts off and remaining discharge must take place through the 68,000-ohm resistor.

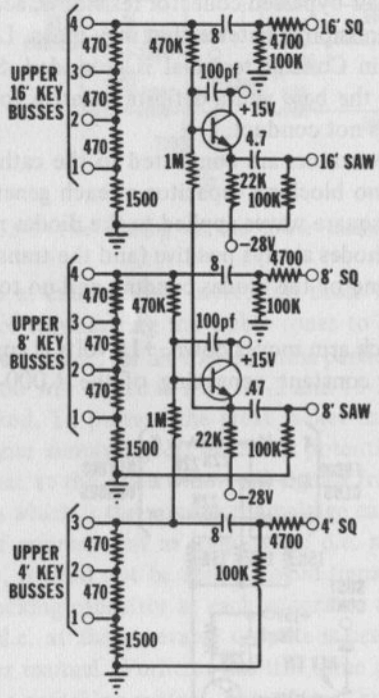


Fig. 15-7—Tone collection and staircasing for the upper manual.

As with the lower manual, there are actually four 8′ and four 16′ collector busses, each collecting tones for one octave, except the lowest which takes care of the lowest eight notes. There are only three 4′ busses because the top octave of keys repeats the same pitches for the 4′ register as the next to top, due to lack of Octave 7 in the generators.

Upper-Manual Voicing

Fig. 15-7 shows what happens to the tones picked up on the collector busses for the diode keying system. The signals from the 16′ busses are combined by a simple resistor network which leaves the upper octave of keyed tones at maximum loudness and each succeeding lower octave at lower volume. Then two outputs are derived. One consists of nothing more than the keyed tones. Since these are the original square waves from the generators, this output is labeled 16′ SQ. We will get to the second in a moment.

The 8′ tones are combined in the same way and fed, also through an 8-μf d.c. blocking capacitor, to an 8′ SQ output. And the three octaves of 4′ tones are treated in the same way.

Notice now that the 8′ combined signals are also fed to an emitter follower through 470,000 ohms. The output of the 4′ combined key busses also feeds 4′ square waves to the 8′ emitter follower. Since it does this through 1 megohm, the base of the 8′ transistor receives 4′ square waves at half the amplitude of the 8′. Its emitter output therefore shows an 8′ staircased tone, which we have labeled 8′ SAW, even though it is only one step removed from an 8′ square wave. This one step, however, is sufficient to destroy the symmetry of the waveform and remove much of the hollow sound.

In the same way, 16′ combined signal is also fed to an emitter follower whose base also picks up tone from the 8′ transistor output. The output of the 16′ stage, therefore, contains a 3-component staircase composed of 16′, 8′, and 4′ tones.

The actual voicing is shown in Fig. 15-8. The 16′, 8′, and 4′ square tones are used for the three tibia stops, all three using the same L-C cutoff filter. The remaining stop filters should require no explanation. For each of the three groups of stops, tibia, string, and reed, there is an Accent tablet which shorts out a series resistor when the organist wants to make that group of stops louder.

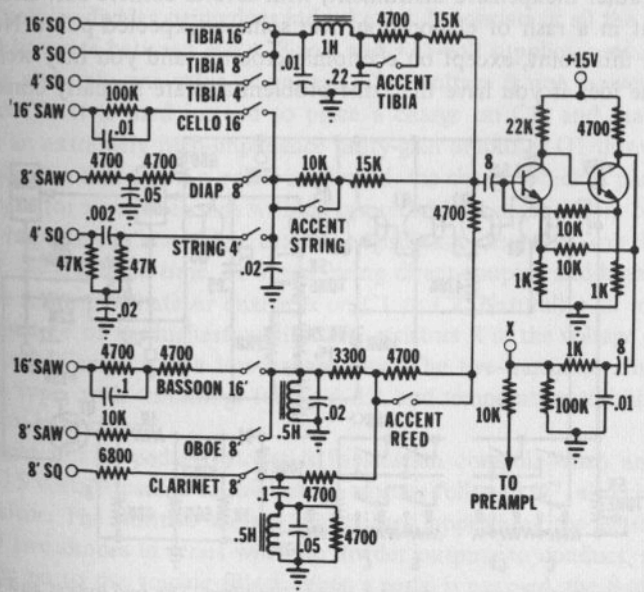


Fig. 15-8—These are the voicing networks for the stops of the Solo (upper) manual.

All stops are combined and fed through a two-stage voltage amplifier with direct coupling and both a.c. and d.c. feedback for stability. The amplifier output goes through a little more filtering to the preamplifier.

Points X in Figs. 15-8 and 15-5 are connected to the two poles of a switch, the arm of which goes through 2200 ohms to ground. The switch is controlled by a Solo tablet. With the tablet in Upper position, the lower X point is grounded through the 2200 ohms, reducing its volume—and *vice versa.*

The 32-note chime system involves an extra contact under each of the 32 uppermost keys of the upper manual. When chimes are turned on, these contacts energize for each key three additional diode keyers, so that the notes heard in each case are a major chord such as C-E-G-C, the top note corresponding to the key. At the same time, the upper-manual voicing output is disconnected from the preamplifier and chime tones, taken from the 8′ SAW and 4′ SQ points (Fig. 15-7) are passed through a voicing filter and substituted. Long sustain is also automatically turned on.

Pedal Generator

This organ is one of the very few spinets to employ a completely independent pedal generator, rather than simply one or two frequency dividers triggered by selected tones from the main generators. The only possible reason for this rather expensive and painful refinement would be elimination of vibrato from the pedal tones. While vibratoless pedal notes are highly desirable, they are, so far as the general public is concerned, one of the subtler advantages. This is, therefore, another example, among several in this book, of a designer going the usual road toward a rather inexpensive instrument, with several corners cut, and suddenly coming out in a rash of elaboration over some unexpected point. No one can argue with this point, except on economic grounds, and you may well be interested in the idea if you have the same problem and are similarly conscientious.

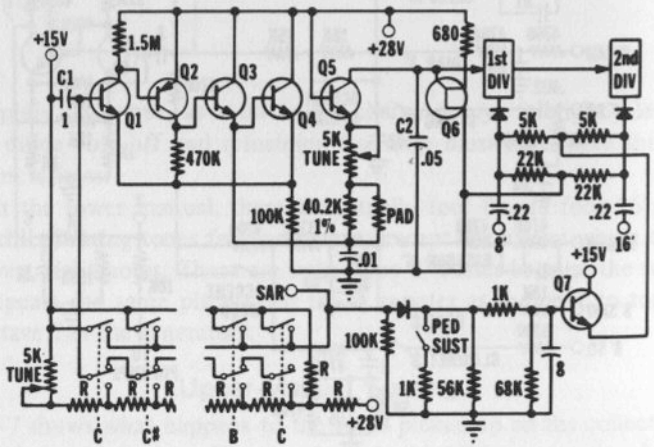


Fig. 15-9—The pedal generator and pedal sustain circuitry. The two lowest and two highest pedal switches, which are shown, are representative of all the remaining ones of the 13.

Since the pedal generator is monophonic and there is a pedal sustain, the first problem is to maintain the oscillator at the pitch of the last pedal pressed after that pedal has been released and the decay is going on. The solution is highly ingenious, as illustrated by Fig. 15-9.

The generator itself is a relaxation oscillator formed with a unijunction transistor Q6 and capacitor C2. When a certain critical potential is reached at its emitter (the terminal to which C2 is connected) the transistor suddenly conducts and its emitter shorts to ground. Knowing this, we can see how it oscillates. Assume that at least this much potential is being supplied the Q6 emitter by the collector of Q5, which has high internal resistance. Q6 conducts, shorts emitter to ground, in the process discharging C2. The trigger potential, having been shorted, no longer exists and the transistor ceases to conduct. However, before the potential at the emitter can again be enough to cause conduction, it must charge C2. This charging is a gradual process. When the charge on C2 has reached the critical potential, the transistor again conducts and the process starts again. You will see that this quick discharge of C2 followed by gradual charge is a sawtooth wave, which appears at the top of C2.

A very important point is that the repetition rate of the oscillations can be controlled by the value of the potential supplied through Q5. If that potential is high, C2 will charge quickly after each discharge and the repetition rate is high—and *vice versa.* We thus have a voltage-controlled oscillator. If it were not for the complication introduced by the sustain requirement, it would be a simple matter to connect the pedal switches along a voltage divider so that each switch passed to the top of C2 (through a high resistance) the voltage necessary to produce the correct frequency.

Such a voltage-divider string does indeed exist. It consists of all the resistors R connected in series between the +28-volt and +15-volt supplies; pressing a pedal selects one and only one value of voltage. That voltage is not, however, passed directly to C2. It is used instead to place a charge on C1, and that charge is passed via an extremely high-impedance unity-gain amplifier Q1 through Q5, to C2. The point is that when a pedal is released, the charge which it placed on C1 remains there for several seconds without appreciable change, simply because the amplifier impedance is so high that no significant discharge can take place through it. At the same time, however, being direct-coupled and highly stabilized, it does transmit whatever charge is on C1 to C2. Naturally, all major components must be of the highest quality. The resistors R in the voltage divider are 1% wirewounds and C1 is a low-leakage unit. The five-transistor amplifier is a Darlington type, with trimmings for rigid d.c. and temperature stabilization. Q5 is a common-base circuit.

The upper tier of pedal switches is for sustain control. When any pedal is pressed, +15 volts is passed to the base of emitter follower Q7, also charging the 8-μf capacitor. The positive voltage which then appears at the emitter of Q7 causes the two diodes in series with the divider outputs to conduct, so that 16′ and 8′ tone go to the voicing filters. When a pedal is released, the 8-μf capacitor discharges quickly if the Pedal Sustain switch is closed, through the 1,000-ohm

resistors. If the switch is open, it must discharge through 56,000 ohms, which causes a gradual decay.

Fig. 15-10 shows the voicing filters for the three pedal stops. If you were wondering what the SAR terminal was in Fig. 15-9, your question is about to be answered, for it has to do with Select-A-Rhythm.

Select-A-Rhythm

The Seeburg Select-A-Rhythm is an electronic device which takes the place of the percussion player found in popular combos or the sideman who accompanies the solitary saloon piano player. The original electronic rhythm device was the Sideman, made by Wurlitzer, which worked on entirely different internal prin-

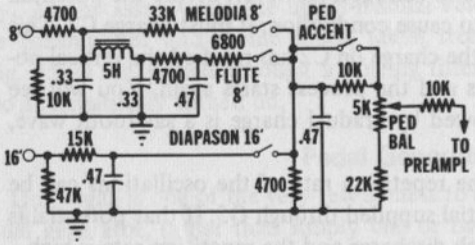


Fig. 15-10—Voicing for the pedals.

ciples. Select-A-Rhythm automatically produces any of 18 different rhythmic patterns, using selectively the sounds of 10 different percussion instruments: sand block, brush cymbal, snare drum, tambourine, claves, cow bell, two differently tuned conga drums, bass drum, and cymbal.

To illustrate, a typical rhythm, which we shall use throughout to exemplify the system, is the Viennese waltz. A waltz is an *um*-pah-pah or *1*-2-3. On the count of 1 there is a bass drum, on 2 a cymbal, and on 3 a snare drum.* This rhythmic pattern is repeated continually as long as the Viennese waltz is turned on. It proceeds at a fixed tempo determined by the setting of a knob, unless the organist turns on the "follow" function, in which case it gradually adjusts its tempo to that of the organist's own rhythm on the pedals of the organ.

Select-A-Rhythm is so complex that it really would not warrant an explanation in this book if it were used only on the Seeburg organs. It is, however, also sold separately for use with other instruments and small ensembles and will undoubtedly long survive the demise of Seeburg organs, so it has a greater importance. It has the expected drawback that it is really too monotonous for the best musical effect, but it is definitely useful and is certainly the best of the available devices of its type at this writing. The "follow" function is supplied only in the organ, and it takes a good deal of practice and control for an organist to lead the rhythms, as it takes time for them to adjust to his playing and the instinctive tendency is to *follow* the rhythm instead of leading it. The 18 preselected rhythms cover a good range—two waltzes, ballad fox trot, march polka,

*In actual fact, there is also a combined cymbal and snare drum a sixteenth-note before beat 2, but we shall ignore that as it would only muddy the waters of what is going to be a rather complicated explanation anyway.

Dixieland, plus several Latin American types and others—all controlled by a row of interlocked pushbuttons. Two or more buttons can be pressed at a time, mixing the rhythms to obtain additional effects.

Let us not worry, for the moment, about just how the actual sounds are made. Let us assume simply that they are made somehow, and that *each one speaks when it is fed a negative pulse.* Our greatest concern is with the method of providing the negative pulses at the correct rhythmic instants.

Timing Generator

The first requirement in a system of this kind is a decision on division of the musical bar. For example, if the basic bar were to be considered as having four equal parts, rhythms could be created for 4/4 time, in which each part would be a quarter-note, or for 2/4 time, in which each part would be an eighth-note. However, a rhythm in 3/4 or 6/8 time would be impossible. Since the important rhythms are based on both three and four beats, the basic count of a measure must be some number divisible by both these numbers, meaning a minimum of 12.

With 12 counts per bar, 3/4 time would have 4 counts per beat and 4/4 would have 3. One could also break down 3/4 time into eighth-notes, each having 2 counts; but there are not enough counts to break down 4/4 into eighths. Select-A-Rhythm therefore uses the second multiple of 12, or 24 counts to a bar. This allows eighth-notes for both 4/4 and 3/4 time, as well as a good measure of flexibility in smaller divisions for certain rhythms. And since the designer wished to deal with 2-bar units, he chose 48 as the total count of the system. There is, therefore, a sort of electronic commutator or rotary switch having 48 positions. One can visualize the arm of this imaginary switch rotating in discrete, evenly timed steps like the sweep hand of a nonelectric clock or watch, contacting each of the switch studs from 1 through 48 over and over again, each full rotation corresponding to two bars of music.

If the system were thus mechanical, one would number the studs 0 through 47 (48 being the return to zero). Then one could connect the bass drum to stud 0, the cymbal to stud 8, and the snare drum to stud 16, repeating this sequence

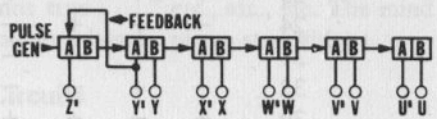


Fig. 15-11—Block diagram of the 6-flipflop Select-A-Rhythm counter.

again for studs 24, 32, and 40. An early rhythm device called Rhythm King, put out by the Kinsman precursors of Seeburg in the organ business did indeed use a mechanical commutator, as did the Wurlitzer Sideman.

The actual method of triggering the rhythm sounds begins with a binary counter operating to a scale of 48, consisting of six flipflops exactly like those in the organ tone generators. They are diagrammed in block form in Fig. 15-11, each with an identifying letter from U through Z. Each is divided into two parts, A and B, corresponding to the two transistors of the flipflop. The outputs are

taken from the collectors. Each flipflop can, of course, have two outputs of opposite polarities at any instant, and these are designated by a plain letter and the same letter with a prime mark (such as U′). The first flipflop is triggered by a

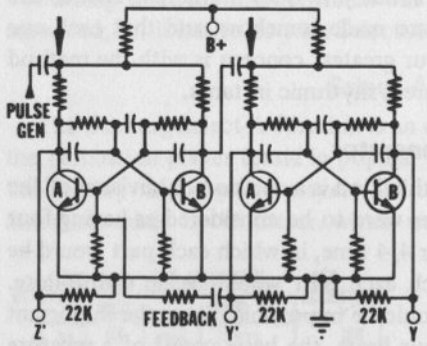


Fig. 15-12—First two counter stages show feedback connection. Flipflop values are the same as generator dividers in Fig. 15-3.

constant train of equally spaced negative pulses from a relaxation oscillator whose rate can be controlled for the purpose of controlling the tempo.

If we ignore the feedback connection, the six flipflops will generate a count of 64; that is, the last one will give one output for every 64th input generator pulse.

The feedback connection modifies the first two stages, however, so that stage Y goes through only three cycles instead of four for every four generator pulses.

TABLE 15-1

COUNT	V	W	W′	X	X′	Y′	Z′	WALTZ
47	+	+	−	+	−	−	−	
0	−	−	+	−	+	+	−	1A
7	−	+	−	−	+	−	+	
8	−	+	−	−	+	−	−	2A
15	+	−	+	+	−	+	−	
16	+	−	+	+	−	−	+	3A
23	+	+	−	+	−	−	−	
24	−	−	+	−	+	+	−	1B
31	−	+	−	−	+	−	+	
32	−	+	−	−	+	−	−	2B
39	+	−	+	+	−	+	−	
40	+	−	+	+	−	−	+	3B

X thus produces a cycle for every 6th pulse, W for every 12th, V for every 24th, and U for every 48th. Just how this feedback produces three output cycles from stage Y for every four generator pulses is a rather long-winded explanation which can be found in any good book on computers. However, I have provided Fig. 15-12, a diagram of the first two stages, so that you can figure it out yourself if you enjoy mental exercise. The one bit of necessary knowledge is that only a negative pulse applied to the arrowed point of each stage will cause it to flip or flop.

Component values are exactly the same as in the generators, Fig. 15-3, with the addition of the 22,000-ohm resistors at the outputs.

If we assume, as has been arranged by the designer, that before events begin all B transistor collectors are positive (and thus all unprimed outputs), we can construct a table listing each of the counts from zero to 47 and showing whether at each count each of the 11 counter outputs is positive or negative. We have made Table 15-1 which gives this information, but we have included only selected counts and selected counter output terminals—those necessary to understand the Viennese waltz in detail. It is obvious that for each of the 48 counts (which we label 0 through 47 because the first one comes at "time zero," musically speaking, the start of the selection of the bar) the distribution of positive and negative at the counter terminals is different. To say it differently, each of the 48 counts produces a wholly unique pattern of polarities at the terminals, as logic indicates it must.

You now have the essential clue to the system. If we can provide a series of gate circuits, each of which will recognize one and only one of these unique patterns of polarities, we can connect the gate inputs to the counter terminals and the gate outputs to the sound generators. Then, whenever that unique pattern comes along during the count, and then only, the sound will be heard. Such "recognition gates" are common in computer logic circuitry. They are known as AND and OR gates. We shall not use this terminology because we are not giving a course in computer logic. Briefly, however, they are circuits of diodes and other elements which allow an output signal to occur either (a) when *all* of the circuit inputs are provided signals of the correct polarity (an AND gate) or (b) when *any* one or more of the circuit inputs is provided the right signal (an OR gate). AND and OR circuits can be combined and cascaded in such complex ways that the arguments of the most accomplished sophist seem childlike by comparison. A simple transmission gate may also, of course, be used; it has only one input and transmits an output whenever that single input is given the correct polarity.

A very complex gate may, in effect, say, "I will transmit an output signal when A and B are true, as long as C is not true, unless D is also true, in which case E must also be true, unless F is not true . . . " etc., etc., etc. The mind boggles at the possibilities, but even if you are a boggler please stay with us.

Logic Circuits

In Table 15-1 we have shown the conditions at counter output terminals for those counts which occur on the 1-2-3 beats of the waltz rhythm (and also those counts which just precede each of the actual waltz beats). In the right column, headed Waltz, we show the three waltz beats for each of the two musical bars covered by the 48-count system; the first bar is indicated by A, the second by B. Thus the *um* occurs at count 0 and again at count 24, while the *pah-pah* occurs at counts 8 and 16 and again at 32 and 40. We have also shown only those counter output terminals which are used for this waltz rhythm.

Fig. 15-13 shows the simple diode gate used for the bass drum part of the

waltz. The bass drum is heard only on beat 1 of each bar. The gate input is connected to counter terminal V through a capacitor. This means that output will appear from the gate, not when the polarity at V *is* negative (note diode polarity) but when it is negative-*going*—that is, *changing* from positive to negative. The reason is, of course, that a steady potential cannot pass through a capacitor; only a *change* in potential can do so. Looking at Table 15-1, note that V changes



Fig. 15-13—Bass drum gate.

from positive to negative at counts 0 and 24 only—so at counts 0 and 24, and at no other time, a negative output pulse is produced in Fig. 15-13. Counts 0 and 24 coincide with the first beat of each measure of the waltz. So when the output of the gate is connected to the bass drum generator, a bass drum sound is heard on the first beat of each measure. (You may take my word for it that even if the full list of which Table 15-1 is a part were printed, you would find no other counts on which V changed from plus to minus.)

Consider next the gate which is to operate the cymbal on the second beat of the waltz, corresponding to counts 8 and 32. It is shown in Fig. 15-14. Before we discuss it, recall that we said a negative pulse is necessary to operate each sound generator. Remember that a negative pulse is obtainable in two ways. It can be an actual pulse of electrons passing through the gate and down to ground through the load resistor, if in the absence of these electrons there are none; that is, the change in potential at the output is from ground potential to one more negative than ground. A negative pulse may also be a change from a normal positive voltage across the output resistor to zero voltage. The latter applies to Fig. 15-14.

Let us see first what happens to this gate on a count *other than* those in which we are interested—47, for instance. The table shows that V is positive, and

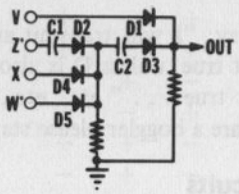


Fig. 15-14—Cymbal uses two cascaded gates.

we need look no further, for regardless of what may be appearing at the other terminals, the positive potential at V causes current to pass through the diode D1 and makes the output of the gate positive with respect to ground.

On count 0, V is negative, so D1 is blocked. Z′ is also negative; D2 and D3 do not even pass a pulse. X is negative, so D4 is blocked. But W′ has just gone positive (it was negative at count 47) so that unblocks D5 and passes the positive-going potential through C2 and D3, so that the output is still positive.

At count 7 V is negative; no conduction through D1. Z′ is positive, however, having just changed from negative at count 6 (as you would see if you had the whole table). That positive-going pulse passes through C1, D2, C2, and D3, and again the output is positive. If you continue this analysis you will find, in fact, that due to at least one of the inputs, the output of this gate is positive for every one of the 48 counts except 8 and 32.

On count 8, V, Z′, X, and W′ are *all* negative—and the same is true at count 32. Therefore, for these counts only, the normal positive output of the gate disappears. Connected by the switch selector system to the cymbal generator, the cessation of positive potential is seen by that generator as a negative pulse and the cymbal goes to work, furnishing the first *pah* after the *um*.

On the third beat of the waltz, counts 16 and 40, pulses are needed for the snare drum. They are furnished by the gate of Fig. 15-15, which operates in the same way as that of Fig. 15-14. Let us see how its output is kept positive except during counts 16 and 40.

On count 47, W is positive. On 0, Y′ is positive. On 7, W is positive. On 8, W is still positive. On 15, Y′ has just gone positive, so a pulse goes through the Y′ capacitor. And so on. However, at 16, Y′, X′, and W are all negative, and while

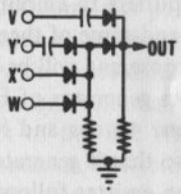


Fig. 15-15—Cascaded gates for the snare drum.

