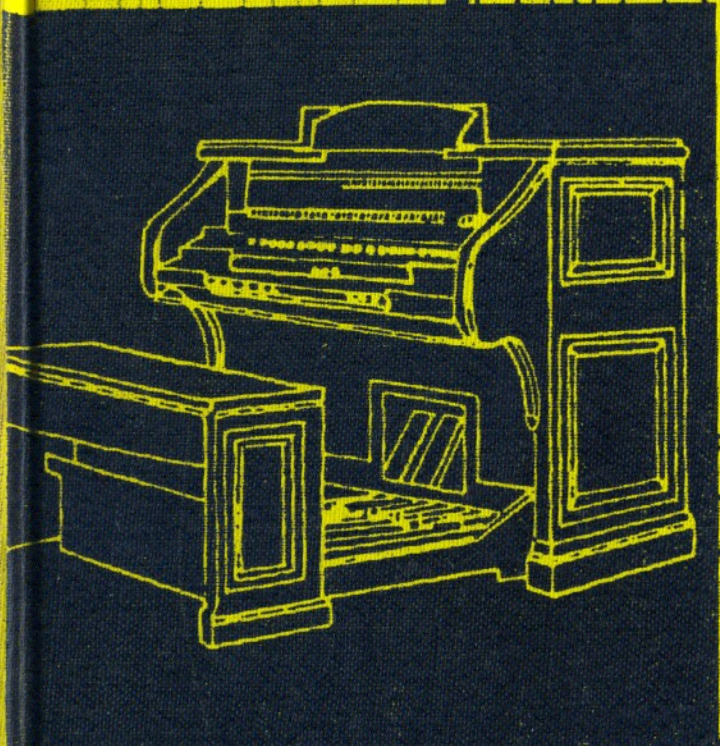
RICHARD H. DORF



ELECTRONIC MUSICAL INSTRUMENTS

SECOND EDITION

Electronic Musical Instruments

Second Edition

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Contents

Introduction

My recent series of articles on electronic musical instruments in *Radio-Electronics* magazine and my reviews of patents on musical instruments in *Audio* continue to bring a flood of mail, proving beyond doubt that electronic music is a subject of great – and sometimes time consuming – interest to a great many people. Since the subject has never been fully covered before in book form in the light of present-day practice, this book has been prepared to provide data for those interested in the particular field.

The first section of the book contains a very general discussion of what is needed in an electronic musical instrument and why. This is followed by very detailed descriptions of almost every commercial instrument on the market today. This section serves not only to describe given products but, more important, it explains the principles and possibilities of instrument design by multiple example. Because the descriptions contain complete schematic diagrams, photographs, and other explanatory illustrations and tables, and because the reason for each point of design is given, it is believed that the section on commercial instruments is a more useful and practical guide to future original design than an abstract instructional text would be.

The final section of the book is devoted to helping the reader design and construct his own instruments. Chapters are devoted to actual circuits taken from practice, from patents, and from the writer's own experiments, and these circuits can be incorporated into designs. Most welcome to many readers will be the two chapters giving construction data for two complete instruments designed by the author, one a monophonic type and the other a complete organ.

Various readers will have different goals in reading this book. In addition to those who wish to build and service instruments, many are simply curious about how these fascinating devices work. Others

– musicians (both one-finger and educated) and members of church, association, and other organ purchase committees – will want to compare the merits of the available instruments on a sound basis in the absence of a salestalk. While the makers of each of the instruments described have cooperated fully in supplying information and photographs, not one has attempted to censor any comments or expected to receive a specially favorable write-up. Since the material in this book is thoroughly unbiased and objective, the impressions the reader receives of the instruments should be as fair and truthful as it is possible to make them. Though the field is highly competitive, no two instruments have the same features and none can claim simply "higher quality." This makes it possible for the reader to select from a large variety on the basis of his own requirements, for *all* the instruments are well made and of high quality.

Grateful appreciation is extended to all the cooperating manufacturers, who have, without exception, gone out of their way to help make this material complete and authoritative.

INTRODUCTION TO THE SECOND EDITION:

Industry sales figures show that interest in electronic organs has grown even faster than we anticipated when the first edition was published in 1955. The extraordinary growth of activity with Schober Organ Kits, which the author designed, is another indication of this interest.