V is positive, it has been that way and is not positive-*going*, so transmits no pulse through the capacitor. The gate output is thus negative with respect to its usual positive condition, and the snare drum sounds.

You will now be able to see how pulses can be produced at any desired counts. To avoid duplication, there are two groups of gates, and the total circuit for any particular combination of counts is usually made up of two cascaded, such as the pair in each of Figs. 15-14 and 15-15. In each figure, the first gate's output goes not only to the second one shown but also to other "second" ones. As you can imagine, figuring out the details of a system like this is a head buster, though it is only time consuming and not really difficult.

The final result of combining all the gates is 39 different rhythmic patterns, such as the three we have described. Each is meaningless by itself, but when combined, as we have shown these three to be combined, a familiar rhythm is established. This is done by the pushbutton switches, each of which is 6-pole, single-throw, normally open. For the waltz pattern only three were needed, one each for the bass drum, cymbal, and snare drum. Others are more complex and can include up to six instrument sounds. In the process of combination, several gate outputs can be channeled to operate one sound, or several sounds can be operated by one or more gates. Wherever a combination of things might result in sending potential back up a line where it would interfere with something else, a diode is inserted to block back-voltage.

The pulse generator which drives the counter system is a unijunction transist-

or oscillator whose frequency is corrected for the "follow" function by an integrated voltage derived from the organist's pedal repetition rate.

Instrument Generators

The instrument generators contain ideas that designers will find interesting. Let us take them one by one. The noise generator referred to in some cases is a reversed transistor operated as a zener diode, with its output amplified by another transistor, just as in the Wurlitzer Ssh-Boom. Some of these generators are preceded by amplifiers to amplify and invert the negative pulses received from the logic system and some of them are followed by preamplifiers. All this is irrelevant for our purpose and will be ignored. All B+ points are at 15 volts.

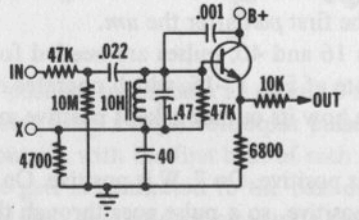


Fig. 15-16—The bass drum generator.

The Bass Drum generator of Fig. 15-16 is nothing more than a tuned circuit operating at about 40 cps and having reasonably good Q. The incoming pulse shock-excites it so that it generates a short damped-wave train which is passed to the output by the emitter follower. Point X is an a.c. ground used by several of the generators.

The Lo Conga Drum generator of Fig. 15-17 vaguely resembles a Colpitts oscillator but does not have enough feedback to sustain oscillation. The negative incoming pulse shock-excites it into momentary oscillation, again producing a damped-wave train, but at a higher frequency and with different decay characteristics from that of the Bass Drum. The Hi Conga generator is identical except that its inductor is 5 mh and the unspecified capacitor across the coil is missing.

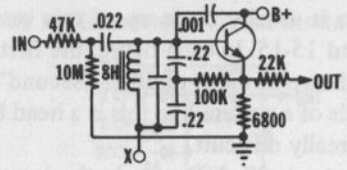


Fig. 15-17—Lo conga generator.

Fig. 15-18 shows the Cow Bell generator. This, as you will know if you have spent time in a farmyard, is a very short metallic sound—it does not ring like a true bell—with a strong "bonk." The input pulse shock-excites both a circuit like that of the conga drums and a higher-pitched one without an amplifier for the metallic sound. The figure also shows the Clave generator. (This is pronounced *clah'-vay;* they are usually used in pairs with different resonant frequencies.) This again is a shock-excited tuned circuit.

The Brush Cymbal generator of Fig. 15-19 is typical of the ones using the

noise generator, the idea behind which was discussed in the description of the Wurlitzer Ssh-Boom. Noise is fed to an amplifier with a tuned circuit in its collector load, shaping the pass band so that with the proper transients the sound will be that of a cymbal lightly struck with a wire brush. This transistor is cut off

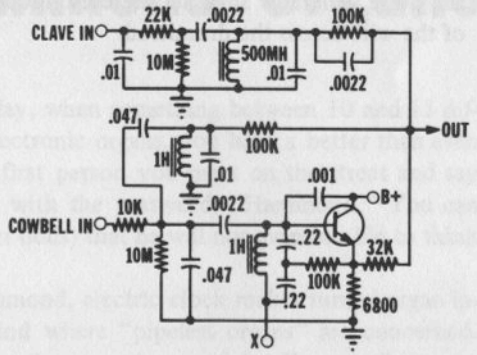


Fig. 15-18—Cowbell generator is compound circuit; clave uses no transistor.

by having its base and emitter basically at the same d.c. potential due to the 970,000-ohm (that's what it says!) resistor, aided by some positive emitter bias from the power supply, maintained constant by the series diodes.

The first transistor is a pulse shaper and inverter. It normally conducts, maintaining collector voltage negative with respect to the B+. When a brush input pulse (negative) comes along, it cuts off the transistor. Collector voltage promptly rises to the B+ value. The positive-going pulse goes through D1 to charge C. The positive charge goes to the base of the generator transistor, which it turns on so that there is output. C cannot discharge through D1 after the input pulse

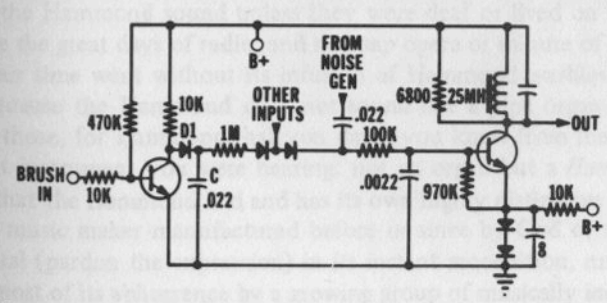


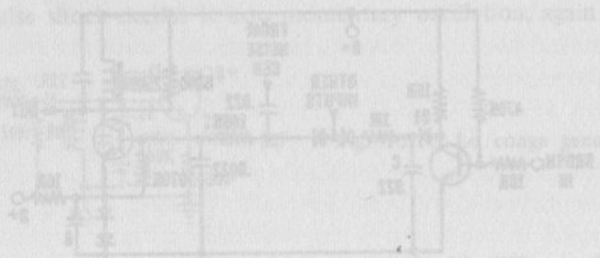
Fig. 15-19—The brush sound comes from a timed bandpass gate, also used for other sounds.

leaves, so it discharges through the high-impedance path relatively slowly (though still not so slowly as a sustain) and the generator output dies away gradually.

The struck Cymbal sound is generated both by this and by another similar generator simultaneously, while the Tambourine sound is created by this generator, keyed by a pulse amplifier with a much larger capacitor, giving a longer decay. The Sandblock sound, which is continuous for the duration of its counts rather than pulsed, is also obtained from this generator, keyed directly by the

logic circuits rather than through a decay circuit. Cymbal, Tambourine, and Sandblock keyers are connected at the point marked Other Inputs.

The Snare Drum uses a generator like that of Fig. 15-19, plus a shock-excitation circuit like that of the Clave, Fig. 15-18. The latter gives an initial rap and the noise generator suggests the continuous beating and the vibration and noise of the wire across the drumhead.



Chapter 16

Hammond Organs

Even today, when something between 10 and 15 different manufacturers are making electronic organs, you have a better than even chance of winning a bet that the first person you meet on the street and say "electronic organ" to will counter with the password, "Hammond." You can even bet (though with much smaller odds) that he will not even be able to think of another brand.

Such is the impact Laurens Hammond, electric clock maker turned organ inventor, has had on the public mind where "pipeless organs" are concerned. There are two excellent reasons for the ascendancy of the Hammond organ—priority and reliability.

The Hammond organ was commercialized in the early thirties and until after the war it had no serious rivals. There just was no other nonpipe organ that could be easily purchased, so wherever an organ was wanted that did not require the space, money, and maintenance of pipes the Hammond was the sole choice. (The Everett Orgatron, a reed-electronic organ, was available before the war but never really competed with Hammond.) So Hammonds turned up in astronomical numbers in radio stations, funeral parlors, night spots, homes, and darkest Africa. Even people who never went anywhere or bought anything became familiar with the Hammond sound unless they were deaf or lived on the moon, for those were the great days of radio, and no soap opera or minute of otherwise unoccupied air time went without its infusion of Hammond warbles and squeals.

And because the Hammond does not sound like a pipe organ—especially as played in those, for Hammond, halcyon days—you knew from the first note exactly what instrument you were hearing: not an organ but a *Hammond* organ. The fact that the Hammond had and has its own highly distinctive sound, unlike any other music maker manufactured before or since by God or man, has been instrumental (pardon the expression) in its instant recognition, much of its success, and most of its abhorrence by a growing group of musically inclined people.

To get rid of the subject of Hammond sound at the outset, let us say that Mr. Hammond, according to the rumors I have heard for many years, was under no illusion that his instrument sounded like a pipe organ—and we shall later detail the reasons for this organ's aural distinctiveness. No one, the story goes, was more surprised when the organs became so popular for church work. And there is—and always has been—a large body of opinion, principally but not entirely among the musically literate, that the instrument's sound is totally unacceptable and even unpleasant. Pipe organists often object purely on the ground that an organ should sound like a pipe organ. My own opinion is that this is true

when the organ is used for liturgical purposes and to play the works of the classic masters; but if someone wants to sell and others to buy a completely *new* kind of instrument for popular and light music, which is never played without arrangement and adaptation anyway and thus has no "authentic" sound, why not? It then becomes a matter of personal taste, with which it is hard to argue.

However, there are those who cavil on more or less objective grounds–if you can really be objective about a subject like music. Hammond total tone, as we shall see, is put together by a method that actually could not exist, except to a limited extent, in an acoustic instrument–combination of harmonics in fixed ratios regardless of fundamental pitch. And because a separate generator element is required to produce each harmonic, the practical limit in the number of elements limits the highest available harmonic in the organ to about 6,000 cps, automatically eliminating any possibility of producing convincing strings or bright

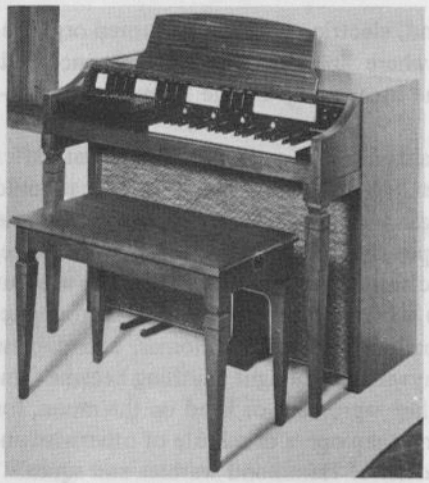


Fig. 16-1–A current Hammond chord organ, the S-101.

reeds–or in fact any bright sounds even without reference to traditional tone colors. And indeed Hammond tone is characteristically bland, though this carries dividends by making it impossible for inept or tasteless players ever to create a harshly unpleasant sound.

To sum up and finish the subject, a Hammond always sounds like a Hammond and nothing else–but if that's what you enjoy, who can argue with you? There has certainly not been enough argument to make Hammond Organ Co., 4200 West Diversey Ave., Chicago, Ill., take any poverty oaths. Until a very short time ago, Hammond sales exceeded those of the rest of the industry combined. According to figures whose source I cannot find, this impressive lead no longer exists, Lowrey and Thomas having achieved ever increasing penetration of the market. It is interesting to note that these companies, too, have displayed more brilliance in merchandising than in tone colors–so maybe soothing syrup is pretty much what the majority of the public wants from an organ after all!

About one Hammond attribute there can be no argument at all. Hammond organs are and always have been inherently trouble free. The principal element,

the tone generator, is mechanical rather than electronic (Hammond prefers its organs to be called "electric" rather than "electronic") and the other major components such as keyboards, switches, etc., are built like the proverbial battleship. I have had occasion to play one of the first commercial Hammonds sold and found it in nearly mint condition aside from some dirty drawbar contacts. Part of this ruggedness is in the nature of the beast–lack of complex electronic circuits–and an equal part in the very conscientious and knowledgeable mechanical design which was done very early in the game and has not changed materially since. I am certain that this long record of reliability remains an important factor in Hammond leadership. Despite the distaste of many electronics people for the idea, it happens to be true that sometimes a well designed mechanism will work longer and more predictably than the best of electronisms. If this be treason, make the most of it!

Product History

The original Hammond organ was the electromechanical instrument for which Hammond is principally known, of which we shall describe a typical modern example in this chapter. However, the company has by no means confined itself to



Fig. 16-2–A drawbarless, tone wheelless Hammond electronic–not "electric"–organ, the 3022.

this type of product. In fact, everything Hammond has produced, and even almost everything it has developed without producing commercially, as evidenced by its patents, has been unconventional, different from other musical instruments and often daringly so. Its late chief development engineer, John Hanert, probably had more issued patents in electronic musical instruments than any other individual, almost all of them highly ingenious and, at least in their times (and in some cases still), most unusual. One, for instance, was a voice-operated

instrument. Humming into a sort of tube turned on electronic generators at the correct musical frequencies nearest the voice pitches, following which the generated tones were colored and vibrated as in an organ.

The earliest major evidence of this originality was the Novachord, a single-keyboard instrument with a large variety of percussive effects. It was very complicated to play and even more so to service, so much so that after the war production was not resumed. Before the war, however, it was often heard on the radio, though few of them reached homes.

Hammond was the first maker of an electronic chord organ—a sort of counterpart of the accordion, providing a keyboard for melody and a large number of buttons, pressing any one of which produces a complete chord and bass note.

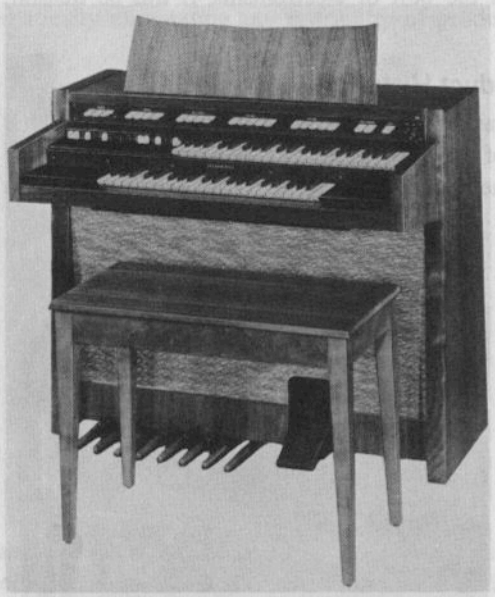


Fig. 16-3—One of the Hammond spinet organs, Model M-102.

The S-100 shown in Fig. 16-1 is the latest version of the chord organ. These were extremely popular for several years because the public was intrigued with the idea of being able to play almost immediately without needing to learn to play full chords. However, as the organ field added manufacturers and customers by leaps and bounds, the rather stringent limitations on what could eventually be played with a chord organ became apparent, and people leaned more toward standard instruments. The chord organ is still made, but it enjoys a very minor part of the market.

Another Hammond innovation was the Solovox, a small keyboard which could be attached to the front of a piano. It was used to play one-note-at-a-time melody, with sustained organ-like (more or less) tones, while accompaniment was played on the piano. It, too, was very popular for years, especially with public entertainers, but it died and production was stopped. It lives on, however, in the pedal solo unit common to some present Hammond organs, which is

built on the same lines as the Solovox, circuitwise. The Solovox is fully described in the first two editions of this book.

In recent years a significant portion of the public has shown itself interested in purely electronic organs. Hammond's first answer to pure electronics (aside from the chord organs, which were fully electronic but too different otherwise to count) was the Extravoice. Following its tradition, Hammond made the playing and registration facilities of the Extravoice highly unconventional. While the attempt showed plenty of ingenuity, people found it too complicated. Despite much advertising and publicity, the Extravoice had to be laid to rest and, with the Edsel, belongs to history.

With a certain (unmusical) sadness, one must note that today Hammond has, to an extent, joined the mob and is now producing some purely electronic organs very much like those most common in the industry—frequency-divider generators and formant tone coloring. Typical is the 3000 series, represented by the 3022 of Fig. 16-2. But most Hammonds sold are still of the original genre—electric organs with the mechanical generator, not very much changed from Laurens



Fig. 16-4—Organs of the D-100 series, described in this chapter, differ mainly in console styling. This is the D-152, of Tudor design in walnut.

Hammond's first public offering back in the thirties, still clung to with love by its adherents and still sold in extremely respectable quantities. The M-102 of Fig. 16-3 is a popular spinet type (Hammond initiated the "spinet" organ), and the D-152 in Fig. 16-4 is a typical larger model. It is the D-100 series, of which the D-152 is an example, which we shall describe in this chapter.

Tone-Wheel Generator

The heart of the Hammond organ and its real distinguishing feature is the

electromechanical tone generator, the complicated looking assembly shown in Fig. 16-5.* It is a collection of motors, gears, shafts, springs, and special metal discs, plus a few electrical items, mounted on two long steel angles and handled as a single unit. Let us dissect it.

Fig. 16-6 shows the essential parts of the assembly for generating a single note. The tone wheel is a steel disc about 2 inches in diameter, accurately mach-

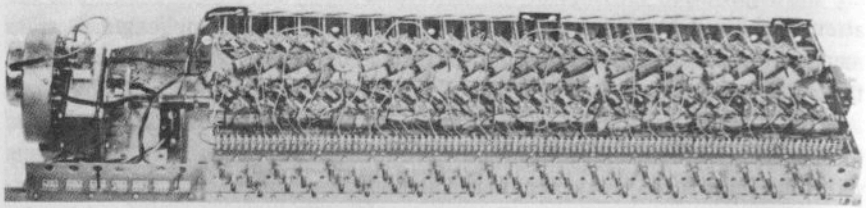


Fig. 16-5—One of the world's most famous collections of hardware, the Hammond tone generator, with its motors, shafts, gears, couplings, and the rotating phonic wheels.

ined with a certain number of high and low points about its periphery. The wheel rotates at a constant speed. Associated with the wheel is a permanent magnet in the form of a rod, the tip of which is ground to a sharp chisel edge and placed as near as possible to the edge of the rotating wheel. A coil of wire is wound around the rod near its ground end.

Each time a high point on the wheel passes the end of the rod, it causes a change in the magnetic field; and this induces a small voltage in the coil. The frequency of the coil voltage depends on how many wheel points pass the rod

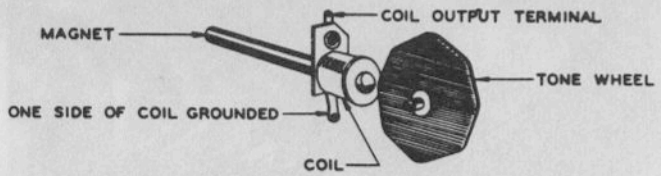


Fig. 16-6—The essential elements for generating one tone: rotating phonic wheel and coil wound on permanent-magnet core.

per second. If, for example, the wheel has 8 "teeth" and rotates at 55 rps (3300 rpm) the coil provides 440 cps, middle A. There are 91 of these wheel-magnet assemblies, rotating at different speeds and with different numbers of high spots, which give 91 different frequencies. The notes generated comprise 5 notes less than eight octaves and include all the frequencies of Table 1-1 except the five highest notes of Octave 8.

Small coils are used for the higher frequencies and larger ones for the lows.

*Please forgive the rather poor quality of Fig. 16-5. As an attestation to how long ago this generator was designed and pictures of it printed in Hammond's various service manuals, Hammond, with the best will in the world, was unable to find an original photograph of it for use in this book. Fig. 16-5 has therefore been rephotographed directly from a printed page. The same comment goes for the drawbar photo of Fig. 16-11.

Larger rods are used near the bass end to give sufficient output (which, as in any magnetic generator, is a function of velocity, lowest at the bass end) and smaller ones at the higher frequencies to reduce iron losses.

For reasons you will see later, an object is to generate coil voltages which are as nearly sine in waveshape as possible. Various methods are used to reduce the harmonics. First, the wheel shapes are designed to give as nearly a sine wave as possible. Second, low-frequency coils have copper rings on them. The eddy current losses in the rings are small at the fundamental frequencies but higher at the harmonics.

Third, filtering is used, as indicated in Fig. 16-7. For generators 44 through 48, a transformer which actually has a single winding with a grounded tap is the

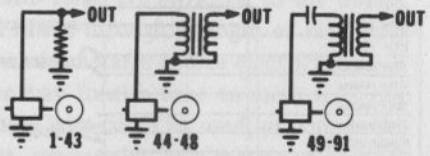


Fig. 16-7—Filters help eliminate harmonics and unwanted tones from generator outputs.

filter. That part of the winding which is across the generator coil resonates around the generator frequencies, which effectively reduces harmonics in the output section. For generators 49 through 91, a series capacitor aids resonance; its value is .255 μf for frequencies 49 through 54 and .105 for the rest. The cores of the various transformers have different numbers of laminations to secure the inductances required. Coils below frequency 44 have only shunt resistors, in the form of lengths of resistance wire. The output level of each generator is adjusted at the factory by setting the gap and then tightening a setscrew.

The entire generator assembly actually consists of 48 separate rotating assemblies, each running in a separate compartment magnetically shielded from the others by steel plates which divide the whole thing into a series of bins.

Fig. 16-8 shows what is in one of the 48 bins. There are two of the tone wheels, each coupled to a Bakelite gear, both Bakelite gears meshed with a driving gear which is rotated by the main shaft running the length of the assembly. There are 12 different sizes of both Bakelite and drive gears, corresponding to the 12 notes of the chromatic octave, and four of the assemblies of Fig. 16-8 (a total of eight tone wheels) run at each of the 12 speeds. This would make for a

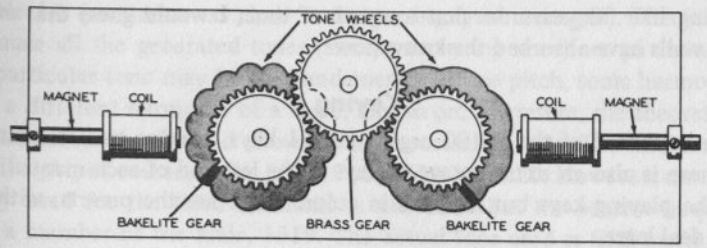


Fig. 16-8—Contents of one bin in the generator—drive gear driven by main shaft through resilient couplings and two tone wheels and gears. Bakelite gears help to eliminate noise.

total of eight octaves of generated tone, except for the fact that the five highest tones are not wanted, so five of the compartments have one "blank" tone wheel each, left there just for mechanical balance.

The main shaft is driven by two motors. One is a shaded-pole induction motor which has enough torque to start the assembly from rest but is not synchronous. The other is a synchronous motor with enough power to keep the assembly running but not enough for starting. The starting method, familiar to the very oldest among us, involves two switches with large handles. The first, labeled START, is spring loaded. It is held in the ON position for 8 to 10 seconds. Then the RUN switch is turned on and the START switch released. After about 4

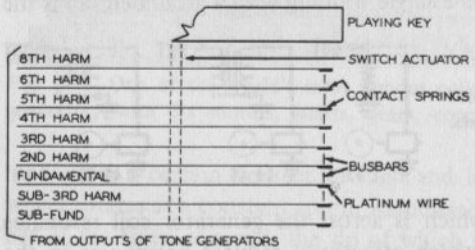


Fig. 16-9—This stack of switch arms and busses is under each playing key. Late instruments use palladium instead of platinum.

seconds the system slows down to synchronous speed and is ready to go. When the starting motor goes off, it is disengaged from the shaft by a spring. There are now some late instruments with only one motor.

The synchronous motor does not deliver steady power, but rather a series of pulsations, one per half-cycle, from the 60-cps power line. To prevent these pulsations from reaching the tone wheels and causing spurious frequencies in the output, resilient couplings are used throughout the drive system. The synchronous motor is resiliently coupled to the shaft, the shaft is divided into several sections by flexible couplings, and each Fig. 16-8 assembly is resiliently coupled. Porous bronze bearings are used for the tone wheel shafts, lubricated by oil from a trough, carried to the bearings by capillary threads. Oil should be added once a year.

The mere description of this generator assembly carries the impression (at least to an electronics-oriented individual like me) of great complication. And indeed it is an assembly which must be made with precision and strength. Hammond has the inestimable advantage of having manufactured this system for something like 30 years. In that amount of time, I would guess that even the factory walls have absorbed the know-how!

Keying

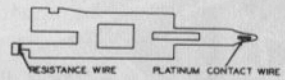
Each keyboard of the D-100 organs has 61 keys, making two standard manuals. There is also an extra octave of keys at the left end of each manual, identical to the playing keys but reversed in color. These are the presets, with which we will deal later.

Fig. 16-9 shows the electrical contacts under each of the playing keys. An insulating actuator is pushed down by the key; in turn, it presses down nine movable contacts so that each one touches a bus. There are nine of these busses, each running the length of the manual.

Fig. 16-10 is a drawing of one of the nine movable contacts as seen from the bottom. The contact is a flat spring of roughly the shape shown (actually it is longer than this for the width shown), fixed to a nonmoving support near its rear end (left end in the drawing). The actuator passes through the rectangular punchout near the center and presses the spring down at that area. At the front of the spring (right), a small piece of palladium alloy wire is welded to the spring to form a noncorroding contact. When this goes down, it strikes a bus, end views of which appear in Fig. 16-9; the upper edge of each bus is also palladium.

When a key is pressed, nine very specific tones are switched to the busses, each from one of the generator coils and filters through a length of resistance wire which serves to isolate the key switch and generator source electrically, as is necessary in all designs where a generator may furnish tone to more than one circuit at a time. Fixed resistors and ordinary wire could be used, of course, but Hammond has always used resistance wire, presumably as a matter of convenience. The length must always be what is required for the right resistance, and this length does not always correspond to what is actually necessary to reach from switch to generator terminal. So the wire is sometimes doubled back in the harness just to have someplace to put the extra length.

Fig. 16-10—Switch arms look roughly like this.



The nine switches under each key and the specific tones they switch are an essential element of the unique Hammond system of tone coloring.

The switch labeled *fundamental* carries the tone normal for the key at what would usually be called 8′ pitch. Using from here on the A key above middle C as an example, the *fundamental* switch carries 440 cps. The *2nd harmonic* switch carries 880, the *3rd harmonic* 1320, the *4th harmonic* 1760, the *5th harmonic* 2200, the *6th harmonic* 2640, and the *8th harmonic* 3520. The *sub 3rd harmonic* switch carries 660, and the *sub fundamental* 220.

Now, the frequencies in the preceding paragraph are theory. In actual fact, all the tones coming from the generator are at the frequencies of the regular scale, as in Table 1-1 (or almost—as close as the gearing system can get). This is necessary because all the generated tones (except some top ones) are put to several uses. A particular tone may be the fundamental of one pitch, some harmonic of another, a different harmonic of a third, and so on. Therefore, the theoretically perfect harmonics enumerated above do not necessarily exist. The fundamental, 440, is all right, of course, as are 880, 1760, 3520, and 220. However, the actual frequency used for the third harmonic is not 1320 but the nearest frequency which is a member of the scale, 1319. The actual tone used as fifth harmonic is 2218, not 2200. And so it goes.

There are those who feel that these harmonic "errors" (as well as small errors

in the basic pitches because of the gearing) contribute strongly to the peculiarity of Hammond tone. These things are not important in themselves, though they would be important if the generated tones had any natural harmonics, for then the natural and synthesized harmonics would beat unpleasantly. But harmonics are deliberately and rather effectively filtered out. In fact, if you tried to judge the tuning of a Hammond by striking, say, C and G, listening only to the tones of the fundamental bus, you would hear no temperament beats; such a beat is normally caused by the third harmonic of the C and the second harmonic of the G. Even if you listened to all the busses of Fig. 16-9 at once while striking a musical fifth, you would hear no beats. The third harmonic of middle C would be taken from the G-784 generator, which is nearest to the theoretical 784.9—and so would the second harmonic of the G. So an outstanding characteristic of the Hammond is that it has no temperament beats. I can tell you with certainty that the absence of temperament beats greatly changes one's impression of an organ, so that this *result* of the harmonic errors is indeed a detectable peculiarity. I had occasion to experiment at length with a device which eliminated beats in a frequency-divider organ by automatically retuning the master oscillators each time

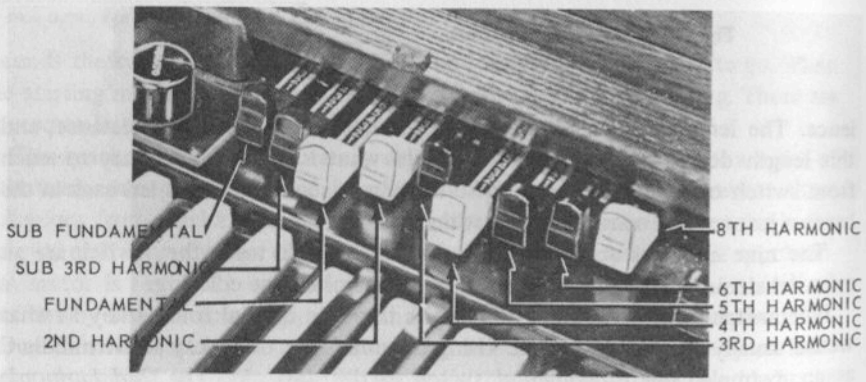


Fig. 16-11—Two sets of drawbars like these control the tone color for each manual.

notes were played. No-beat results are outstandingly apparent. We rejected the device because we felt strongly that the lack of beats made the sound lifeless and robbed it of fire, even with full reed choruses. There is, however, one effect of the lack of beats which is very useful to many Hammond owners. It is practically impossible to make an unpleasant sound on the organ even when wrong notes are played. On conventional instruments it is the clashes between harmonics that cause most of the unpleasantness when errors are made in harmonies. With no natural harmonics and all synthesized harmonics in agreement, this cannot happen, and discords hardly sound discordant at all!

The Harmonic Drawbars

We come now to the most obvious element of the Hammond organ to anybody who has ever seen one—the harmonic drawbars with which tone color is

controlled and by means of which it has been said, with theoretical accuracy but little practical meaning, that astronomical numbers of different tone colors can be obtained.

A group of nine drawbars appears in Fig. 16-11. To use them, the organist pulls them out or pushes them in, reaching up to their location above the upper manual to do so. The nine drawbars correspond to the nine key switches of Fig. 16-9 and each controls the intensity of the fundamental or a harmonic or subharmonic. The farther out you pull the drawbar, the greater the intensity.

With all nine drawbars pushed all the way in, keys on the associated manual produce no sound. As the third drawbar (the first white one) is pulled out, fundamental is heard. There are eight numbers on the metal part of the drawbar to show how far out it has been pulled. To set up a particular tone color, the organist pulls out the drawbars according to a predetermined formula which is expressed as a series of nine numbers. In doing so, he gives the resulting tone any or all of nine harmonic components, each in any of eight different intensities.

Knowing how to figure combinations and permutations, one could establish the maximum possible number of total settings at a very high figure indeed, and some Hammond salesmen (I do not mean the company itself) have used this "logic" to convince customers that the Hammond has a wider capability for tonal results than any other instrument. Human musical ears and brains, however, do not necessarily respond directly to mathematical propositions, and my personal estimate of the number of noticeably different combinations of drawbars available is somewhere in the thirties. The fact is that when several drawbars are in use, changing some of them by one or two steps or even more simply may not alter the total tone enough to be noticeable. And where a drawbar is fairly well pulled out, the effect of an adjacent one may not be audible at all until it, too, approaches the same setting. What is noticeable as a change depends on what exists to begin with in a number of different ways. But the number of *noticeable* changes is a very small fraction of the number of *possible* ones. Add to this the restriction that the highest harmonic available is below 6 kc, ruling out all brilliance. In sum, a considerable number of tone colors are available, but by no means more than on a good organ of another type, and very possibly fewer.

The most important point, however, is that the tone colors formed in the Hammond manner are almost wholly artificial in character, as reference to Chapter 5 will indicate. A normal musical instrument has a sound which is characterized by the effects of the shape of its *spectrum response.* This inevitably causes the harmonic structure—the number of harmonics for a given tone, their distribution, and their relative amplitudes—to vary depending on the fundamental pitch. The Hammond tone, on the other hand, has a harmonic structure which is constant for all fundamental pitches (except those too high to be able to obtain all higher harmonics from the 91 pitches available). The principle is known as *harmonic synthesis.* It is this factor almost entirely which makes Hammond tone unique and completely recognizable.

To a very small extent, a few other instruments share this constant-harmonic-structure character. Pipe diapasons do tend to have somewhat emphasized

second harmonics at all pitches; and a Hammond with drawbars properly adjusted can suggest a diapason within reason. Pipe organs also have intramanual couplers which can add octaves and fifths, and sometimes thirds, and constitutes a harmonic series with some harmonics missing. To an extent this parallels the Hammond idea, except that pipes always have natural harmonics which change with the fundamental, so that the aural effect of such couplers ends up being quite different from Hammond's.

Fig. 16-12 shows the simple manner in which the drawbars function. Eight busses are connected to eight taps on a transformer primary, and an additional bus is connected to ground and to the bottom of the transformer winding. Each of the busses of Fig. 16-9 is connected to a contact on one drawbar. As shown in Fig. 16-12, all drawbar contacts are grounded and no tone passes through the transformer.

If the fundamental drawbar (the first white one) is drawn (is pulled upward in the electrical diagram of Fig. 16-12), its contact, carrying fundamental tones of whatever keys are pressed on the manual with which this group of drawbars

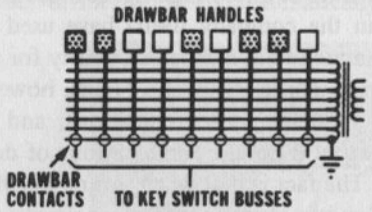


Fig. 16-12–How drawbars control harmonic mix. All drawbars are in off position in this diagram.

is associated, touches busses higher and higher on the transformer winding. This places more and more tone across the transformer. Obviously, each drawbar can be placed as desired, so that its contact touches the bus which will place the desired amount of that harmonic across the transformer. And in this way the function of the drawbars, as described earlier, is fulfilled. The transformer secondary leads to amplifier and output circuits which we shall describe later.

The Preset System

The octave of reverse-color keys at the bottom of each manual is a system of interlocked presets. Mechanically they operate in such a way that when one is pressed it stays down until a second is pressed; the second then stays down while the first comes up. The exception is the bottom (C) key, which is a cancel. When pressed, it releases any other key which is down and then comes up itself.

The organ actually has four identical sets of drawbars for the manuals, two per keyboard. Pressing the A# preset for a manual brings one set of drawbars into operation; pressing the B activates the other set. This means that two drawbar settings can be made in advance for each manual and then selected at will.

The remaining preset keys produce tone colors which have been set up by a system of wires and binding posts. Certain combinations are set at the factory, but they can be altered.

Technical operation of the presets is rather simple in principle. The nine harmonic busses of Fig. 16-9 run along under the preset keys as well as the playing keys. And, like the playing keys, each preset key (except the C cancel) has under it nine movable switch contacts. The only differences, in fact, between the preset keys and the playing keys are the reversed colors of the presets and their mechanical interlocking. The difference in *use* of the key switches under the presets is that tones are *taken from* the movable switch fingers, rather than fed to them by generators.

Let us first take the A# and B which bring drawbars into operation and dispose of them quickly. In Fig. 16-12 we have shown the drawbar contacts as going to the nine keying busses. Actually, they go to the nine movable contacts under the A# or B preset key. There is a second set of drawbars associated with the same transformer and transformer-tap busses, whose contacts are connected to the movable contacts under the remaining preset key. Thus, if one preset is pressed, the nine keying busses are connected by its moving fingers to one set of drawbars; the other set of drawbars cannot affect organ tone, as the contacts of the preset key from which they derive tone are not down touching the keying busses. And *vice versa.*

The preset wiring panel in the back of the organ contains nine horizontal lines of screw terminals. Each horizontal line of terminals is connected to one of the transformer taps of Fig. 16-12. Coming up from the bottom of the panel is a set of nine color-coded wires for each of the preset keys C# through A. Each set of wires is connected to the movable contact fingers under one preset key and thus each wire carries one harmonic when that preset key is down. If the fundamental for that preset is to have "amplitude 4," for instance, the orange wire, which carries fundamental, is attached to a screw terminal in the horizontal line of terminals which is connected to the fourth transformer tap in Fig. 16-12. And so on for the harmonics. This same setup–nine wires for each of nine preset keys, connected to nine rows of terminals–is repeated for the other manual.

Thus, whatever preset key is down is the only one whose contacts are touching the nine keying busses, and the only one which can take keyed tones from them. Those contacts send the tones either to a set of drawbars or to specific transformer taps. The system works just as described for both manuals, except that when the upper-manual B preset is pressed, there is additional switching for the percussion.

Percussion Electronics

Fig. 16-13 shows the electronics of the Hammond percussion; it does not include the switching, which we will go into later.

An input signal (source discussed later) which consists of tones selected by playing the upper-manual keys is applied to terminal H and amplified in a standard way by V1, which has some inverse feedback from plate to grid for stabilization of gain and a small capacitor from grid to cathode (actually a.c. ground) for chopping down the highest frequencies and preventing any possibility of oscillation. The plate load is winding A of transformer T1. Contain your curiosity about winding C for a few minutes.

V2 and V3 (actually two halves of a 12AU7) constitute a push-pull keyer stage, with the expected centertapped transformer load T2. The cathodes are fixed-biased at +37 volts.

Assume for a moment that V4 is invisible, along with everything connected to its grid and plate. The 30,000-ohm cutoff control is adjusted to a point where the grid voltage, obtained from the arm of the control through R1 alone or through R1 and R2 in parallel, depending on the position of the fast-slow decay switch, is just low enough (around +15 volts) so that V2 and V3 do not conduct. Signal applied to input terminal H will now *not* appear at output transformer T2.

Now V4 and the other components reappear. V4 is used simply as a diode, for which purpose plate and grid are tied together. Since the voltage on the plate-grid from the +33-volt source through R3 and R4 is more positive than that applied to the cathode, the diode conducts. The voltage at its cathode, and thus

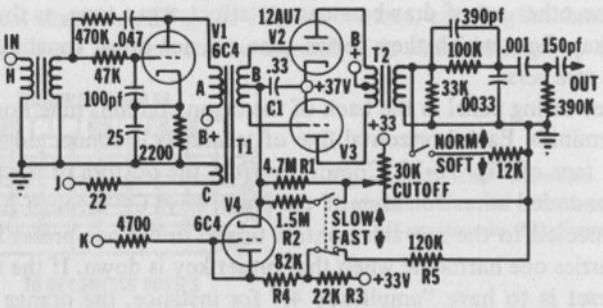


Fig. 16-13–This is the electronic circuitry required to produce the Hammond percussion.

at the V2-V3 grids, is the result not only of what is applied from the cutoff control, but of V4 diode current passing from ground (bottom of the control) through R1-R2 to cathode. This adds to the no-V4 voltage, so that the amplifier grids are at about +25 volts, which is enough to make them conduct normally and pass audio signal to the output transformer. As things stand, then, there is about 25 volts at the transformer centertap (and the tube grids) whenever the percussion is turned on and no organ keys are being pressed.

This is a good time to notice C1, whose left plate in the diagram is at about +25 volts and the right at +37, meaning that it is so charged that there are excess electrons on the left plate. It is actually charged to the difference voltage–about 12 volts–which represents the grid-cathode voltage for normal operation of the push-pull amplifier.

When one or more upper-manual keys are pressed, the corresponding audio tones go to terminal H, through V1 and the push-pull amplifier to T2 and thence to the percussion output, which contains a high-pass filter.

At the same time as the keys are pressed, a genie, whom we will unmask a few paragraphs from now, grounds terminal K through an insignificantly small resistance (and the 4700 ohms). This immediately removes most of the positive voltage from the plate of diode V4. V4 stops conducting and effectively disappears. This puts matters back at the point when we adjusted the cutoff control–the voltage at its arm is about 15, which would be low enough to stop V2-V3 from conducting.

At this moment, however, C1 comes to the rescue. At the instant the key was pressed, C1 was charged as we stated. Now the voltage applied to its left plate decreases, and additional electrons must travel to its left plate. They can only do this through R1-R2. R1-R2 and C1 are thus a time-constant circuit and the change of voltage at the transformer centertap is controlled by the rate at which C1 can alter its charge. So the tube grids do not stop conduction immediately. They do so gradually as the C1 charge changes, and the audio transmission through the amplifier decreases likewise gradually, until finally C1 has acquired its new charge and the grids have reached cutoff.

If the key or keys are held down, grounding the plate of V4, until cutoff, the action is just as described. Pressing any further keys can cause no additional action or any sound. Another note can be "percussed" only if all keys are first released so that V4 can again come into action and restore grid voltage to its conducting value. Restoration takes place relatively quickly, since this change in the C1 charge takes place through R3-R4, which are relatively small. The circuit is now ready to go again.

An important and distinguishing point about the Hammond percussion is this necessity for the circuit to recover after a note is played before it can produce another percussive attack. Hammond originated electronic percussion, which is true percussion rather than the sustain effect more common today. However, it has always worked just as we have described, which means that the percussion must always be played in a "detached" fashion–each note or chord released before the next is struck. This is cheap, but not good. An organist who feels percussion is important will prefer the much more elaborate circuitry of other percussions or even, perhaps, a sustain system. I should add that it is possible to make a percussion which is almost as cheap as Hammond's but can be played *legato,* producing a new percussive strike when a new note is played even with the original notes still held. Its drawback is that each new percussive strike sounds all keys being held, not just the newest note played, and the system is thus still a compromise compared to percussions having circuitry for each note, though a great improvement over Hammond's. Schober Organs had it, back in the tube days.

Especially with tubes, thump is always a problem with electronic keyers when you cannot slow up the attack because you want it to sound percussive. For most practical purposes the push-pull stage of Fig. 16-13 could be single-ended and both transformers eliminated. However, in a single-ended stage, the fast onrush of plate current at the time of circuit recovery would cause a thump in the output. With the push-pull stage, the resumption of plate current in both tubes occurs in the same time in the same direction and about the same quantity. They therefore cancel in the output transformer primary. This is an old Hammond trick, used earlier in the keyer stage of the Solovox, and still used in the pedal solo units of modern organs.

Percussion Switching

The switching system for the percussion involves four tablets and an extra contact under one of the preset keys. Two of the tablets—Soft-Normal and Fast-Slow decay—appear in Fig. 16-13.

The Soft-Normal tablet switch simply shunts additional resistance across the output transformer secondary to reduce the volume for the Soft setting. There is also another contact, not shown, which we shall discuss in a moment.

The decay-rate tablet determines whether R1 is in the circuit alone or is shunted by R2. Since the charge on C1 changes at a rate determined by its value and that of R1-R2 when a key is held down, and this therefore determines how fast the sound fades away, shunting R2 across R1 with the switch in the Fast (down) position makes the resistance smaller, reduces the time constant, and causes faster fade.

With the smaller R1-R2 value switched in (in Fast position) the on voltage applied to the grids is somewhat smaller because the drop caused by the diode current is less; this causes less tube conduction. To restore conduction—and volume—to normal, another contact on the switch removes R5 from the circuit. When in place, R5 acted as a voltage-divider shunt, reducing the voltage at the junction of R3 and R4. Its removal raises this voltage and the V4 plate voltage, and thus restores the V2-V3 grid voltage to the standard value.

Fig. 16-14 shows the rest of the percussion switching. It looks messy, but we will dissect it gently. First a description of salient features.

You will recall that nine busses run the length of the keyboard. As any key is pressed, its contacts place the fundamental and harmonics of that note on the busses. When one of the reverse-color A# or B preset keys is pressed down, its contacts, identical to those of a playing key, pick the harmonics off the nine busses and place them on drawbar contacts, as shown in Fig. 16-12. The drawbar

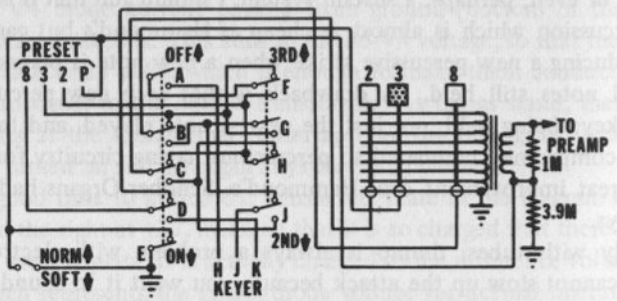


Fig. 16-14—The percussion switching system looks complicated, but it will clear up with a little trail following according to the text.

positions determine which of eight taps of a transformer primary (or ground) each harmonic connects to, to determine its relative volume. There are two sets of drawbars for one manual, and pressing the A# or B preset key determines which set of drawbars controls the manual.

Both sets of drawbars for the lower manual work in exactly this way, and so

does the set of drawbars selected by the A# preset key for the upper. However, the upper-manual B preset is tied in with the percussion system. Percussion works only when that upper B preset is in operation; and it is that B preset and its contacts which we shall discuss exclusively in this section.

Five of the nine contacts under the B preset are connected just like all those of the three remaining drawbar presets of the organ. However, those contacts under the B preset key which touch the busses carrying 8th, 3rd, and 2nd harmonics do not. They are shown in Fig. 16-14 labeled Preset. The wire ends shown, labeled 8, 3, and 2, are connected to the movable contacts under the upper-manual B preset for the three harmonics indicated. Forget about the wire labeled P for the moment.

The drawing also shows two tablet-operated switches, one for percussion off-on and the other, labeled 3rd-2nd, for determining whether the tone passed through the percussion keyer shall be taken from the second-harmonic or the third-harmonic manual keying bus. Only one of these two tones can be used at a time, and that harmonic constitutes the entire percussion signal. Each of the switches has several sections, each with a moving arm, some with both top and bottom contacts, and some with only one or the other. We have labeled each switch section with a letter. As shown, both sets of switch arms are in the up position (just for drawing purposes—not physically in the organ); the first switch is at Off and the second one at 3rd.