The revisions in this new edition are not wholesale – no really new principles have appeared (except for the new neon-generator circuit used by Kinsman, which has made commercial success of a neon organ possible for the first time). Since the prime purpose of this book is to acquaint readers with principles and practicalities, we have described new models of Minshall and Conn (no longer Connsonata) instead of the older and less useful ones, and substituted a thorough description of the Schober Kit Organ, a completely satisfactory and reliable instrument, for the old chapter on the early Electronorgan, which was essentially an experimental project.

No instrument or principle can be judged as to musical value just by looking over diagrams and descriptions. So whether you want to buy or build, visit dealers in the various brands whenever possible and actually listen to the instruments or play them to help your decision. Possession of the information in this book will give you essential aid in question asking, judging construction, understanding, and servicing. But never forget that the end results of your labors and those of the men who work in this field must be a musical instrument – not a technical curiosity.

Richard H. Dorf.

Chapter 1

The Nature of Music and Musical Instruments

The function of any electronic musical instrument is to provide the musician with an effective medium of musical expression. In this it differs not at all from the traditional acoustic-mechanical instruments. The fact that it may have loudspeakers and *R-L-C* tone filters rather than *f*-holes and soundboards does not give it license to excuse poor musicality by pleading electronic ingenuity. It must be first a musical instrument and only second attention may be paid to the elegance of its design or its cheapness compared to its acoustic-mechanical counterpart.

The technology of electronics gives us many advantages over traditional musical instruments in designing both new and imitative media for musical artistry. This is because electronics is able to act as an analog or simulator of almost all the qualities of ordinary instruments. An obvious example is the loudspeaker and audio amplifier which allow us to control the volume of emitted sound to simulate that of any instrument from the flute to the pipe organ. Another set is the electrical switch and the audio oscillator which allow us to generate and select in the desired order tones of any pitches we like without using the strings and hammer action of a piano or the mouthpiece and valves of a trumpet. Throughout this book you will see how electronics takes the place of mechanical instrumentalities to produce the same end result.

To do all this electronically is interesting for its own sake, but interest in electronics is only one reason for working with electronic music. Another is that, especially for complex instruments like the organ, the organ effects can be achieved much more cheaply with tubes, resistors, capacitors, and switches – a ratio of as much as 10 to 1 in cost. Space is conserved as well; people who live in ordinary homes (especially city apartments!) could hardly have a pipe organ in the living room. But an electronic organ need take no more space than a spinet piano. Electronic music opens the way to musical expression for people who have little if any musical training; instruments such as the writer's Thyratone, and the Hammond Solovox and Chord Organ (all described later in these pages) produce large varieties of pleasing effects even for those with strictly one-finger ability. As another attraction, ordinary musical instruments can hardly be built by those not specially trained; but electronic ones can be built by anyone with some training in electronics.

It is not enough, however, to create a device which will furnish tones of the correct pitches on demand; this does not constitute musical expression. So let us, before we examine the instruments in detail, see what it is we are after.

MUSIC AS AN ART

Music is an art, which is to say that it is not easily definable in nonphysical terms. We must begin with these terms, however, because the desired end result of an instrument is more than just the production of tones. This can be likened to the phonograph which must do more than vibrate the speaker cone. It must faithfully reproduce the material recorded on the record in order to please the listener.

Music is, so far as we can define it, a succession of tones, usually (though not always) of definite pitches, sounded sometimes singly and sometimes in simultaneous groups, in such a way as to please the listener by appealing to his emotions. Music has melody – sounding of tones in definite order designed to be pleasing – and it has rhythm – sounds made with definite (though not necessarily constant) intervals between accented points. It has over-all form and organization and the sounds themselves have various characters or timbres. It is impossible to make up any kind of formula showing how the various elements must be assembled, for the variety of assembly is infinite. It is impossible to define a "pleasing" or "unpleasing" assemly of the elements, for these factors depend on time, place, desired effect, and the tastes of the listener.

It is easy to see from the above that music is really not definable. Probably the closest we can get to pinning it down is to say that it is the art of affecting the emotions by the use of sounds which have no specific intrinsic meaning.