The first requirement is that if the percussion is turned off the organ must operate normally. To see how it does, begin at the preset wire labeled 2. This is the wire attached to the movable contact under the B preset which strikes the 2nd harmonic keying bus.

The 2 wire goes, among other places, to the top contact of switch section C. The arm of section C is connected to the 2nd harmonic drawbar. This drawbar is thus connected to the corresponding contact under the B preset in exactly the usual way, except that the connection is made through a percussion-control tablet. (We have shown only the three drawbars for the harmonics under discussion, as the rest are connected directly to the preset contacts as normal.

The 3rd harmonic wire from the B preset goes to the top contact of switch section B, the arm of which goes to the 3rd harmonic drawbar. And the 8th harmonic preset wire goes to the arm of section D, the top contact of which takes the signal to the 8th harmonic drawbar. So all is normal.

Now let us turn the percussion on, meaning that the arms of all five switch sections of the On-Off switch move down. Let us leave the harmonic selector tablet in the 3rd position, however, as shown in Fig. 16-14.

The 2 preset wire no longer goes to the 2nd harmonic drawbar in the same way as it did before, since the arm of section C is now down, touching its bottom contact. However, that bottom contact goes to the upper (and only) contact of section H, and the arm of section H does go to the preset 2nd harmonic contact; so 2nd harmonic tone continues to go to its drawbar.

Tone from the 3rd harmonic preset contact, which is connected to the top contact of section B, no longer goes to its drawbar through section B, as the arm

of B is now down. The 3rd harmonic signal goes to the top contact of section F. The arm of F takes this signal through the closed contacts of section A to terminal H. This is the same terminal H as is shown in Fig. 16-13, and it means that the 3rd harmonic signal is now the input signal for the percussion circuit.

Now take a look at winding C on transformer T1 in Fig. 16-13. A small amount of the signal amplified by V1 appears in this winding; in this case, it is, of course, 3rd harmonic signal from the B preset contact. That signal goes through terminal J, which is shown in both drawings, and goes to the arm of switch section J (Fig. 16-14). The signal is picked up by the top contact of section J and goes to the bottom contact of section B. Since the B arm is down, it takes the signal to the 3rd harmonic drawbar, so that, again, all the drawbars are supplied normal signals.

By this time you should have a good notion of what these two switches do, though I suspect it would take any of us more than a few minutes to devise the switching circuit, starting only with the idea of the functions to be fulfilled. Assume that the second switch is in the 2nd position now, and trace the connections to see how the 2nd and 3rd harmonic signals get to their drawbars while furnishing the percussion keyer 2nd harmonic signal.

What of the 8th harmonic? It is sacrificed when percussion is in use. Remember that terminal K in Fig. 16-13 had to be grounded whenever a key was pressed to make the keyer work. It is connected to the 8th harmonic keying bus so that when any key is pressed terminal K is connected to one of the tone-generator coils. These have quite small resistances and for d.c. control purposes terminal K may then be considered grounded. However, it is not possible to use the 8th harmonic keying bus both for control purposes and to supply tone to the drawbar.

The mechanism is simple enough. Preset wire 8 goes to the arm of switch section D. With the percussion switch at Off, the top contact of section D takes the 8th harmonic signal to its drawbar. With the switch at On and the arm down in the drawing, the bottom contact of section D takes the 8th harmonic B preset contact, and thus the keying bus which it is touching, to terminal K in Fig. 16-13, grounding terminal K through the small resistance of one or more generators when the manual is played.

There is one more switching function. When percussion is in use, it produces an extra signal which is mixed into the total output as we shall see later, making the total volume of the upper manual higher than normal. In Fig. 16-14 we see the same transformer as in Fig. 16-12, this time with its secondary, one end of which is normally grounded and the other end feeding preamplifier circuits, etc. There is, however, a 3.9-megohm resistor between the bottom of the transformer and ground. With this resistor in place (and the secondary bottom ungrounded), signal from the transformer is much attenuated.

The B preset for the upper manual has one physical peculiarity. In addition to its nine regular contacts, it has a tenth. This contact (shown as preset contact P in the diagram) is normally closed; and when closed it shorts the 3.9-megohm resistor by connecting the transformer bottom to ground so that the upper-

manual output from the drawbar system is normal. However, when the B preset key is down, it opens. If the percussion switch is on, the arm of section E goes down and its upper contact, to which P is also connected, is removed from ground. Thus, when preset key B is down *and* percussion is on, the short across the 3.9-megohm resistor is removed and the drawbar system produces less volume so that the percussion will stand out and the total volume will not be greater than normal.

P is also connected to an extra contact on the Soft-Normal switch. When percussion is made soft, the volume-reducing resistor is shorted so that the drawbar-system volume comes back up to normal and again the total volume is approximately as usual.

Main Pedal Switching

Pedal tones are derived from two different sources. The first is the main tonewheel generator and the second is a pedal solo unit, described in the next section.

Each of the pedals has nine contacts under it, very similar in construction to those of the manuals. Two of these contacts are used for the pedal solo system; the remaining seven are used for the main pedal tones. Fig. 16-15 shows the idea.

In addition to the fundamental, six harmonics are used: 2nd, 3rd, 4th, 6th, 8th, and 10th. Each has its own switch arm, but those for fundamental and 3rd

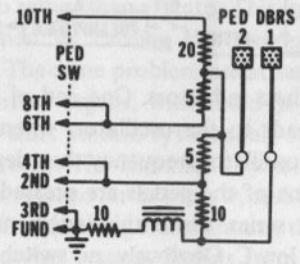


Fig. 16-15—Seven of the nine pedal-switch arms supply fixed harmonic mix from main generator to two drawbars.

are connected together, as are those for 2nd and 4th, and for 6th and 8th. As the diagram shows, the tones are fed to a simple network (one network for the entire system) from which two outputs are derived. One, fed to the first pedal drawbar, emphasizes the lower harmonics, and tones fed to the second drawbar have emphasized upper harmonics. The inductive filter serves to remove any higher harmonics or transients that may be present in the lower frequencies.

There are in the organ two tapped transformers like the one shown in Fig. 16-12, one for the upper manual and one for the lower. The two pedal drawbars are placed with those of the lower manual. They operate in just the same way as other drawbars, so that the organist may set up pedal tone in varying loudness values and in either of two kinds of tone (or both).

Pedal Solo Unit

A second source of pedal tone is the pedal solo unit, which is, in effect. a completely separate musical instrument. It is extremely like the old Hammond

Solovox, and if you are interested in a full diagram of that device, you will find it in the second edition of this book. Since this is a teaching text and not a service manual, I do not believe a full diagram worth the space it takes, especially since I believe this unit to be almost an anachronism today. We shall make do with the block representation of Fig. 16-16 and a brief explanation.

The basis of the system is a tone generator consisting of a master oscillator and five frequency dividers. The oscillator is variable over a 32-note range, its lowest note being the low C of Octave 5 in Table 1-1. The dividers are aperiodic and each provides a note an octave lower than that fed to it, whatever that note may be. The fifth divider, therefore, has a low output one octave lower than the lowest 16′ C, that is, 16.35 cps, and what it gives is the 32′ pitch register.

The manner in which the master oscillator is tuned over its range is shown by the tuning contacts in Fig. 16-16. Thirty-two inductors are connected in series and between each pair is a moving contact which touches a grounded bus when the corresponding pedal is pressed. In the diagram we show only the two lowest

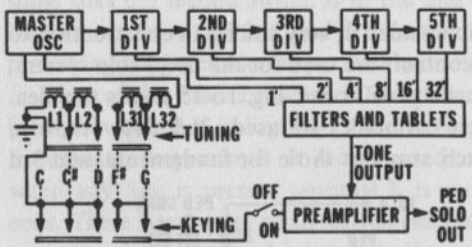


Fig. 16-16—Block diagram of the pedal solo unit.

and two highest inductors. One end of the series inductor rank is grounded and the other leads to the oscillator, where the inductance forms part of the tank circuit. The oscillator frequency thus depends on the net inductance value.

When none of the pedals are pressed, the net inductance is the sum of all 32 inductors in series. Since this is the maximum possible value, the note so produced is the low C. Obviously, no switch is necessary to provide this, so no low C switch is furnished. When the C# pedal is pressed and the C# switch closes, the junction of L1 and L2 is grounded, reducing the net inductance and raising oscillator frequency a half-tone. The same thing happens right up the line, until the top pedal G, grounding the junction of L31 and L32, leaves only L32 in the circuit and produces the highest note. The system makes it impossible to produce odd frequencies, as would a simple system in which each pedal separately placed an inductor from oscillator to ground.

Each of the frequency dividers is a flipflop (of a rather antiquated 3-triode design), and the output of each is fed to an R-C tone-coloring filter. There are two 32′ filters, one for Bombarde and the other for Bourdon. There is no separate 1′ filter, this tone being added to the 2′. At each filter output there is a normally open series switch controlled by a stop tablet, and the tones appearing on the output contacts of the switches are commoned and fed to the preamplifier input. There is also a Mute tablet which adds a simple treble-cut filter across the preamplifier input to slope off upper harmonics when desired.

The preamplifier consists of a triode voltage amplifier feeding a transformer and a push-pull second amplifier feeding a second transformer. The latter stage is also used for keying control. The grids of the push-pull stage are normally biased off by a negative voltage fed to the centertap of the first transformer secondary through an R-C time-constant network. When no pedals are pressed, the unit is therefore silent even though the oscillator and dividers are producing the signal of the low C. When a pedal is pressed (and the on-off switch is also on), one of the keying contacts shown in Fig. 16-16 grounds a bus connected through the on-off switch to a point where it shorts and removes the negative off bias of the push-pull stage. This causes the stage to conduct and transmit tone. The R-C network in the bias supply causes a pleasantly gradual attack and decay. The use of a push-pull stage for keying avoids thumps for the same reason one does in the percussion.

The output of the preamplifier is connected to ground and to tap 4 on one of the drawbar transformers. The hot side of the line includes a series variable resistor for presetting the volume level.

The Hammond Vibrato

The original Hammond organs had a problem with vibrato, since the tone generation system is a group of frequencies fixed by the rotational speed of the motor and these frequencies would be extremely hard to vary. As a result, Hammond organs for some years had tremolo rather than vibrato. This much inferior amplitude-modulation effect was obtained by causing the running motor to vary a rheostat in series with the audio. The same problem has arisen with other instruments such as the reed-type Wurlitzer organs, with fixed-frequency generators. All such organs now use phase-shift vibrato systems, all of them with the exception of Hammond being electronic. The Hammond system is electromechanical. It was the first phase-shift vibrato, so far as I know, and is unique.

Fig. 16-17 shows the secondaries of the matching transformers, the primaries of which are tapped for use with the drawbar system. Signals from these windings are passed to a pair of switches, each of which determines whether the signals from that manual shall be vibrated or not. In the off position of each switch, the signal goes through a preliminary one-stage voltage amplifier and then on through the preamplifier, acquiring no vibrato. In the on position, the switch passes the signal through the vibrato system before it goes to the preamplifier. The first element of the vibrato system is a fairly ordinary two-stage "vibrato amplifier," whose only distinguishing feature is its cathode-follower output.

The heart of the vibrato system is a lumped-constant delay line consisting of 18 low-pass filter sections. The signal from the vibrato amplifier is fed into the start of this line and is shifted in phase somewhat by each filter section, so that if signal is taken from the line at various taps, each tapped-off signal is phase retarded with respect to the previous one. A rotating scanner consists of 16 sets of fixed capacitor plates equally spaced around a circle, and a single set of rotating plates which pass through the air gaps between the fixed plates. Each stack of fixed plates is connected to a tap on the line, so that when the rotor is driven

(by the organ's generator motor), the signals picked up by the rotor are constantly changing in phase. The constant phase change is equivalent to a frequency change; and when the rotor is connected to the organ preamplifier, organ signals with vibrato are carried by the preamplifier.

There is a total of 16 taps on the delay line. To obtain the widest vibrato, the stator plates are connected so that the entire line is scanned from beginning to end and back again. As the rotor goes around, it picks up signals from these taps in this order: 1-2-4-7-10-13-14-15-16-15-14-13-10-7-4-2. On the way from 1 to 16 it decreases the frequency, and from 16 back to 1 the frequency increases by a similar amount, so that average frequency remains the same as it would without vibrato. The line is scanned from start to finish in about 1/14 second and the maximum frequency change is about 1½% (¼ semitone) in each direction.

The amount of vibrato and its effect are controlled by a knob on the console; there are six positions. Three of these positions simply control vibrato depth by determining how much of the line is to be scanned in the 1/14 second. As described above, the maximum vibrato position, labeled V3 on the switch, scans the whole line, skipping some taps along the way to do it. When the switch is in the V1 position, minimum vibrato depth is obtained by switching the taps to the stator plates in such a way that one revolution of the rotor picks up signals from these taps in this order: 1-2-3-4-5-6-7-8-9-8-7-6-5-4-3-2. That is, it scans only a part of the line, in more closely spaced steps. In the V2 position, as you might expect, more of the line, but not all, is scanned for an intermediate effect.

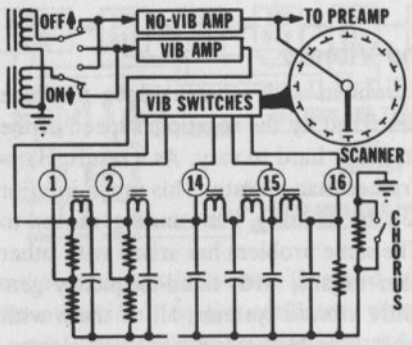


Fig. 16-17—Outline of the vibrato system.

The three remaining switch positions, C1, C2, and C3, produce Hammond's "vibrato chorus" effect. This consists of the regular vibrato (the same switching as in positions V1, V2, and V3) plus opening the switch labeled Chorus in Fig. 16-17. The effect is to interpose a resistor between the bottom of the filter network and ground, across which appears a net signal without phase change. A part of this constant-phase signal appears at each line tap and is picked up by the rotor, so that the signal from rotor to preamplifier contains both vibrated and unvibrated signal.

The preamplifier proper is a three-stage affair which includes the swell-shoe volume control, a Soft-Normal tablet for two-step volume control, and a tone control. The swell shoe operates a capacitive control rather than a potentiometer. There are two adjacent sets of stator plates. One is connected to the plate of a previous stage and the other to an R-C network at whose terminus volume is minimum and tonal balance is such that the lows are emphasized. A set of moving plates travels between the two stators, its position governed by the shoe. The volume and tonal balance of the signal on the rotor, which is connected to the grid of the following tube, depends on how much it is picking up from each stator.

The Soft-Normal tablet operates a switch that shorts a series resistor for maximum volume. The "tone control" is actually a volume preset potentiometer. The preamplifier output is obtained from a transformer and is push-pull with a grounded centertap.

Sound Production

The amplifier chassis for this organ contains three separate amplifiers; the common amplifier power supply is on a separate chassis. All the amplifiers have push-pull 6BQ5's in the output stages. One of the amplifiers feeds treble to an 8-inch speaker and another bass to a pair of 12-inchers, the frequency break coming in the area of 400 cps as the result of series and shunt capacitors in the two amplifiers. The third amplifier is entirely for treble sound with reverberation, which is furnished by one of Hammond's smaller spring units. Bass reverberation is also available to some extent, but this is added through the bass amplifier itself. All this equipment is in the console, and there is the usual connector for adding external tone cabinets if they are wanted. High-frequency response of the audio system is not extended. The highest sound available from the organ is slightly under 6 kc, so there is no need for response beyond that point. In fact, if "better" response were available it would be detrimental, as it would merely accentuate the transients traditionally caused in the organ by keying sine waves. This is an important point to remember should you ever wish to add audio equipment not born in a Hammond factory to a Hammond organ. Leslie systems sold by Hammond dealers for Hammond organs are, of course, designed with this in mind.

Chapter 17

Rodgers Organs

The Rodgers Organ Company, 1300 N. E. 25th Ave., Hillsboro, Ore., is one of the younger firms in the organ industry and an unusual one as well. It makes almost no instruments with fewer than three manuals or for a retail price of less than around $4500 and is not in direct competition with companies whose chief clientele is the "popular" market, such as Thomas, Lowrey, *et al.* Its nearest competitor is Allen; and, like Allen, Rodgers' approach to organ design is prompted by an interest in the subtler and more elaborate approaches to pipe tone. This orientation is not surprising to anyone who has met and talked with Rodgers people, many of whom are excellent organists in both serious and theatrical styles and are deeply interested in the organ as a medium of artistic expression. A number of them came from Techtronix (including the company's head, Rodgers Jenkins), probably this country's leading manufacturer of premium quality oscilloscopes, near which Rodgers headquarters is located.

Rodgers makes both classic and theatre organs. While there are standard models, each instrument more often than not has some special alterations or additions, a customizing procedure that the company's relatively small production of relatively expensive organs makes possible. Fig. 17-1 shows the Custom Theatre Organ, Model 36-E. The number, in Rodgers' standard coding system, means three manuals, six ranks of generators (usually plus a celeste rank), "entertainment" style. (Evidently classic music is not considered entertaining.)

Fig. 17-2 is a photo of the 32-B, a classic instrument which we shall describe in this chapter. It has two generator ranks plus an abbreviated celeste rank, 38 speaking stops, and three couplers. Like all Rodgers organs, it employs various ways of doing different things and thus cannot instantly be placed in a simple classification, except to say that all oscillators are individual and there are no frequency dividers. In describing it, we shall divide up the functions, and the result will be a little like a description of two or three organs in a single case.

As this is the first three-manual organ we are describing, let me remind you that the lowest manual is the Choir, the center one the Great, and the upper is the Swell.

The Unit Flute

Most stops of the Rodgers 32-B are obtained by unification and duplexing. The meanings of these terms were discussed briefly on pages 95 and 96. which you might reread at this time if you have forgotten. The terms, fairly abstract up to now, will be made quite clear in this Rodgers chapter. The one point we might add before plunging in is that a rank of pipes which is unified and duplexed so as to act as the source of stops of several pitches and on more than one keyboard is often known as a *unit rank.* A rank of flute pipes used like this would be called a *unit flute.*

Of the two principal ranks of generators in this Rodgers organ, one is a unit flute which furnishes 85 tones in Octaves 2 through 8, Table 1-1, plus a top C taken from Octave 9. It furnishes tone for flute stops at all octave-sounding



Fig. 17-1—The Rodgers 36-E Theatre Organ, a model which has 6 ranks plus a celeste.

pitches from 16′ through 1′ and mutation pitches $2\frac{2}{3}'$ and $1\frac{3}{5}'$ for the Great and Choir manuals. Where the pitches run out at the top or bottom of the 16′, 1′, and $1\frac{3}{5}'$ registers, the pitches "break back" to the previous octave.

The unit flute in the Rodgers organ is exactly equivalent to a unit flute rank in a pipe organ. Each pitch is generated by an oscillator which is normally off. When the appropriate key is pressed, the oscillator is furnished B-voltage and goes on. The outputs of all 85 oscillators are permanently channeled through amplifiers, etc., to the speaker system, and stop switching merely involves channeling B– through the correct key switches for the pitch register required. The only difference between this and a unit flute pipe rank is that for certain stops less than normal B-voltage can be furnished through the keys to the oscillators so that the output volume is softer than normal, thus giving one kind of "tonal change."

It is obvious that the unit system has some disadvantages. Plainest is that there can be no difference in voicing among the stops concerned, since they all come from the same sine-wave sources, through the same audio path. All differences involve only pitch register and loudness. The second major disadvantage is somewhat more subtle. It is that once an oscillator is keyed, from whatever source, it can do nothing further. Suppose we turn on two flute stops, an 8′ and a 4′. If we press middle A key, we will hear A-440, the 8′ tone, plus A-880, the 4′ tone. If we keep this key down and also press the A key an octave lower, we will hear the added A-220 8′ tone, but the A-440 which is the 4′ tone from the lower-octave key is already sounding as the 8′ output for middle A, so the 440 cps which should also be heard as the 4′ pitch of the lower key adds nothing. If,



Fig. 17-2–This chapter describes the 32-B church organ, one of the "lower-priced" models.

on the other hand, there were a rank of generators for each stop, the original 8′ 440 would be augmented by the 4′ 440, both in loudness and in the slight accidental pitch interference produced by the additional generator. Even in single-generator-rank electronic organs of more common types, pressing the lower A key would at least augment the volume of the 440. But in the unified scheme, the 440 pitch shows no evidence at all that the 4′ stop is on. In other words, where two stops and two keys should produce four results, they produce only

three. The effect is actually like that of an intramanual coupler working only on flute stops, rather than like that of a genuine second stop.

Why, then, does a serious organ maker use the unification principle? The reply comes from me, not from Rodgers, but it seems rather obvious. Rodgers is interested in the authenticity which requires (a) a gradual attack and decay for each note, (b) independently tuned sources of tone rather than frequency dividers, and (c) perfect scaling. These advantages can be obtained in no other reasonable way.

There are 12 stops on the organ which use this generator, one of which keys two pitch registers at once. In a "straight" organ this would require 13 ranks of 61 oscillators each, a total of 793 oscillators just for the flutes. The job could be done instead with a single rank of 85 oscillators constantly running instead of being powered by the key switches, but to achieve the attack and decay delay would require electronic (diode or transistor) keying. This would mean at least 12 ranks of 61 keyers each, or 732 keyers, each with a number of parts. Again we are dealing with a highly expensive and space consuming quantity. But since the waveshape is sine and an electronic keyer of any reasonable type would clip the wave during rise and fall, the transients would be intolerable (except with very fast attacks and decays) and electronic audio keying is not practical anyway. The multiple-rank approach is not actually impossible, of course, and Allen has used it in various degrees. However, Rodgers does not ordinarily make instruments as expensive as many Allens have been, nor do they make a practice of locating racks of components outside the console.

Fig. 17-3 is one of the 85 generators of the unit flute. It is a simple oscillator in which the transistor is a grounded-base amplifier, with feedback from collect-

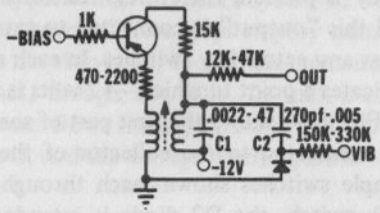


Fig. 17-3–One of the 85 oscillators of the unit flute rank.

or to emitter through a transformer. The primary of the transformer is the inductor of the tank circuit. The coil has a ferrite cup core. C1 is the normal tank capacitor, and the frequency is adjusted by rotating the top of the cup core (see page 48).

Vibrato, which Rodgers calls "tremulant" but is actually frequency shake, is supplied with the aid of the diode and C2. With vibrato off, a small steady negative d.c. is applied to the vibrato terminal. This causes the diode to conduct, but not fully, so that C2 is connected to ground effectively through a resistance and thus affects frequency less than it would if the diode were shorted. When vibrato is turned on, a vibrato-frequency sine wave is superimposed on the d.c. so that the diode alternately conducts to a greater and lesser degree than normal and oscillator frequency rides up and down equally from its resting value. Vibrato

speed and depth can be adjusted internally in the vibrato generator, which is an oscillator like that of Fig. 17-3 plus amplifiers, but the only external control is an on-off tablet.

Outputs of all 85 flute oscillators go to the unit flute preamplifier, thence to the organ output through a swell-shoe control.

Unit Flute Keying

Each flute oscillator is normally off because the −12 volts shown in Fig. 17-3 is missing. There is a single normally open switch under each manual key, which is used for all organ keying. One of the functions of the key switches of the Great and Choir manuals is to cause −12 volts to be supplied to flute oscillators of the correct pitches, depending on which flute stops are turned on and thus at what pitch register(s) the flutes are to be keyed. An electronic keying

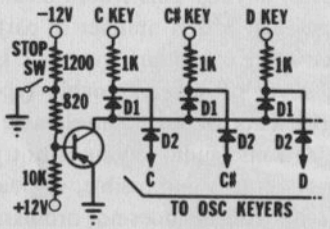


Fig. 17-4—Each stop employs one of these on-off circuits, with a connection to every key.

system is used, even though the keyers are keying d.c. for the oscillator rather than audio, because other things must sometimes happen when flutes are keyed, such as air noise, chiff, and sustain action for the Harp and Carillon.

Note well: All key switches are connected to a bus carrying −12 volts. When a key is pressed, therefore, −12 volts is placed on the "output" of the switch, and this "output" is connected to many keying circuits. We shall not, therefore, show any actual key switches. In each applicable diagram, a terminal labeled Key indicates a point to which −12 volts is applied when a key is pressed.

Fig. 17-4 shows the first part of some flute keyers. All of the 61 key switches are connected to the collector of the transistor in the same way as the three sample switches shown—each through a 1,000-ohm resistor and diode D1. For each switch, the D2 diode is intended to pass −12 volts to following circuits when the stop is on and a key is pressed. Note that *for each stop* all the components shown in Fig. 17-4 exist—one transistor, with its three bias resistors and the stop switch, plus 61 resistors, D1's, and D2's.

When the stop switch is open (stop off), on bias is applied to the transistor base, so that whenever B− is applied to its collector it will conduct fully. If, in this case, a key is pressed, the −12 volts from the key switch passes through the 1,000-ohm resistor and D1 to the collector. Since the transistor now conducts fully, it effectively shorts the −12 volts to ground so that the junction of the resistor and D1, being practically at ground potential, can pass no voltage through D2 to the following circuitry.

When the stop switch closes, −12 volts can no longer be applied to the transistor base; in fact, only +12 volts is applied (through 10,000 ohms), so that the

transistor is entirely cut off. Now when a key is pressed, its −12 volts finds the collector floating—effectively not connected to anything except the other D1's—so the −12 volts is passed on through D2.

The D1's prevent negative voltage placed on the floating collector from backing up into the D2's of other keys which have not been played. The 1,000-ohm resistors are for current limiting, so that when the stop is off, the power supply will not be shorted to ground if a key is pressed, and unlimited current will not flow through the transistor.

As you will see, there is a single keying circuit associated with each of the 85 flute oscillators, which is energized by the −12 volts supplied through D2. If Fig. 17-4 is the circuitry for an 8′ flute stop and the C key shown is middle C, then the D2 for that key is connected to the keying circuit for the 262-cps oscillator, etc. If Fig. 17-4 were for a 4′ stop and the C shown still the middle C key, its D2 would be connected to the 523-cps oscillator keying circuit—and so on. It is obvious, therefore, that there are several D2's going to each of the oscillator keying circuits, one for each flute stop. This is the reason for the existence of the D2's—so that −12 volts supplied to a particular oscillator keyer for one stop cannot back up the line into the Fig. 17-4 circuit for another stop which may be turned off, and thus be shorted to ground. This business of supplying voltage to one point from several possible sources through a diode from each source to prevent voltage going from one source to another is common. However, Rodgers is the first I have seen apply a name to it. They call them *contact multiplying* diodes. The term is apt, for the alternative would be a separate switch from each source, often resulting in piled-high switch stacks.

One modification is sometimes made in the arrangement of Fig. 17-4. For those stops which are to sound more softly, less than the full keyed −12 volts must be supplied to the following oscillator keyers. Those stops employ transistors with base bias circuits in which, when the stop is on, enough bias remains so that the transistor still conducts somewhat, though not fully. In such a case, when a key is pressed some collector current flows through D1 and the 1,000-ohm resistor, causing a voltage drop across it, and the voltage reaching D2 is −12 less the drop across the resistor.

Flute Oscillator Keyers

The −12 volts coming through the D2's of Fig. 17-4, or the lesser voltage for soft stops, could simply be applied to the −12 points in the oscillators (Fig. 17-3) to turn them on, and this would make a suitable keying system. However, more complications exist, so the circuitry of Fig. 17-5 is needed. These components exist for each of the 85 oscillators, except for the components below the dashed line.

Since there are eight Choir stops using the unit flute generator, one of which, a Clarinet, is made by keying both 8′ and 2⅔′ pitches at the same time, there are nine Choir D2's from keyers of the type of Fig. 17-4 which are connected to each oscillator keyer of Fig. 17-5 at the point marked From Choir D2's. When any of these stops is turned on and a key switch appropriate for the oscillator

associated with the components in Fig. 17-5 is pressed, the resulting negative voltage passes through R1 and D3 to the oscillator, turning it on. Since C1 is between the oscillator B– input and ground, it must charge before full B– can exist across the oscillator, and this causes a small delay in tonal buildup, giving the gradual attack typical of pipes. When the key is released, voltage remains on the oscillator until C1 discharges, giving a gradual decay as well. C1 is graded from top to bottom of the oscillator rank for the delays appropriate for various pitches–greater (larger capacitor) at the bass end.

There are four Great flute stops and thus four Fig. 17-4 groups for that manual. Four D2's are therefore connected to the Great input in Fig. 17-5, and these key the oscillators on and off in the same way.

The effects which make all the components of Fig. 17-5 necessary are *sustain, air noise,* and *chiff.*

Chiff is a transient effect which occurs when air is admitted to an ordinary flue pipe. It causes certain harmonics to be heard briefly prior to the fundamental. The subjective effect is very much like a *ch* sound at the start of each note.

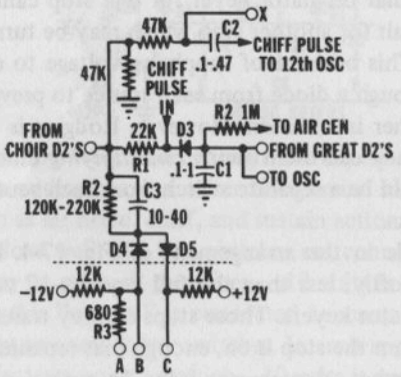


Fig. 17-5–The actual keyer circuit which is associated with each flute oscillator.

Chiff is actually a defect in pipe speech, and it is almost always removed by a process of cutting nicks into the languids of the pipes.* However, some ranks of pipes on some organs are deliberately left unnicked because the *ch* sound can sometimes be useful to give a sort of staccato accent to certain passages in polyphonic music. Use of chiff should, in the opinion of most good musicians, be sparing, and those organs with several (or even all!) ranks of unnicked pipes which chiff away like mad can be very wearing indeed to listen to.

The chiff effect is obtained in the Rodgers organ by passing the negative keying voltage from the Choir D2's through C2 to the oscillator a musical 12th (an octave and a half, corresponding to approximately the third harmonic) above the note being keyed. This passes a pulse through C2 at the instant of keying, so that the higher-frequency oscillator is energized and sounds very briefly. A diode

*For a discussion of chiff and for general background on pipe organs, see William H. Barnes: "The Contemporary American Organ," 8th edition, Fischer & Bro., Glen Rock, N. J., 1964. Chiff is covered on page 93. Take Barnes' material on electronic organs with several grains of salt.

control system is connected to point X so that effectively this point is usually grounded, preventing any pulse from passing through C2. When the Choir Quintade 8′ stop is turned on, this ground is removed, and the chiff is heard for all Choir flute voices. Great flute stops cannot produce chiff, even with the Quintade on, since Great keying voltage cannot pass through D3 to C2.

Some people feel that greater realism is achieved when some air noise accompanies organ tones. In this organ a white-noise generator (a zener diode plus amplifier) supplies hiss which goes to a gating transistor, normally off. When keys are pressed, negative keying voltage is applied through R2 to the gate, turning it on slightly. The more keys being pressed, the more negative voltage is transmitted to the gate through the R2's for various oscillator keyers. Maximum hiss output to the preamplifiers and speakers is reached when about 10 keys are down. In my own opinion, this kind of "air noise" is a defect if heard in a pipe organ and has no place where it must be created artificially. The effect is about as legitimate as putting ants on the dining room table to simulate the atmosphere of a picnic. This comment does not necessarily apply to the Allen "whind" effect, which does something quite different.

The sustain effect is used on this organ to create the Harp and Carillon. For the Harp, the stop switch turns on the Great Flute 4′ and Piccolo 2′ plus short sustain. For the Carillon the switch turns on the long sustain and flute stops of pitches which will give tones of the fundamental pitch, a musical third lowered one octave, the 12th (approximate third harmonic), an octave above fundamental, and two octaves and a fourth above fundamental. For the middle C key, for instance, these frequencies are 261.6, 164.8, 784, 523.3, and 1397 cps.

C3 in Fig. 17-5 gives the sustain. Each oscillator keyer has one of these, with graded values, plus D4 and D5. Components below the dashed line serve for all keyers.

As shown, with no connections at points A, B, and C, the +12 volts blocks D5 and the −12 volts blocks D4. These diodes, therefore, cannot conduct at any time, even when keying voltage from one or more D2's is applied to the D2 input point. Suppose, however, that we ground points B and C. This means that the bottoms of both diodes are now at ground potential and the fixed potential sources are shorted out. Now when a key is pressed, the −12 volts from a D2 quickly charges C3, electrons going down through D4 to ground, thence through the −12-volt supply to the upper plate of C3 as excess electrons. When the key is released and the D2 point floats free, C3 discharges. When it does, the excess electrons on the upper plate can only pass through the oscillator to ground, thence up through D5 to the lower plate. This takes time, and as the discharge electrons pass through the oscillator, they keep the oscillator energized for some time after the key is released, so that the sound has a slow decay. The graded values of R2 also help the capacitor discharge, adjusting its timing for best realism at different pitches. (R2 also discharges C3 if sustain is turned off.) As only a single setup below the dashed line is furnished for the organ, all D4's and D5's are connected to it.

Suppose now we keep point C grounded, so that the capacitor discharges in

the same way as it did during the long sustain, but ground point A instead of B. In this case, C1 can charge only to the difference between the keying potential and that across R3 to ground, since as soon as the lower plate is slightly more positive than the D4 anode, D4 stops conducting and halts the charging process. Now when the key is released C3 discharges and maintains the oscillator in action as before, but since the charge on C3 was less than during long sustain, oscillator output immediately drops to a lower value and takes less time to fade completely. Both the sustain and stop switching are done by the single switch for each sustain stop through a group of "contact multiplying" diodes, and in actuality C is grounded by a reed relay, which is merely an extraneous detail.

Unit Diapason

The main tone generator for the 32-B is a second rank of 85 oscillators which provides tone for all the remaining organ voices, though two voices also include tone from the celeste generator. Fig. 17-6 is a schematic of one of the oscillators.

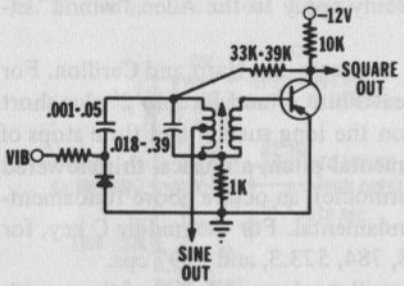


Fig. 17-6—Main-generator oscillators furnish two outputs each.

Vibrato is furnished (from a separate "main" vibrato generator) in the same way as for the flute generators. Each main oscillator provides two outputs, an approximately square wave from the collector and a sine wave from the transformer primary tap. Unlike the flute oscillators, the main generators have B— applied permanently and thus run all the time. It is the audio which is keyed.

Rodgers points out that a half-wave rectified sine wave, with a little low-pass filtering to remove the sharp corners from the clipped-off portion, makes an excellent diapason sound. This was news to me and I confirmed it by experiment. There is therefore associated with the sine-wave output of each oscillator a diode keyer which not only keys the tone on and off but also chops off the positive alternations, giving the desired half-sine wave. The outputs of the 85 diapason keyers are used in the same manner as the outputs of the flute oscillators, as a unit diapason, keyed from the Great manual at 8′, 4′, 2⅔′, and 2′ pitches to give the stops for a diapason chorus: Open Diapason, Octave, Twelfth, and Super Octave. A soft stop in the Choir, the Viola 8′, and a still softer stop, a Dulciana 8′, also use the unit diapason. There is, in addition, a "3-rank" mixture on the Great, using 1′, 1⅓′, and 2′ keying.

For each individual diapason stop there is a preliminary keying circuit exactly like that for the flutes in Fig. 17-4—a transistor and 61 resistors, D1's, and D2's. The resistor from the key switch to the diodes is 68,000 instead of 1,000 ohms.

The negative voltage passed through a D2 when the stop is on and a key is pressed is passed to the circuit of Fig. 17-7, one of the 85 diapason keyers.

D3 is the gate diode. Some negative voltage normally appears on its anode from a source in the diapason preamplifier, to which all 85 keyers are connected. D3 is therefore cut off and signal from the sine-wave output of the generator of

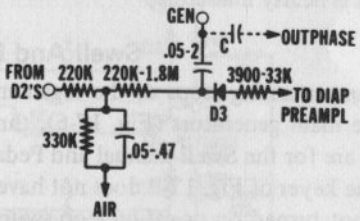


Fig. 17-7—One of the 85 unit diapason keyers.

Fig. 17-6 cannot pass through it. When negative voltage is applied to the cathode of D3 from one or more D2's, the diode conducts. However, the potentials are so proportioned that during the positive alternation of each cycle from the generator the diode blocks. Thus what goes into the diapason preamplifier is only the negative portion of the sine wave.

The negative keying voltage from D2 is also used to operate a gradual gate in the air noise generator. The 330,000-ohm resistor feeds this keying voltage from each of the 85 keyers into the air gate, so the more keys pressed the more negative voltage at the air gate and the louder the air noise up to a maximum of 10 notes.

With a simple diode keyer of this kind, capacitance between the electrodes of the diode causes tone leakage, which becomes greater at high frequencies, fundamental and harmonic. To combat the leakage, Rodgers takes two steps. First, the keyers for higher notes differ from that shown in Fig. 17-7 in that following D3 is a second diode, with a 120-pf capacitor from the junction of the two to ground.

Second, signal from the generator is permanently taken through a second capacitor C in Fig, 17-7 to the preamplifier, where it is inverted in phase and then mixed with the keyed signal input. When the amplitude of the inverted



Fig. 17-8—The printed-circuit capacitors for cancelling the leakage.

signal is the same as that due to leakage through D3 the two cancel. The amplitude of the needed cancellation signal is, of course, very small, and, like the leakage it is to kill, it must pass through a capacitor so that higher frequencies will predominate. Instead of a normal capacitor for C, Rodgers uses a "printed-circuit capacitor." Fig. 17-8 shows a small part of the printed circuit containing the diapason keyers. Each of the pads like the one marked Gen carries signal from the generator transformer tap, and a normal capacitor is connected in the usual way between this pad and D3, per Fig. 17-7. The area marked Pickup, which runs the length of the board, forms the right plates of all the capacitors C

of Fig. 17-7, the Gen pad being the equivalent of the left plate. The Pickup area is then connected to the "outphase" input of the preamplifier. The capacitance between each Gen pad and the Pickup area is quite small, but so is the leakage capacitance through D3, and this scheme does the job with minimum waste of space and money. Cancellation of leakage is not quite complete, but what remains is nearly inaudible.

Swell And Pedal Keying

The remaining stops of the organ are produced from the square-wave outputs of the main generators (Fig. 17-6), through keyers like that of Fig. 17-9. These stops are for the Swell manual and Pedals only.

The keyer of Fig. 17-9 does not have a preliminary transistor network because it is not turned on or off by stop switches. It derives its on voltage directly from the key switch. When the key is up, output from the oscillator transistor collect-

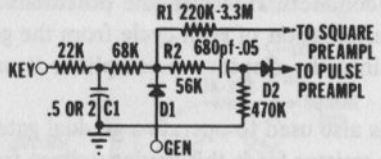


Fig. 17-9—Keyers like this are used for Swell and Pedal.

or is blocked by D1 because the collector carries a negative potential which is placed on the diode anode. When the key is pressed, −12 volts is applied through two resistors to the diode cathode and D1 conducts. Square-wave signal goes directly through R1 to a square preamplifier* which collects and amplifies all the square signals. At the same time, the square-wave input from the generator passes through differentiating network R2-C2 and rectifier D2, so that pulses result and are fed to a single pulse preamplifier. This method of forming pulses is much the same as in Baldwin's sustain keyers (see Fig. 9-13 and associated text).

The Pedal and Swell keyers are identical except that for the pedals the value of C1 is 2 μf instead of 0.5 μf. C1 is the keying delay capacitor, and pedal tones should speak more slowly than those of the manuals. Also, the pedal key switches furnish only −8 volts instead of −12. The reason is that the negative ends of the square waves from these oscillators are somewhat rounded, so that as the tones come on more slowly for the pedals, the slight change of tone color would be apparent. With only 8 volts applied to D1, the gate is not open wide enough to admit the bottoms of the square waves.

There are actually four sets of keyers like these, one each for Swell 8′ and 4′ registers and for Pedal 16′ and 8′. (These two keyboards have stops at only two pitch registers each.) There are thus eight outputs to which tones always go when Swell or Pedal keys are pressed—Swell 8′ Pulse, Swell 8′ Square, Swell 4′ Pulse, Swell 4′ Square, and four more in the same way for the pedals. Each output goes to a preamplifier; and the output of each preamplifier contains tones of its keyboard, pitch register, and waveform.

*A preamplifier which is not hip.

Swell And Pedal Voicing

All the Pedal and Swell voicing is done by passing these tones through formant filters. The organ contains six filters, and they are shown in Fig. 17-10.

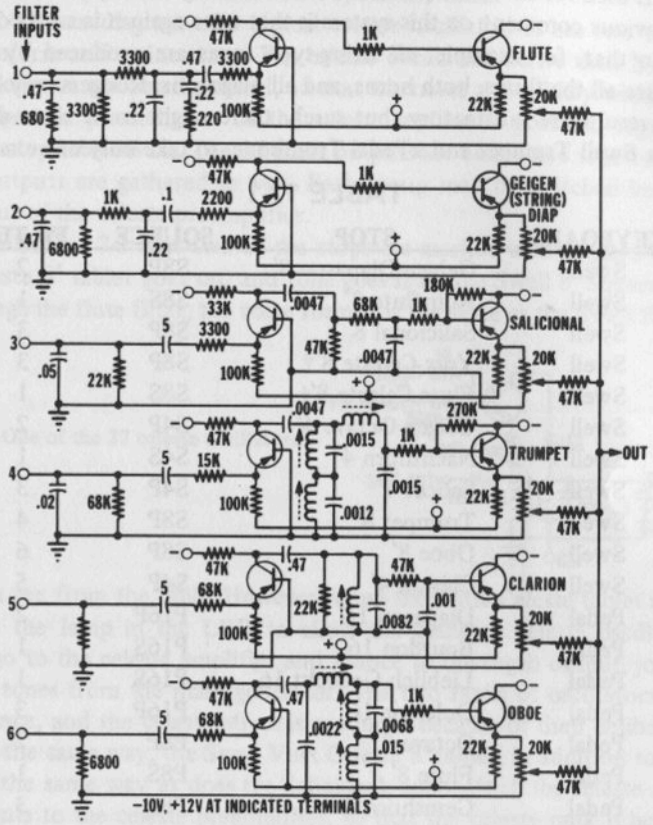


Fig. 17-10—All Pedal and Swell stops are voiced by passing square and pulse tone through these formant filters from the main generators through keyers which handle the audio.

Notice that the inductors in the reed filters and all of the output levels are adjustable. Normally these are set at the factory in a standard way, but they can be altered in the field to suit different tastes.

Each of the eight preamplifier outputs is connected to filter inputs by reed relays actuated by stop switches. We can best show what goes where by using Table 17-1, with easily understood abbreviations. A preamplifier output is designated as S or P for Swell or Pedal, followed by pitch register, and finally S or P for square or pulse waveform. S8P, for instance, is Swell 8′ Pulse. The filter inputs of Fig. 17-10 have simply been arbitrarily numbered for identification. The first line of the table means, therefore, that when the Swell Geigen Diapason 8′ stop tab is pressed, a relay contact connects Swell 8′ Pulse tone to filter input 2. (Note that each "filter" is usually composed of two filters, one before the first

transistor and another between the two transistors.) The names attached to the filters are to describe their nature in general terms. Actually, almost all of the filters are used for more than one stop. The one labeled Geigen Diapason, for example, is used for all the diapasons of these two keyboards, as the table shows.

The obvious comment on this system is that once again it is a sort of unitized scheme, in that, for example, *all* string-type tones are produced by the same filter, as are all the flutes, both horns, and all diapasons. Rodgers may argue that the tonal results are satisfactory, but surely there ought to be some differences between a Swell Trumpet and a Pedal Trombone, to take only one example, and

TABLE 17-1

KEYBOARD	STOP	SOURCE	FILTER
Swell	Geigen Diapason 8′	S8P	2
Swell	Rohrflute 8′	S8S	1
Swell	Salicional 8′	S8P	3
Swell	Voix Celeste 8′†	S8P	3
Swell	Flute Celeste 8′†	S8S	1
Swell	Geigen Octave 4′	S4P	2
Swell	Nachthorn 4′	S4S	1
Swell	Salicet 4′	S4P	3
Swell	Trumpet 8′	S8P	4
Swell	Oboe 8′	S8P	6
Swell	Clarion 4′	S4P	5
Pedal	Diapason 16′	P16P	2
Pedal	Bourdon 16′	P16S	1
Pedal	Lieblich Gedeckt 16′	P16S	1
Pedal	Dulciana 16′	P16P	3
Pedal	Octave 8′	P8P	2
Pedal	Flute 8′	P8S	1
Pedal	Gemshorn 8′	P8P	3
Pedal	Trombone 16′	P16P	4

it does not seem to me that filters are so expensive as to warrant this particular economy. It is also usually true that using a single filter for more than one pitch register tends to degrade results. It is difficult enough with a filtering scheme to maintain tone color and scaling over a five-octave range; extending the range to six or seven octaves with good results approaches impossibility. A common comment on this organ, for example, in which I concur, is that the Pedal Trombone 16′ is a good reed stop but the Swell Trumpet 8′, which uses the same filter, is not.

Celeste

Pipe organs often have one or more ranks of *celeste* pipes (pronounced *se-lest'*, not like the celesta, pronounced *che-lest'-a,* which is a percussion instrument). These pipes are tuned one to five cycles or so sharp of those in ordinary ranks, so that when they are played along with normally tuned pipes there is a

sort of rolling, undulating effect—sometimes thought to be eerie, supernatural, or *celestial.* The Rodgers 32-B has a rank of 37 celeste oscillators tuned to Octaves 2, 3, and 4 (plus a top C). These oscillators are tuned 2 cps sharp in the lowest octave, 3 cps in the second, and 4 cps in the third.

Fig. 17-11 is a diagram of a typical celeste oscillator. It has two outputs, a pulse waveform and a sine wave. The celeste oscillators derive their B– directly from the keys of the Swell manual, so that whenever a Swell key within the 37-note compass is pressed, one of the celeste oscillators goes on, regardless of whether a stop tablet is on. The 37 pulse outputs are connected together and the 37 sine outputs are gathered as well. Each group may be switched by an LDR to the input of the celeste preamplifier.

Note in Table 17-1 that two of the stops are marked with daggers. When the Flute Celeste 8′ tablet goes on, and tone goes from the Swell 8′ Square preamplifier through the flute filter, the voice formed is the same as the Swell Rohrflute,

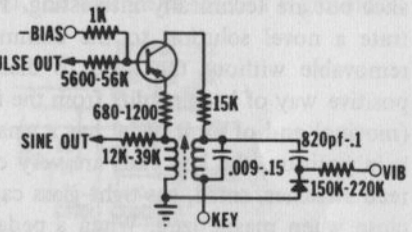


Fig. 17-11—One of the 37 celeste oscillators.

as you can see from the table. However, when the Flute Celeste tablet is used, it *also* lights the lamp in the LDR to allow the gathered celeste oscillator sine waves to go to the celeste amplifier and thence to the organ output, joining the Rohrflute tones from the main generator. The two ranks of oscillators are thus heard at once, and the celeste effect is produced because of their deliberate mistuning. In the same way, the Swell Voix Celeste 8′ tablet, in addition to forming a voice in the same way as does the Salicional, connects all the celeste oscillator pulse outputs to the celeste preamplifier, so that the celeste rank is heard as an additional source of detuned string tone. The celeste effect, while it cannot be overused lest fatigue (human, not electronic) set in, is a valuable adjunct and adds much to the organ.

To my knowledge, no one has yet discovered a way to produce the true celeste effect without a special rank of generators, despite such attacks on the problem as Baldwin's abandoned Choratone idea or the more simple minded slowly rotating Leslie speaker system. I mention this because I have a nagging feeling that exactly the same aural effect *can* be obtained much more simply and cheaply. I may not be smart enough to think of it, but perhaps you can!

An additional effect somewhat related to celeste is the Full Chorus. When this tablet is pressed, the fixed d.c. present at the vibrato inputs of the flute and main oscillators (Figs. 17-3 and 17-6) changes somewhat, changing the pitches of all oscillators of both ranks. The two ranks now beat with each other (and with the celeste rank if one of the celeste stops is on) and create a very romantic mishmash.

Sound Production

The basic tone cabinet for Rodgers organs is the W-4-50, which contains a 50-watt amplifier (all-transistor, naturally), two 12-inch woofers, a midrange horn, and a tweeter, with crossover network. At least two of them must be used with the 32-B, one for the unit flute voices and another for the remainder. It is possible to add a third cabinet, dividing the voices other than the unit flute into unit diapason and Swell-Pedal. In addition, a pedal tone cabinet B-2-50, which has two 15-inch speakers and an amplifier, can be employed to separate out the pedal voices from the rest. In the normal speaker systems, both midrange and tweeter units have level setting adjustments which can be altered to fit acoustical conditions.

Points Of Interest

The 32-B contains some additional features which are not of major importance but are technically interesting. The pedal key switches, for example, illustrate a novel solution to the common problem of making the pedal clavier removable without the need to disconnect pedal switches, as well as a very positive way of keeping dirt from the floor away from the contacts. The forward (moving) end of each pedal has a small magnet cemented to it. When the clavier is in position, the key ends are very close to a board on which are mounted 32 reed switches, small, gas-tight glass capsules containing a pair of contacts which close when magnetized. When a pedal is pressed, the magnet on its end comes into the right position to cause the reed switch contacts to close. This trick is a trifle expensive, but it is effective and interesting.

The photo of Fig. 17-1 shows a number of white "piston" buttons and toe studs. One of each operates the *sforzando,* a control common in pipe organs, which brings on at one fell swoop a large number of stops and couplers (but not usually all) to give a "full organ" effect, without moving the tablets. Usually the *sforzando* is "reversible," meaning that pushing the button or toe stud once turns it on and pushing it again turns it off. You wouldn't believe the mechanism used to achieve this reversibility in pipe organs, so I won't describe it; I'm not sure I could.

In this organ there is a bus which is connected through a diode to each stop switch (such as the stationary contact of the one in Fig. 17-4), so that if the bus is grounded all these stops go on. The bus is connected to contacts of a reversing relay (probably a ratchet type, though I don't really know) so that it is grounded when the relay contacts are closed. As the diodes are connected with their cathodes toward the stationary switch contacts, they do not allow interaction between stops in normal switching. The bus also grounds a lighting circuit so that when *sforzando* is on an indicator light says so.

The organ has three couplers, Swell To Great, Great To Pedal, and Swell To Pedal. Each coupler involves a complete rank of preliminary keyers like the one in Fig. 17-4. The D2 outputs are simply connected to the keyers for the other keyboard. The coupler is turned on and off like any stop, by switching the base bias of the transistor.

Some larger Rodgers organs have combination actions of the setterboard type which move the tablets in the usual way. The 32-B usually has instead a preset system which is a "blind" action, meaning that it does not move the tablets, and is cumulative. There are four momentary-contact buttons. Pressing the first turns on seven of the stops. Pressing the second turns on nine different stops. The third and fourth buttons turn on 12 and 14 stops, respectively. Any one of these may be pressed to bring into play the stops it controls. If two or more are pressed in sequence, all stops controlled by all of them come on and stay on. This is in contradistinction to the usual action, in which pressing any button removes the effect of buttons previously pressed. A fifth button is provided for cancel; it turns off all stops selected by the other buttons.

Fig. 17-12 shows circuitry for one of the presets. The coil is that of a reed relay (one of the glass-encapsulated magnetic switches in the center of a coil) whose normally open contacts are shown above the coil. When the Comb switch is closed as one of the four buttons is pressed, the bottom of the coil is grounded

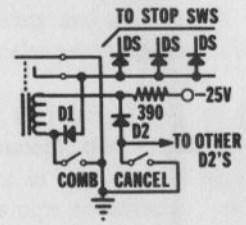


Fig. 17-12—There are four preset circuits like this.

and the contacts come together. This places ground on the bottom of the coil through D1. When the button is released, the relay remains energized and its contacts closed due to the holding action of the current path through D1.

When the Cancel button is pressed, the −25-volt source to the coil top is grounded through D2, removing current from the coil, so that the relay opens. The 390-ohm resistor prevents the cancel circuit from shorting the supply. The arm of the Cancel switch is connected through similar D2 diodes to the tops of the other three relay coils so that it shorts them all.

There is a diode DS connected between the normally ungrounded relay contact and each of the stops the relay is intended to operate. The diode connection is made to the stationary contact of the switch in Fig. 17-4, just as for the *sforzando.* When the relay is energized, the transistor bias is removed through DS just as if the stop tablet switch had been closed, and the stop goes on. The DS diodes are again "contact multiplying" diodes used so that several connections can go to the same point without transferring potential from one point to others which may not be intended to have it.

One last footnote on a Rodgers practice which I found amusing when I actually saw it followed during a visit to the factory some years ago. There is a great deal of material inside the console, as you can imagine from the description—207 generator oscillators, hundreds of keyers, and so on, most of it on etched circuit boards. Some of these boards are mounted on fold-out doors at the rear of the console, the edges of the boards fastened to "honeycombs" of wood strips. Be-

lieve it or not, they are fastened by *stapling*. Each is held in place and an ordinary, garden variety stapling gun is held to its head and the trigger pulled. The staples ram through the outer part of the circuit board into the wood. Yes, the board usually cracks, but it doesn't break and is held firmly forever.

Politics is not the only art of the possible!

Chapter 18

Allen Organs

The Allen Organ Company, Macungie, Pa., has for many years been the Rolls-Royce of the electronic organ industry, not in the sense that every organ owner would really prefer to have an Allen if he could raise the cash, but because Allen's uniquely uncompromising design approach appeals most strongly of all manufacturers to a great many of the most serious devotees of the pipe organ. It is Allen's thesis that there is nothing so important to the success of an electronic instrument as the achievement of pipe tone in all its subtlety, together with correct attacks and decays and perfect scaling, even if reaching this goal requires that tonal variety and much of the other machinery common to organs must be sacrificed or else very large amounts of money spent to obtain them.

In the typical Allen Organ, therefore, there may be a good deal of multiplication, depending on the size of the instrument, but very little complication. The design very nearly provides an oscillator in lieu of each pipe of a comparable wind instrument, each oscillator, like its pipe counterpart, voiced and scaled individually. There is almost no "fooling around" to try to obtain several voices from a single set of generators or to produce any other effects by methods which would compromise the perfection of what tone *is* provided. Compromises do exist, in that, for example, to achieve a reasonable registration in an organ of reasonable size, the "unit" system is used–unification and duplexing–but in any Allen organ, the quality of the tone cannot be faulted.

It does not necessarily follow, however, that an Allen is necessarily a "better" organ than one of another manufacturer in a similar price range, because the definition of quality in an organ may very well take in more than simply the tonal character of what sounds are heard. An Allen which costs several thousand dollars, for instance, may have only three different tone colors, keyed in various pitch registers. While each of these three voices may well be a gem of perfection, a particular individual may prefer instead a much smaller and cheaper organ of another manufacturer which has a dozen or more different sounding voices, even though none of them has the authenticity of each Allen voice. In other words, many people–probably most people–are willing to forgive some lack of absolute perfection in voicing in exchange for more variety; and this is entirely legitimate, indicating no lack of taste or intelligence. At least I hope so, for I am myself in that category.

I point this out quite specifically, not to denigrate Allen in any way, but in an attempt to dispel some of the mysterious aura of invincibility which has surrounded the Allen name on the tongues of many people because of well deserved

but misinterpreted praise from those considered to be among the cognoscenti. I have known of amateurs without any particularly impressive musical background or any deep need for Allen's particular forte of precise fidelity to pipe sounds, who have bought small and medium sized Allens because they "heard it was the best." Many such people have later found themselves much happier with an instrument of greater variety and less subtle tonal perfection.

It is, of course, possible to have your cake and eat it, too, with an Allen, just by spending enough money. As much variety and versatility as anyone might ever want is obtainable from this manufacturer in larger organs, custom made, containing a great many ranks of generators, each retaining its individual perfection but the total instrument providing wide and colorful registration simply



Fig. 18-1—Allen's catalog photo of the TC-4 with the optional drawknob console. The Classic is about the same, plus Whind and chiff.

by providing a sufficient number of ranks on the same basis as a pipe organ. To my knowledge, there is no real limit to the size of an organ Allen will build, and I have heard of prices as high as $90,000. The generator ranks of larger Allen organs are not contained within the console; they are placed in racks in a room remote from the console, in further analogy to pipe organs. In the days when tubes were used, some churches had quite impressive auxiliary heating plants made up of tube filaments. Today, of course, now that transistors have taken over, things are much cooler and less monstrous.

While Allen does have standard models, almost none of them are as standard as those of other manufacturers, being subject to changes, modifications, and additions for the purchaser; and a good deal of Allen's yearly sales figure is accounted for by instruments which are custom made from the ground up. It is difficult, therefore, to describe a single model as we have done in other chapters, and the difficulty is compounded by the fact that Allen, alone among organ makers, is extremely reluctant to give out technical information to anyone—including customers. We shall therefore give a fairly sketchy description of general practices, related in part to a typical medium size model, the Classic or the TC-4,



Fig. 18-2—The Custom Theatre organ is a three-manual model with the customary curved stop board. Allen maintains its own cabinet shop, which would seem almost to be a necessity in view of the amount of custom building the company does.

which differ mainly in that the former includes two important Allen features, chiff and Electronic Whind (correct spelling, not a misprint). On these and larger models, the console may have either drawknob stop controls, as in Fig. 18-1, or the more usual tongue tablets. Fig. 18-2 shows the 3-manual Custom Theatre organ, with the usual curved stop board and tongue tablets. The odd looking instrument of Fig. 18-3 is an organ built especially for the Church Center of the United Nations. The structure atop the console houses the speaker system.

The Classic Organ

The Allen Classic illustrates as well as any model the way a typical instrument of a standard type is made up. The Classic has three ranks of tone generators. Each rank is made up of oscillators like the one in Fig. 18-4. The coil is a toroid and is used by Allen to obtain maximum tuning stability. (Allen also sells these coils to industrial users, along with oscillator units for various purposes.) The inductance of such a coil cannot be varied, so a part of the tuning capacitance is connected across the coil through a variable resistor. The resistor allows the oscillator to be tuned over a limited range, but a capacitor must be changed if a larger amount of retuning is needed.

Each of the oscillators of the diapason rank is followed by a fixed and variable resistor with which the signal level can be adjusted for perfect scaling, and then by a half-wave rectifier which passes on only the positive alternations to the following amplifier. This method of transforming sine-wave oscillations to diapason tone is the same as that used by Rodgers.



Fig. 18-3—This odd looking object is an organ with the speaker system on top, made by Allen for the United Nations.

Each of the flute oscillators also passes signal to a level adjusting potentiometer, following which there are in series with each signal two diodes in parallel, with each cathode connected to the other diode's anode. These diodes pass both alternations of the signal, but since the effective resistance of a diode changes in proportion to the current through it at low current levels, the slope of each alternation near the baseline is altered and the wave passed on to the amplifier is no longer quite sine. This puts some harmonics into the flute signal and thus makes them what Allen terms "open" flutes—meaning flute tones having harmonics. In pipe terms, open flutes are those produced by pipes not stopped at the far end and having both odd and even harmonics, while stopped flutes are those produced by pipes of half the length, stopped at the end, giving a symmetrical waveform substantially without even harmonics. In Allen's private language, however, open flutes have harmonics; stopped flutes do not. None of them have even harmonics to speak of, and all would be called "stopped" by anyone else.

The output of each reed oscillator is, like the others, controlled by a variable resistor, and then passed through a diode and a very simple formant filter. The diode wipes out the negative alternations so as to distort the waveform and produce harmonics. The filter forms the spectrum to produce the desired reed sound. Since there is a filter for every note, the values can be graded as necessary to produce the same smooth voicing and scaling for the entire rank as a pipe voicer is able to achieve.

Keying is by the simplest possible system, schematically—so simple it is not worthwhile showing. An oscillator is normally off. Note in Fig. 18-4 that the transistor is a PNP and the collector grounded for d.c. The key switch supplies +12 volts to the emitter and the bottom of the base bias voltage divider through R. Before the full 12 volts can be impressed on the bottoms of the base and emitter resistors C must be charged. R and C are a time-constant circuit which causes the oscillator to speak slightly slowly to simulate the attack time of a pipe. When the key is raised, C must discharge through the transistor before all operating voltage disappears, and this delays the turn-off as well. A sustain effect is made available by shunting additional capacitance across each C.

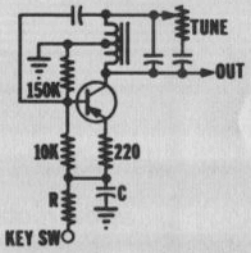


Fig. 18-4—This is the standard tone-generator oscillator. The coil is a toroid and the variable resistor allows tuning over a limited range.

The entire organ makes available only these three tone colors, and the similarity to a unit pipe organ with three ranks is extremely close. On the Great, for example, there are four diapason stops, Principal 8′, Octave 4′, Nazard 2⅔′, and Super Octave 2′. Each key moves one switch finger for each of these stops, and for each stop there is a slider running from one end of the keyboard to the other. When the tablet goes down, an electromagnet moves the slider into position so that the contact finger for each key for the stop will pick up keying voltage and send it to oscillators of the correct pitches. The four stops mentioned are, of course, simply the same diapason oscillators with a different name for each pitch register.

The Great manual also has four flute stops, Quintaton 16′, Quintadena 8′, Quintadena 4′, and Spillflöte 2′. Here again, the manual is simply keying the flute rank at four different pitches. There is a key finger for each pitch of the flutes and a slider. There are also two reed stops, keyed in the same way with the reed rank. These stops, both called Regal (pronounced *re-gal′*), are at 8′ and 4′.

The Swell and Pedal operate in the same way. The Swell keys flutes at seven pitch registers, diapasons at two, and reeds at three. The pedals have two flute pitches, four diapasons, and three reeds. We thus have an organ with 31 speaking stops (plus a mixture), all made by keying only three ranks at various pitches and with only three different tone colors. The scheme of this organ, which runs in the area of around $10,000, is an excellent illustration of both the virtues and faults of Allen in general. Each of the tones available is about as close to pipe quality as one can get; but in return for that, the organ presents an extreme poverty of the kind of tonal variety which is normally at least one very important reason for the organ's existence as a serious musical instrument. For the polyphonic literature such as Bach, where the structure of the music is overpoweringly the most important aspect of it, such an organ would be satisfactory in many ways, and would be comparable in resources to some of the portative or

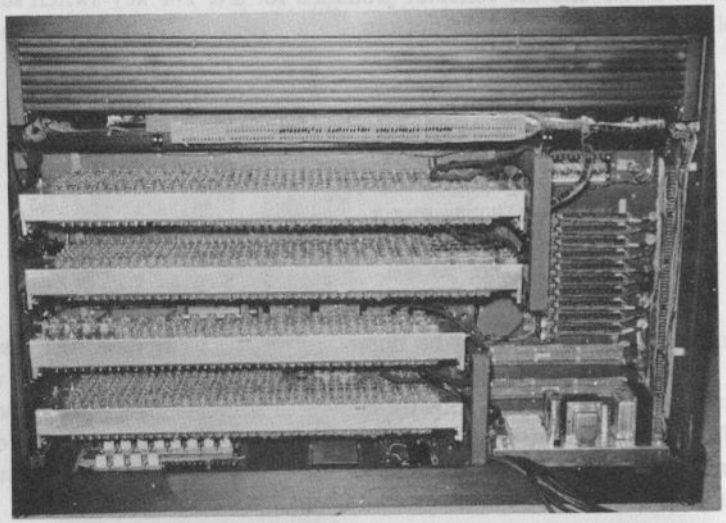


Fig. 18-5–This rear view of a Classic organ shows the inside stuffed full of generator oscillators, three ranks plus celeste. Larger Allen models require external generators.

small unit pipe instruments available for baroque work (at around the same price, often), or to the practice organ often found in a music school. It would be suitable for the kind of liturgical music used routinely during a standard and not overly pretentious service. In my opinion, it would be hopelessly inadequate for much romantic music or for modern organ compositions where tone color variety assumes real importance.

It has one point of superiority over a comparable pipe organ–it can make lots more noise than three ranks of pipes if enough amplification equipment is added and thus can be more effective in accompanying singing by large groups. Placing the speakers at widely divergent points in the auditorium and having the sound of each rank come from more than one speaker system can also give a more spacious effect than three pipe ranks.

An optional celeste rank is available for this organ. It consists of 44 oscillators similar to the others, which are keyed simultaneously with the diapason oscillators. They are tuned 2 to 5 cycles sharp of the diapasons. A gimmick or two is provided on some Allen instruments to increase the variety slightly by changing the over-all volume and/or tone color of a rank somewhat in the signal path between the joining point of all the oscillator outputs and the amplifier input. These are called "Diapason Becomes Dulciana," "Diapason Becomes Viole," and "Flute Becomes Softer." In some organs the keying voltage can be changed and this alters both the amplitude and harmonic content of the oscillator outputs. The effect, of course, is to change the existing tone color, not to add another one in the conventional sense. There are, I am certain, some organists who like these things, but it just happens that to my knowledge I have never met one.

A separate amplifier and speaker system is used for each of the three main ranks, with LDR's inserted in series with the signal paths to provide swell-shoe control. The shoe operates a variable shutter which controls light falling on the cells. There is no vibrato built into the organ. Vibrato, if any, is furnished by a Gyrophonic Projector (page 126).

Fig. 18-5 shows the back of a Classic organ, with the four ranks of oscillators occupying most of the console space. Oscillator tuning and volume controls are reachable from the organ rear. This photo is an excellent illustration of what Allen organs are made of–oscillators, and very little else!

Chiff And Random Noise

Two important effects have been introduced by Allen in recent years to add to the authenticity of Allen tone. One of these is chiff, which we discussed in general on page 352. The Allen chiff is sometimes created by keying an oscillator a twelfth higher, as in the Rodgers organ of Chapter 17, but Allen also has a way of doing it with natural harmonics of the keyed tones. Fig. 18-6 is a sample chiff circuit of the latter type.

Output of one generator oscillator is connected to the Tone In point, so that a sine wave appears there. Initially, capacitors C are discharged. Resistors R are very large and have no function except to discharge the C's when the key is re-

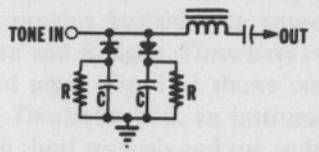


Fig. 18-6–True harmonics of the oscillator tone create chiff.

leased and tone is no longer presented to the circuit. When the tone is first keyed on, each sine-wave alternation causes current to pass through one of the diodes to charge a capacitor. The charging takes a number of alternations. Until the capacitors are charged, therefore, the tone appearing at the output is clipped sine wave, producing harmonics, all of which are odd because of the symmetrical clipping. The L-C filter emphasizes a particular odd harmonic, which causes the chiff sound. As time goes on, the capacitors become more nearly charged and

less and less current can pass through the diodes, so that the clipping of the sine wave and harmonic production decrease. After about 75 milliseconds both capacitors are fully charged and all clipping stops, after which the output consists of sine wave. The chiff effect can be switched on and off, probably (Allen didn't tell me) by lifting the grounds of all the chiff circuits.

One of the strongest objections to electronic organ sound is the artificial sounding steadiness of each note's pitch and loudness. To break up at least the amplitude steadiness, Allen has what it calls Electronic Whind. It consists of a source of random noise with frequency components in the d.c. to 20-cps portion of the spectrum. The keying voltage supplied to the oscillators through the key switches is modulated by this random noise so that the voltage reaching the oscillators is not steady. The effect on oscillator amplitude is, to quote Allen's words, similar to that caused on pipes by air turbulence resulting from irregularities in the air path between blower and pipes and by pressure changes produced by varying loads on the air supply during playing. This is, of course, substantially different from merely supplying air hiss to the speaker system.

However, in addition there is a resistive path between the keying point of each oscillator and the output of the chiff circuit. Some of the keying voltage is carried through a large-value resistor into the audio output in this way, so that a hiss is produced to simulate the actual air noise in pipe organs. Since this is done for each oscillator, the hiss naturally builds up in proportion to the number of oscillators speaking at any instant. My opinion of this particular part of the system—creating a nonmusical noise for the sake of realism—appears on page 353.

An amplitude modulation tremolo is sometimes added by modulating the keying voltage with tremolo-frequency sine wave as well as by noise.

Chapter 19

Conn Organs

One of the old established firms in the electronic organ business is located in Elkhart, Indiana. Conn Organ Corporation is a part of C. G. Conn Ltd., whose name was made long before the advent of electronic music by its long history as a band instrument manufacturer. Conn continues to make band instruments, some of the most widely sold in the world, backed by a good deal of original research in the field of acoustics. Its line of organs, however, has come through the years to constitute a larger and larger part of its business.

In common with Baldwin, another old-line musical instrument house (though its instruments are pianos), the Conn background has always seemed to influence its organ designs, which have tended to stress purely musical values, rather than the economies which lead to low prices and the peripheral "features" designed to dazzle prospective customers unfamiliar with organs. Conn, for example, has always stuck to individually tuned oscillators rather than frequency dividers, on the ground that the chorus effect justifies the resulting cost and voicing restrictions. Conn organs have always sounded good to my ears, even though they have sometimes lacked the voicing variety obtainable by other methods. I am not going to try to settle the controversy over frequency dividers *vs* separate oscillators here, any more than I did in Chapter 3. But one must admire Conn for sticking to its individual-oscillator guns out of, I believe, musical rather than commercial considerations, even in their smaller instruments. In the past, this has sometimes hampered them competitively for price reasons. That is no longer really true, and Conn organs are today among the most widely sold and highly respected of brands, though not one of the two or three sales leaders.

The Conn organ catalog ranges from fairly small spinets up to classic instruments of moderately large size, though nothing beginning to approach the medium and larger sizes and prices of Allen and Rodgers. There have even been noble efforts at combining electronics and pipes. Fig. 19-1 shows one of the most popular modern Conn organs, the Theatrette 550, an instrument with excellent theatre voicing and effects, with short manuals and toe pedals like a spinet. The Classic 825 which appears in Fig. 19-2 is one of the larger models, though not the largest, and we shall discuss it in this chapter.

The Classic 825 is the latest of a long line of Conn organs carrying the Classic name.* It has 34 speaking stops, plus chimes, in 16′, 8′, 4′, 2⅔′, 2′, and 1′ pitch

*The same names have been used for many years to denote various organ sizes and types, only the number changing to show a model change. This might be the sixth Classic.

registers, with a range of 97 fundamental pitches–eight octaves plus a top C. There are two versions, one of which has three output channels and the other six, all speaking through external tone cabinets or custom speaker installations,



Fig. 19-1–The Conn Theatrette, a spinet with curved stop board, has become very popular.

including, if desired, a Leslie system. There are two swell shoes, one for each manual, plus a crescendo shoe. A tripper-type combination action and a sforzando reversible are included.

Tone Generation

In earlier years, the Conn scheme usually included a complete set of generators for each manual, each generator being an oscillator normally off and supplied operating voltage by a key switch. Each oscillator had a sine-wave output and a pulse output. All outputs of each waveform were permanently connected together; the flutes of the organ consisted of the sine waves, and the pulses were passed selectively through formant-type filters to form other voices. Each manual thus had one "unit" generator, with all voices in the same pitch register–8′. Couplers were provided, which could cause the same oscillators to be keyed at different pitch registers, but there was no way to produce one voice at one pitch and another voice at a different one. This is, of course, a basic limitation of all unit schemes.

The present Conn operates on much the same lines, with some twists. There is a single seven-octave (plus one note) rank of oscillators which run constantly.

Each oscillator has three outputs–sine, pulse, and sawtooth. There is a transistor keyer for the sine output of each oscillator. These sine-wave keyers are employed to furnish a unit rank of flutes, keyed from all three keyboards at various pitches. There is a transistor keyer for the sawtooth output of each oscillator; these are used as a unit diapason, keyed at four pitch registers from the Great. There is also a separate rank of 32 sawtooth keyers for the 16′ pedal diapasons. And finally, the pulse outputs are directly (not electronically) keyed at three pitch registers for each manual to furnish string and reed tones through filters.

Fig. 19-3 is a diagram of one of the 85 oscillators; values shown are for A-440. The coil core consists of a stack of E laminations with a stack of I laminations across the open part of the E. A simple spring-loaded screw arrangement allows one end of the I stack to move away from the E, and this is the fine frequency adjustment. See page 46.

This is undoubtedly one of the most interesting generator oscillators in the entire industry, for every last ounce of utility is squeezed from it. It is basically a fairly ordinary grounded-plate Hartley; if you will short resistor R to ground, remove the plate resistor and the other funny business at the plate and connect it directly to +36 volts, and take output from cathode or the top of the coil, a simple Hartley oscillator is what you will have. The bias at the point shown is 6.6 volts for this frequency in the actual organ, but this is for chorus and vibrato purposes and you will get the same oscillation by just grounding the bias point.



Fig. 19-2–The Classic 825, described in this chapter, is a full size formal instrument.

However, parameters and voltages are such that the loop gain is high and the oscillator is driven into class C operation, meaning that the tank circuit is kept excited by a pulse of plate current once per cycle, with the tube cut off for the

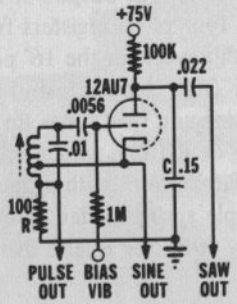


Fig. 19-3—One of the first 73 generator oscillators.

remainder of the cycle. Let us now restore the circuit to its condition as shown in Fig. 19-3 and see what this cyclic pulse of plate current does.

The first thing it does is place a small pip on one side of the sine-wave output, so that the "sine" wave definitely contains some harmonics. This yields a better flute tone than pure sine wave would.

Next, the pulse itself can be taken off across resistor R, through which, of course, the tube's plate current passes. The pulse has roughly the form of a lower case letter h, and it is passed through formant filters to make the string and reed voices of the organ. Conn has always used a pulse like this for the purpose, rather than sawtooth or square waves as is more common. They prefer the pulse to

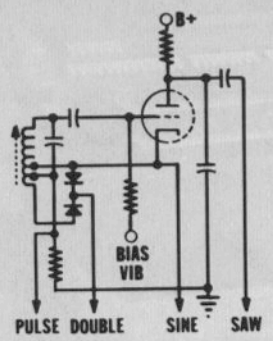


Fig. 19-4—Upper 12 oscillators have doublers, giving an extra octave of tone at the top.

such an extent that the retention of tubes in the generator section of the organ (most of the rest has transistors) is due, so they say, to the great difficulty of obtaining exactly this pulse with a transistor oscillator.

Third, the cyclic plate-current pulse makes it possible to obtain sawtooth waves from the tube plate. During the portion of the cycle when the tube is cut off, capacitor C charges, through the plate resistor, in the usual exponential fashion; C is chosen for each note so that the rate of charge gives a gradual rise of potential across C in the time available. When the tube suddenly conducts, it discharges the capacitor; and when the pulse recedes, the capacitor begins to

charge again. This gradual charge and fast discharge constitutes a sawtooth potential across C. The flyback is fairly slow, not the invisible vertical line usual from most other sawtooth generators; these waves are used solely to form the diapason tones and the diapason filter is built to deal with them. The .022-μf capacitor is simply for d.c. blocking.

But you ain't seen nothing yet. If you have a sharp eye, you may have noticed that a little while back I said the organ provides 97 fundamental pitches—eight octaves and a top C—while a little later I said there were 85 oscillators. This was not an error.

Fig. 19-3 represents (with various value changes) only the first 73 of the oscillators, covering Octaves 1 through 6 in Table 1-1 (plus the C from Octave 7). Fig. 19-4 is one of the top 12 of the 85 oscillators. It has everything shown in Fig. 19-3—plus an extra little winding at the bottom of the inductor with a couple of diodes which are placed back to back across the cathode tap and coil bottom to form a full-wave rectifier. As everybody knows, the fundamental frequency of a full-wave-rectified sine wave is twice that of the original, and we thus have a frequency doubler added to each of the top 12 oscillators, giving an additional 12 generated notes for a grand total of 97! It is true that these 8th-octave tones are exact multiples of those an octave lower, rather than independent as are all other tones, but at these high frequencies (the top C is 8372 cps) no one will ever know the difference—or wouldn't if I hadn't spilled the beans. As you may have guessed, I am filled with admiration for the man who dreamed this one up. Rarely does one find an electronic circuit that works so hard for its living as these oscillators!

An octave of basic tone for almost nothing is Christmas in any organ designer's language. With its total of eight octaves and one note, the organ is now complete to the tops of the manuals for the 2′ register and breaks back only one octave for the valuable 1′ stop.

Flute Keyers

For each of the 97 available pitches there is one flute keyer, a gate circuit which is opened by a key switch to allow flute (sine-wave) tone to pass from the associated oscillator to the flute preamplifier. Fig. 19-5 shows how the flute keyers work. Note first that the diagram has three main parts. At the left I have shown two keyers, each consisting of a transistor, 7 resistors, a capacitor, and a neon lamp. Two keyers are shown so that you will be able to see what the common connections to the other circuitry are. You may assume that all 97 flute keyers are connected to the single sustain circuit at the right of the vertical dashed line in the same way as are the two shown.

At the right of the dashed line is the sustain voltage regulator and the two sustain control switches. There is one of these for the entire organ.

In the dashed enclosure at top is a simple coupling circuit, plus a collector resistor which carries B— to the transistor collectors. There is one of these coupling circuits for each octave of keyers, all 12 collectors in the octave being connected together and to the 10,000-ohm resistor and coupling circuit. This

coupling circuit carries flute tone permanently to the flute preamplifier. The only irregularity is that the top octave of tones, being obtained from the doublers, require special filtering to eliminate the harmonics caused by the rectification of the sine waves. Their coupling circuit is therefore more complex than the one shown.

Let us look first at one of the keying circuits, which are easy to understand. The transistor is used as a common-base amplifier; that is, its base is at ground potential for a.c. The IN terminal is connected to the sine-wave output of one of the oscillators, and the signal is applied through a scaling resistor to the emitter. If the transistor is operating, output will appear at the collector.

The cathode of tube V2 is, as we will see later, at a high positive potential. This potential is applied to the resistor string R1-R2-R3-R4 of each keyer. The resistors form a voltage divider, with the transistor base connected to the junction of R3 and R4. Since this is a PNP transistor, the resulting positive base bias cuts it off and none of the input signal goes to the collector.

When a key is pressed, −30 volts is applied to the Key terminal. Since to reach point A it only has to go through 6800 ohms, while the positive voltage from V2 passes through resistors of much higher value to reach point A, the positive voltage is swamped out and point A becomes negative. This negative

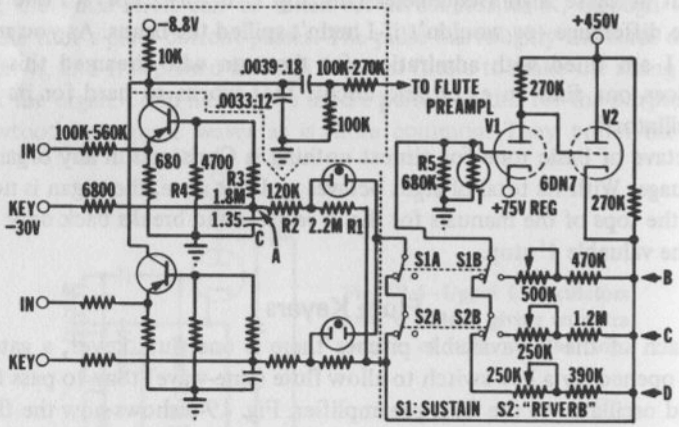


Fig. 19-5—A flute keyer is associated with each oscillator. Two are diagrammed here.

voltage reaches the transistor base through R3, turning the transistor on and producing tone output at the collector. When the key is released, the negative potential disappears and the base goes positive again, turning off the transistor.

Both the on and off conditions are gradual due to capacitor C. When the key is pressed, the 6800-ohm resistor and C form a time constant; the change of the charge on C from positive to negative takes a little time and the transistor passes the tone with a slight delay rather than instantly. When the key is released, the capacitor must again change its charge back to positive. This it must do through R2 and R1; thus the time required for the tone to die is controlled by a different time constant. This, then, is how a keyer works. Let us next examine the way in which the decay of tone—the time required for the charge on C and thus the base voltage to become positive enough to cut off the transistor after the key is released—is controlled. If the decay time is about the same as the attack time, keying is normal. If the decay time is significantly longer, the effect is that of sustain.

V1-V2 is actually a voltage regulator circuit with switch-controlled voltage output. The Sustain and "Reverb" (a short sustain) switches are d.p.d.t. The positions shown in the diagram are for normal keying—no sustain.

The B+ potential of 450 volts is applied to V1 through a plate load resistor and directly to the plate of V2. V1 is, therefore, an ordinary grounded-cathode amplifier (for d.c., by the way—no audio is involved here) and V2 is a cathode follower. The cathode resistor of V2 may be considered to be whichever of the three pairs of series resistors, B, C, or D, is switched in, plus R5 in shunt with a thermistor which adjusts the net grid-ground resistance to compensate for temperature changes which may affect the transistors. Keep in mind that the output of a cathode follower, taken from the cathode, is (over the useful range) independent of the value of cathode resistance. Thus, when the switches bring in string B, C, or D, the effect is not directly to alter the cathode voltage but to alter the voltage applied to the V1 grid. They do this because each string B, C, or D is the series leg of a voltage divider, and R5 is the shunt leg, supplying V2 cathode potential to the V1 grid.

With the switches as shown in Fig. 19-5, potential is applied to the V2 cathode through the 270,000-ohm resistor from the +450-volt supply. This potential is voltage-divided by the D pair of resistors and R5, then applied to the V1 grid. The result is a certain voltage at the V1 plate, which is direct-coupled to the V2 grid, and is reproduced by cathode-follower action at the V2 cathode. This constitutes a complete cycle of d.c. negative feedback, giving regular action. To see it, assume the V1 grid goes plus, then the V1 plate goes minus, and so does the V2 cathode. The minus-going V2 cathode is coupled back to the V1 grid through the resistors, opposing the original plus-going potential we assumed there.

Let us suppose now that switch S1B is turned the other way, so that now resistor pair B is between the V2 cathode and V1 grid. Assuming the total resistance of the B pair is different from that of the D pair, a different voltage is applied to V1. A different voltage therefore appears at the V2 cathode—but it will be maintained just as immovably once there as was the original voltage with the D resistors in circuit.

It is the V2 cathode voltage which turns off the keyers when a key is released. Ignoring the A switch sections for the moment, and the neon lamps, the larger the V2 cathode voltage the more quickly will point A become positive after the key is released and the faster will be the tone decay. If we reduce the V2 cathode voltage, the decays will take longer, because capacitor C will take longer to change charge from negative to positive.

Now, here is what actually happens. With the switches set, as shown, for normal keying, S1A and S2A keep a neon lamp shunted across each R1. The

voltage drop across R1 due to the voltage supplied by the V2 cathode is above the firing voltage of the lamps. The lamps are thus ignited all the time. They constitute low-resistance shunts across the R1's, so that effectively R1 is a very much lower value than the 2.2 megohms shown. This being the case, substantially only 120,000 ohms exists between the V2 cathode and point A, so that when a key is released C changes charge rather quickly and the transistor shuts off almost at once.

When either of the sustain switches is on, its A section opens and the neon lamps are removed from the circuit. When that happens, R1 becomes an actual 2.2 megohms, so that even if the V2 cathode voltage remained the same, C would take longer to change charge and shut off the transistor.

When S1 is turned on, the neon lamps are disconnected and the design value of positive voltage is placed on the V1 grid by adjustment of the potentiometer in the B string (a factory adjustment). With this value, the time taken for C to change charge is long enough to give the full sustain time.

If S1 is left as shown and S2 is turned on instead, the C string takes over. It has higher resistance, so that less positive voltage is applied to the V1 grid. This means more positive potential at the V1 plate and V2 cathode, hastening the transistor turnoff and producing a shorter sustain.

The actual blocking voltage present at the V2 cathode when room temperature is 70°F is 178 volts for normal keying, 314 volts for "reverb," and 214 volts for sustain.

The outputs of all the coupler circuits from the flute keyers come together and go to a flute preamplifier and voicing filter, thence to the flute expression control circuit, of which more later. The voicing filter includes two tablet switches. One, labeled Flute P, is across a series resistor; turning on the tab

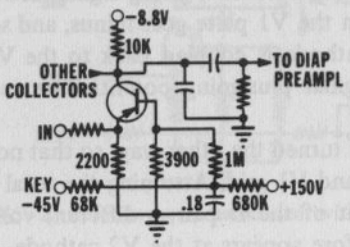


Fig. 19-6—Diapason keyer handles sawtooth waveforms.

opens the switch, placing the resistor in series with the signal and lowering overall flute volume. (The P means *piano,* the Italian word used in music to mean soft volume, the opposite of *forte.*) The second switch, normally open, places an additional capacitor in shunt with the filter, increasing the low-pass action. The tablet is labeled Flute Treble P.

Diapason Keyers

There is also a diapason keyer associated with each oscillator, plus an additional group of 32 diapason keyers for the 16′ pedal tones. The diapason keyers look like Fig. 19-6. As you can see, these are just like the flute keyers, minus the

complication of the blocking voltage regulator; no sustain is applied to diapason tones. The blocking potential is a steady 150 volts, as shown. Input for each diapason keyer is taken, of course, from the sawtooth output of an oscillator. Each group of 12 keyers has commoned collectors connected to the 10,000-ohm collector resistor and followed by the same sort of output coupling circuit as Fig. 19-5. Outputs of all the coupling circuits go to the diapason preamplifier and voicing filter, thence to an expression control which handles all manual diapason tones plus the Great strings.

A tablet-operated switch in the diapason filter shunts an additional capacitor to ground when the tablet goes on, reducing high-frequency response. The tablet

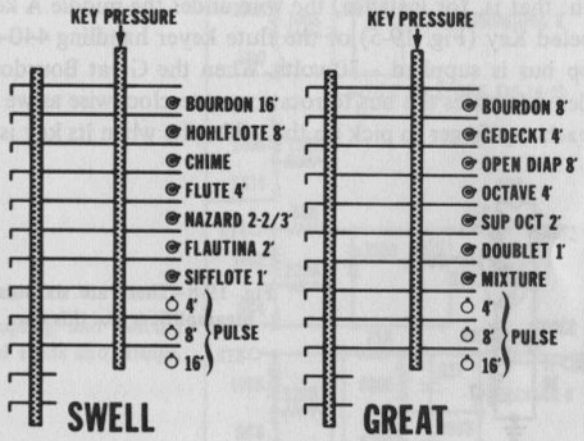


Fig. 19-7—This is the switching scheme under the manuals—fingers and rotatable busses.

is labeled English Diapason. The pedal diapason keyers are the same as the one shown, except that the components are selected for a longer attack and decay time, appropriate for pedal tones.

Keying

Each of the manual keys is required to do several things. Since the organ has flute stops at five pitch registers and diapasons at four, meaning that the two groups of keyers must be keyed correspondingly, there are two choices of methods. The first is the Rodgers path, as described in Chapter 17, using a single switch under each key and "contact multiplying" diodes, at least one per key per pitch register. Conn has chosen the second method, one switch contact per function. Fig. 19-7 shows the under-key contacts for the two manuals, Swell and Great.

Under each key there are 12 wires, the left ends of which, in the drawing (rear ends in the organ) are molded into a plastic dowel which is fixed in place vertically in the organ. The lower two wires are clipped off, as shown, and are simply used as tie points in wiring. The wires also run through holes in a second dowel which moves down when a key is pressed, bringing the free ends of all the

wires into contact with the 10 bus rods, which are horizontal and run the length of the manual.

In each keyboard, the upper seven bus rods consist of a conducting material (silver) with a ridge running the length of each rod. Around the rod is insulation which covers its entire surface except for the projecting ridge. When the rods are in the positions shown, with the ridge pointing at about 2 o'clock, the contact fingers touch only insulation when they come down. Each rod is rotatable, however; when it rotates so that the ridge is uppermost, the fingers make contact. This system is exactly like that used by Gulbransen (Chapter 13) and Wurlitzer (Chapter 14) in principle, though details differ, as you would expect.

All the switch fingers in the top row of the Great are wired to the Flute keyers at 8′ pitch; that is, for instance, the wire under the middle A key is wired to the point labeled Key (Fig. 19-5) of the flute keyer handling 440-cps sine-wave tone. The top bus is supplied −30 volts. When the Great Bourdon 8′ tablet is pressed, a solenoid causes the bus to rotate counterclockwise as we view Fig. 19-7 and allow each top finger to pick up the −30 volts when its key is pressed. The

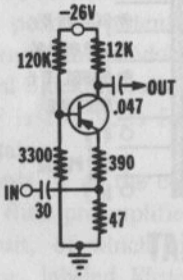


Fig. 19-8—There are six pulse preamplifiers like this one.

second row of Great fingers is wired to the flute keyers at 4′ pitch, and the bus is supplied −30 volts. The third row of contacts is wired to the diapason keyers at 8′ and the bus carries −45 volts, so that when the Open Diapason 8′ tab goes on, diapason keyers send 8′ diapason tone through the system. And so on.

The lowest three busses for each manual are not rotatable; their conducting ridges are always facing up, so that when a key is pressed the lower three contact fingers always make connections to their busses. These fingers are connected to the pulse outputs of the oscillators. Each finger is connected to one oscillator output through a 1,000-ohm isolating resistor. The fingers in the bottom row are connected to 16′ sources. That is, for instance, the one under the middle A key carries 220-cps tone, etc. The next two, going up, are connected to 8′ and 4′ pitches. This scheme is exactly the same as in a number of other directly keyed organs. The rods themselves are made of a conductive core with a coating of vinyl impregnated with a resistive material which is sensitive to pressure. Contact is therefore made somewhat gradually. The pulses picked up by these lower three rods are used for all the manual string and reed voices.

String And Reed Voicing

Each of the six manual pulse rods of Fig. 19-7 is connected to the In point of

a pulse preamplifier identical to that of Fig. 19-8. There is therefore one such preamplifier for Great 16′ Pulse, another for Great 8′ Pulse, and so on. Each amplifier is a grounded-base type, for which Conn appears to have a predilection.

Fig. 19-9 shows the outputs of all six preamplifiers as points labeled SW 4, 8, and 16 and GT 4 through 16. The drawing shows the coupler system and most

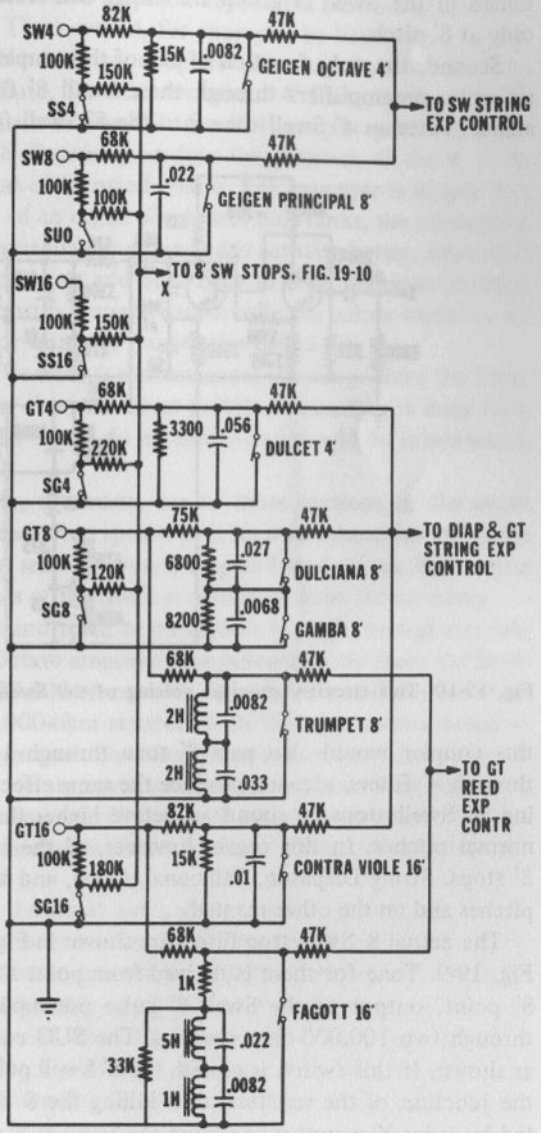


Fig. 19-9—Coupling and voicing for the manual reeds and strings.

of the string and reed voicing. Let us first look at the couplers in some detail.

The organ has six couplers: Swell To Swell 4′ (SS4), Swell Unison Off (SUO), Swell To Swell 16′ (SS16), Swell To Great 4′ (SG4), Swell To Great 8′ (SG8),

and Swell To Great 16′ (SG16). The parenthesized symbols are those shown in Fig. 19-9 to identify the coupler switches.

To understand the coupler system, a few preliminaries must be noted. First, the Swell tablets on the stop board are divided into two groups, Swell I and Swell II. Only the Swell II group is subject to coupling; and the only stops contained in the Swell II group are strings and reeds (pulse-produced voices), and only at 8′ pitch.

Second, the only function of any of the couplers is to pass tone to one of the six pulse preamplifiers through these Swell 8′ filters. The SS4 coupler, for instance, switches 4′ Swell tone into the 8′ Swell filters. In a more normal organ,

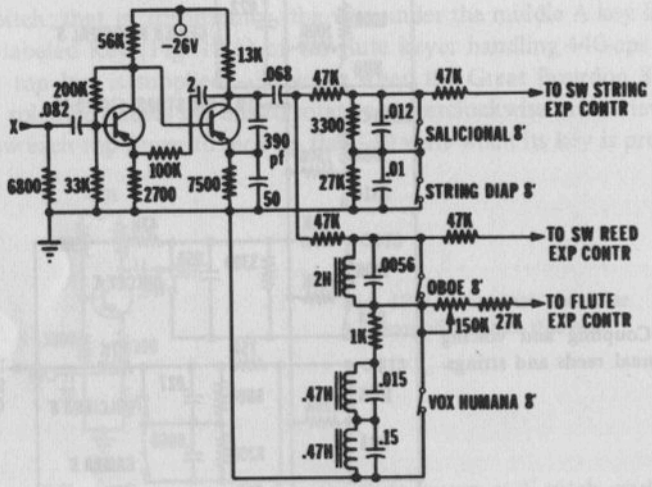


Fig. 19-10—This circuitry provides voicing of the Swell 8′ stops derived from pulse tone.

this coupler would also pass 8′ tone through 16′ filters and perhaps 2′ tone through 4′ filters, etc.—or produce the same effect by other means, that of causing all Swell stops to sound an octave higher than normal in addition to their normal pitches. In this organ, however, all the couplers do is make four Swell 8′ stops, String Diapason, Salicional, Oboe, and Vox Humana, available at other pitches and on the other manual.

The actual 8′ Swell stop filters are shown in Fig. 19-10 to save some clutter in Fig. 19-9. Tone for them is derived from point X in Fig. 19-9. Note that the SW 8′ point, output of the Swell 8′ pulse preamplifier, is connected to point X through two 100,000-ohm resistors. The SUO coupler switch is normally open, as shown. If this switch is closed, the 8′ Swell pulse tone is shorted to ground at the junction of the resistors, thus killing the 8′ Swell unison, so that the stops fed by point X cannot sound from the Swell at 8′ pitch.

The SS4 switch is normally closed. When the tab is pressed and this switch opens, Swell 4′ tone is passed through 100,000 and 150,000 ohms to point X and the Swell 8′ filters. When the SS16 switch is opened, Swell 16′ tone passes

to point X. And tone is supplied through coupler switches from the other manual in the same way.

This appears to be a rather incomplete and unsatisfying coupler system—and it is. If, however, we accept the proposition that aside from couplers the voicing of the Swell is adequate in variety, taking into account the other voices supplied at six pitch registers from the flute and diapason keyers, the absence of more extensive coupling makes sense. There would, for example, be little point in supplying, by whatever means, intramanual couplers affecting flute voices. The Swell already has an 8′ Hohlflute, for instance, and a 4′ Flute, both obtained simply by keying the flute keyers. A Swell To Swell 4′ coupler would have exactly the same effect on the 8′ flute stop as does the presence of the 4′ flute, since both stops are in any case of identical sound. The principle is simply that in a unit organ, or in that part of an organ which uses unit ranks, the unification is itself the substitute for intramanual coupling and is actually better, since with individual tabs for unifying, one can add only flute or only diapason in other pitch registers if desired, while with a coupler tablet *all* tone colors would go up an octave, or down an octave, or whatever the coupler called for.

Exactly the same argument prevails for intermanual coupling. Since the Flute 4′ on the Swell and the Gedeckt 4′ on the Great switch on exactly the same flute tones, an intermanual 8′ coupler could do no more—and would do it less selectively.

Where couplers *are* desirable, of course, are in those sections of the organ which are *not* unified—where each stop speaks with its own tone color and pitch register; and in this organ, that section is the string and reed voices. Restriction of coupling to the 8′ pulse voices of the Swell is definitely done for economy.

The Swell does have strings and reeds at other than 8′ pitch, though not subject to coupling. The Geigen Octave employs tone taken directly from the Swell 4′ pulse preamplifier through 82,000 ohms, with some low-pass effect caused by the .0082-μf capacitor and 15,000-ohm resistor. With the stop switch closed as shown, this tone is shorted to ground. Opening the switch passes it to the Swell string expression control (one of six such controls to be described) and thus turns the stop on. In the same way, an 8′ Geigen Principal is obtained from the Swell 8′ pulse preamplifier. Other stops shown in Fig. 19-9 are obtained in the same way for the Great manual.

While these filters are, for the most part, much simpler than those in the Schober Organ of Chapter 10, the normally closed shunt switching has the same advantages. First, shorting a filter to ground to turn it off assures that none of its tone can leak through to the output, as might happen with the capacitance of a normally open series switch or its wires. More important, switching on a stop adds to the total volume as one should expect, since unshorting the filter (in most cases, at least) adds one more channel through which tone can pass to the output.

Fig. 19-10 shows the 8′ Swell voicing, preceded by a feedback-stabilized amplifier. The only unusual point is the Vox Humana tone, which, when the tab switch opens, goes to two of the channels instead of only one.

Pedals

The pedal key switches are of the same type as those of the manuals—wire fingers and bus rods. The bus rods, however, are all stationary and do not rotate, so that each one makes contact with a finger whenever pedals are pressed. There are six busses for the pedals; one which switches on a rank of 32 16′ diapason keyers especially reserved for the pedals, two more for keying the regular diapason keyers at 4′ and 8′ pitch, two for keying regular flutes at 4′ and 8′, and one for a mixture. Since none of these rods rotate in response to stop tablets, the fingers are all (except for the 16′ special diapasons) connected to the keyers through diodes to prevent interaction. No pulse tones are used for the pedals.

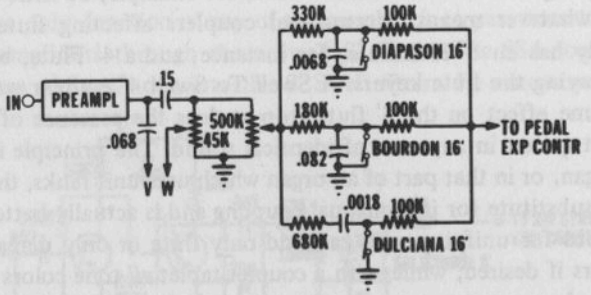


Fig. 19-11—Voicing for three of the four 16′ pedal stops, using sawtooth tones from the pedal "diapason" keyers.

Pedal flute stops at 8′ and 4′ pitch and diapasons at the same pitches are obtained in the same way as for the manuals, simply by keying the manual flute and diapason keyers, from which tones pass to the diapason and flute preamplifiers exactly as though the keying had been done from the manuals.

Four 16′ pedal stops, however, are obtained separately and are voiced by formant filters. In Fig. 19-11, the In point is furnished 16′ signal by the special rank of 32 keyers which handle sawtooth tone from the main generators for the pedals. The preamplifier indicated is the same as the one shown in Fig. 19-10.

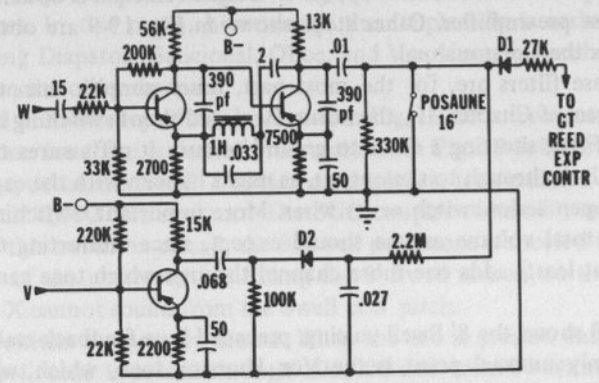


Fig. 19-12—A special preamplifier and gate circuit are provided for the pedal Posaune 16′.

The 500,000-ohm pedal volume adjusting control feeds three filters of the same type as used in the manual pulse voices, though the tone fed through them is, of course, sawtooth for the pedals.

V and W in Fig. 19-11 are takeoffs for the Posaune 16′, a strong pedal reed, which is treated separately, as diagrammed in Fig. 19-12. Signal W, preset in volume with the 45,000-ohm potentiometer in the previous diagram, is fed through a two-stage amplifier whose negative feedback passes through a resonant L-C filter. This imparts a strong formant, so that the output of the second transistor is the fully formed Posaune tone.

There is, however, a problem with noise, so a separate gate is provided for the Posaune. The gate is diode D1. So far as I can see, it is not actually biased off, so presumably the residual noise is not large enough to cause the diode to go past the threshold of conduction or is simply rectified, reducing its amplitude. At any rate, when a pedal is pressed, the signal appearing at point V is amplified by a transistor and rectified by diode D2. Due to the polarity of D2 the result is a negative potential which is applied to the cathode of D1 (through a resistor), making D1 fully conductive. The component values are such that the decay of the negative gating voltage when a pedal key is released takes about the same time as the decay of the note itself due to keyer components.

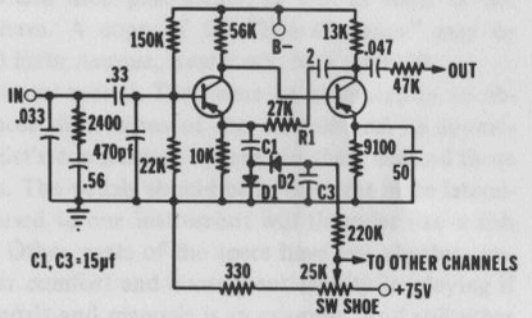


Fig. 19-13—One of six expression controls with shoe potentiometer.

Expression Control

The organ contains six expression control preamplifiers, each handling certain signals and passing them on to one of the three or six line amplifiers representing final output channels. The function of these expression controls, of course, is to vary output of the organ in response to swell shoe operation. There are two swell shoes. One is connected to control all pedal signals plus all Great and all diapason signals. The other handles all flutes and the Swell strings and reeds.

Fig. 19-13 shows one of the six identical expression controls. The swell shoe has the single 25,000-ohm potentiometer, the arm of which goes to five more of the circuits shown.

The amplifier is a fairly ordinary two-stage unit with negative feedback from the second collector to the first emitter through C2 and R1. However, the first emitter has a bypass capacitor C1, which bypasses the emitter resistor if D1 is fully conducting and bypasses it only partially if D1 is not fully conducting.

How much D1 conducts depends on the position of the swell shoe and of the arm of the 25,000-ohm potentiometer.

When the swell shoe is fully in the on position, a maximum positive voltage is applied by its potentiometer through D2 to D1. D1 conducts fully. This bypasses the emitter of the first transistor, thus increasing its gain to the maximum; and at the same time the resulting lowered emitter impedance makes the negative feedback less effective, further increasing amplifier gain. When the shoe is fully off, only 1 volt is applied. D1 conducts very little; C1 does practically no bypassing; and negative feedback is at maximum. Gain is therefore minimum. Capacitor C3 smooths out any transients that may be introduced by irregularities in the swell shoe potentiometer element and even slows the action down to the point where inadvertent "pumping" of the shoe by an inept organist produces little effect.

Appendix

A. G. O. Specifications

The "A.G.O. Specifications" is the common but unofficial name for a document entitled, "Revised A.G.O. Report on Standardization Of The Console," approved by the Council of the Guild December 11, 1961, and signed by the Guild Committee on the Standardization Of The Console, consisting of Alexander Schreiner, George A. Markey, and William H. Barnes, the latter as Chairman. In words and drawings, this single sheet defines preferences of the members of the American Guild of Organists as to the physical dimensions of console parts and their placements, as well as some of the major facilities organs should have. A copy of the "Specifications" may be obtained from the A.G.O. at 630 Fifth Avenue, New York, N. Y., 10020.

The A.G.O. Specifications are not sacred. They were promulgated in an obvious attempt to standardize those dimensions of the console and its appurtenances which involve the organist's conditioned reflexes in their use and those are certainly the important ones. The pedals should be in the right place laterally, for example, or an organist used to one instrument will flounder like a fish when he tries to play another. Other parts of the specs have less absolute importance but will lead to greater comfort and consequent facility in playing if adhered to. Distance between pedals and manuals is an example. And still other points cover facilities most organists would like to have, like what sorts of couplers there should be.

I say that these specs are not sacred because they are by no means followed in complete detail in pipe organs or electronics, nor need they be. When an electronic organ manufacturer refers to a model as "A.G.O." he rarely means that all the specs have been complied with. He means first and mainly that the organ has 61-key manuals and 32 pedals. (Even this is not always true, as he may refer to a 25-pedal organ as A.G.O.) This differentiates it from a "spinet" or other "substandard" configuration, and that differentiation is often the main purpose of the use of the A.G.O. term. A conscientious manufacturer will try to follow those other parts of the specs which make it possible for an organist to go from one console to another without becoming a nervous wreck. Pedal and manual keyboards should be dimensioned and laid out normally, though the pedal clavier does not really have to be concave. The swell shoe should be in about the usual place. And so on. In other words, compliance is and should be a practical matter, not one of dogma.

The following material is my own version of the A.G.O. Specifications. Most of it is taken verbatim from the original, but I have felt free to reword, explain, and delete as best serves, in my opinion, the interests of the readers of this book

and the electronic organ industry at large. At no point, however, have I altered anything substantive. But if you are going to design an "A.G.O." console commercially, I do recommend that you send to the Guild for a copy of the original and read it, too.

Pedal Clavier

Compass: 32 notes, C through G.

Radiation: Ideally 8½′ measured to the fronts of the sharps and 9′ measured to the rears of the sharps. However, deviation from the 8½′ as a maximum is permitted down to a minimum of 7½′.

Heelboard to toeboard distance: 27″. This is measured from the curved edge of the front board to the curved edge of the rear one and is, in effect, the front-rear length of a white key.

Length of playing surface of sharps: 6½″ is specified, but this is inconsistent with the total sharp length as deduced from the radiation specification, which is 6″. The sloping top surface of a sharp could account for a little of this discrepancy, but actually this is just a sample of the general inaccuracy and inconsistency that is manifest from time to time in these specs. Black plastic sharps obtained from Organ Supply Corp. and widely used in the industry actually measure a little over 5″.

Height of sharps above naturals: 1″ at player's end, ½″ more at the other.

Width of playing surface of natural keys: 7/8″ to 15/16″. The top surfaces of these keys always are curved, but this is not mentioned or specified.

Distance, center to center, of adjacent natural keys at player's end of sharps: 2½″. This makes the octave 17½″, which is the length of a line drawn between centers of two keys an octave apart, the line being curved to conform with the arc described by the sharp fronts. This is the most important of all dimensions of the pedal clavier and must be adhered to for obvious reasons.

Depth of touch: ½″ at player's end of sharps. This is the total vertical travel of each key.

Weight of touch: 2½ to 3 lbs at player's end of sharps. Not mentioned is that key movement should be as nearly frictionless as possible and springing should make each key snap back smartly when released. Horizontal slop should be as little as possible, and keys should exhibit no noticeable tendency to twist.

Firing point of action: Midway between top and bottom of travel. This is the point at which electrical keying should take place. This gives only ¼″ of travel to the firing point and makes the playing a bit touchy for many organists. It is not unusual to have greater travel than the ½″ specified above, but the firing point should still be about halfway down.

Pedal To Manual

Left-right location: The pedals should be centralized under the manuals laterally. This is generally checked as "D under D." That is, the second D pedal is directly under the middle D of the manuals. With a 25-note clavier, the "D under D" should still be observed, if possible, so that organists accustomed to the 32-

note clavier will still find the existing pedals in the right places. This results in some blank space at the treble end in most cases, which is all right. Observe Fig. 10-1, page 180.

Front to back: Player's end of the second pedal D# key should be 8½″ to 10″ back from a plumb line dropped from the front edges of the manual white keys on a two- or three-manual console, 11″ on a four-manual.

Vertical: 29½″ between playing surface of white keys of lowest manual and middle natural key of the pedal clavier.

Pedal Accessories

Knee panel and toe board: Should follow the 9′ radius of the rear ends of the pedal sharps, but it usually does not and is straight instead. Its bottom edge, however, usually is cut to conform with the 8½′ radius of concavity so that the clavier will fit under it.

Swell and crescendo shoes: May be placed as far forward as will permit the heel end of the shoe surface to overhang the sharp pedals as much as 1¼″ or so far back that this point is ¾″ back of the sharps, the distance measured with the shoe in closed position. If there is only one shoe it should be directly back of the E-F pedal gap. With more than one shoe, the Swell shoe is at the E-F gap, the Choir shoe (or the Great shoe in small two-manual organs) to left of the Swell, the Solo shoe to right of the Swell. A crescendo shoe is to right of all others and slightly higher.

Bench: The A.G.O. would like a bench with a normal height of 20½″ above the middle pedal E, adjustable 2″ up or down "by means of acme thread screw and ball crank." Let me know next time you see one of these, but I won't hold my breath waiting.

Manuals

Compass: 61 notes, C through C. See Fig. 2-2, page 17, for key shape.

Overhang of keys: The front edges of the white keys of each manual should be 4½″ back of the front edges of the white keys of the next lower manual. The playing surfaces of the white keys should be 2½″ above those of the manual below. This is subject to slight variation without causing trouble, but the dimensions given are good goals. Usually the vertical distance has to be exceeded somewhat because it does not give enough space for keyboard frames and switch assemblies. Many organists like the Great key surfaces to be level and all other manuals to slope gently toward the Great, up or down as the case may be. This is much more important and usual in organs of three or more manuals than in two-manual organs, where it is both unusual and unnecessary.

Depth of touch: 5/16″ to "3/8″ scant." Firing point should be halfway down.

Weight of touch: 3¼ to 3¾ oz is recommended. This is a reflection of the preference of many very serious organists for the "tracker touch" afforded by pipe organs with mechanical rather than electrical actions, as evidenced by the fact that "tracker feeling" is actually recommended. The weight given is too much for the average organist to appreciate, and electronic organs almost in-

variably have lighter touch. Avoid feather lightness, however. It is difficult to play with accuracy when the fingers have little or no force to work against. The same is true of pedal pressure.

Miscellaneous

Order of stops: On tongue-tablet consoles, two rows of stops are permitted above the top manual. Left to right in the top row are Swell, Choir, Solo or Echo, and in the bottom row Pedal, Great, Positiv. Most electronic organs have only two manuals so that a single row of stop tabs is adequate. Normal order is Pedal, Swell, Great. The A.G.O. looks through theatre organs as though they did not exist. The convention on theatre organs is Pedals, Accompaniment, Solo. Within each group, the A.G.O. specifies flues by pitch register, lowest register to highest, mixtures, then reeds by register. Within pitch groups, stops run loudest to softest. All this is left to right. In electronic organs, but also in many pipe instruments, all this is more often honored in the breach. In all cases (on tongue-tab consoles, which is all we need be concerned with) coupler tabs follow other tabs at the right of each group.

Combination action "pistons" (buttons): Where there are 8 or more pistons for each manual, they should start under the middle C of the manual and be spaced $1\frac{1}{8}''$ apart. Nothing is said about setups with fewer pistons, but using the same rule works well. General pistons should be to the left of the manual pistons and should be duplicated with toe studs. Pistons affecting pedals only should have no buttons but only toe studs, at right of the shoes. General toe studs are at left of the shoes.

Over-all console dimensions: This subject is not mentioned by the A.G.O. but is an important practical matter in two respects. Most vital for designers of organs which might under any circumstances ever have to be brought into homes or other "normal" buildings is the knowledge that the standard doorway in the United States measures 30" and very few people are willing to tear out the side of a house to get an organ in. In a full size two-manual organ it is usually entirely possible to keep the maximum front-back dimension (exclusive of pedals, which should be removable) to 29 inches. Second, as a generality again applying to instruments for the home, houses and apartments are smaller than in Grandma's day, and any housewife to whom you try to sell an organ that is larger than it absolutely must be will let you know that fact in no uncertain terms.

A comment on the A.G.O.: The Guild still appears to be unaware that the electronic organ exists and is here to stay, judging by its activities and publications. Its individual members do not share this ostrichism, for many thousands of them play electronics every Sunday. I would like to see these Specifications drastically revised by a more contemporary committee. They should include more practical and feasible points that would help standardize church-style electronics, for which both manufacturers and organists would be grateful. The electronic organ is no longer an unmentionable toy. It is respectable and the A.G.O. should admit it and do something constructive about it.

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(continued from front flap)

ics a thorough understanding of the objectives of organ design and the many methods by which they are achieved.

To acquaint the reader with the job an organ does, the text begins with a chapter on the nature of music and one explaining in detail how an organ differs from other instruments. Six following chapters cover detailed requirements of and approaches to tone generation, keying and coupling, tone coloring, special effects, sound production and enhancement, and tuning and servicing.

Eleven chapters are devoted to specific organs of various makes, dissecting each one in detail to show exactly how each of its functions is performed. The reader finishes with an understanding, not only of *what* has been done, but *why.*

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