Certain factors must be available to the musical composer and performer, however. The most important is variety. It must be possible to vary pitch, rhythm, volume, and tone quality or timbre in the course of every musical selection. A few instruments – such as certain percussion instruments – cannot vary in some of these respects and consequently they are never used alone in music. Other instruments such as the oboe or the double-bass have some but very little variety and they are rarely used alone. Still others, like the violin and the human voice, are capable of great variety, and they are outstanding solo instruments. The "instruments" with the greatest variety are the ensembles – orchestras, for instance – and they are by far the most often heard vehicles of musical expression.

The simplest organ-like electronic instrument would be an assembly of code-practice oscillators tuned to the musical pitches, with switches for keying and an amplifier and loudspeaker. Without much further modification this assembly is only a curiosity, not a musical instrument. The chief reason? Lack of variety. Add a tremolo or vibrato and we are a little better off because at least the pitch or volume of a note will vary a few times a second. Add some selectable filters for obtaining various tone colors and we are a long stride ahead. These are all variety factors and they are tremendously important to the art of music.

THE PHYSICS OF MUSIC

Once we have some idea of what we want to achieve in an instrument, we can begin to consider going about it. This brings us to the realm of more concrete definitions because both the tools of the artist (tones) and the tools of the instrument maker (electronic and mechanical components) are precise and can be manipulated. For a full discussion of musical physics see Harry F. Olson's excellent book *Musical Engineering*, (McGraw-Hill Book Co., 1952). In the following we will talk briefly about such physical knowledge of music as we need for our specific purpose.

From our viewpoint we are most concerned with tones, and with the five important characteristics we must give them: frequency or pitch, amplitude or intensity, form or color, duration, and envelope of attack and decay. We will take them up one by one.

1. *Frequency*. Theoretically, a tone can be produced and heard at any frequency within the range of human hearing – about 10 to over 16,000 cycles per second, depending on the individual. But for the sake of practicality and to impart some organization to music, certain specific frequencies are commonly used, while the other frequencies are considered unmusical. We will speak more of these in the next chapter. Melodies are made by employing a number of tones with different frequencies. This term *frequency* is a physical

one, while *pitch* describes a physiological-psychological effect on humans caused by the frequency of air-pressure variations against the eardrum. They may usually be used interchangeably but we will stick to frequency because it is applicable to all the technical factors with which we will deal.

2. *Amplitude.* Amplitude is the characteristic of a tone which controls its loudness in the ear of the hearer. Here again, *amplitude* is the physical quantity which can be measured – by an oscilloscope, voltmeter, sound level meter, or other means – while *loudness* or *intensity* is the effect on the ear. Amplitude and loudness are not directly proportional. The human ear senses loudness (as it does pitch) in a logarithmic manner. Each time actual sound power is doubled, for example, the ear senses an equal loudness increase, whether the sound power increases from 3 to 6 watts or 300 to 600. The decibel notation, a logarithmic system, was developed as a convenient way of expressing sound power changes in terms of ear response rather than measured amplitude. Apparent loudness also varies with pitch for constant tone amplitude. The normal human ear is capable of listening to sounds with a ratio of maximum to minimum amplitude (power) of 1,000,000,000,000 to 1, a truly astonishing range between the threshold of hearing and the pain level. A symphony orchestra actually has a sound power range of about 10,000,000 to 1, or 70 decibels. Amplitude is one factor which is varied to add to the interest and appeal of music.

3. *Color.* Almost any tone consists not only of one frequency, the fundamental, but also harmonics of the fundamental in varying degree. This is inevitable in even a high-quality signal generator, but is true deliberately in music because a pure tone – one consisting only of fundamental – is devoid of interest or expression. To prove this to yourself, think of the output of an audio test oscillator as a musical tone, then compare it mentally with the tone of an oboe with its sharp, biting quality.

Variation of tone color is obtained by deliberately adding harmonic content to the fundamental-frequency tone in one of a number of methods described in later chapters. In an electronic instrument the coloring is done electrically; in standard instruments harmonic content is controlled by the shape of the instrument, character and dimensions of the tone producer (reed, string, etc.), method of playing, and so on. While the organ is capable of the widest tone variation, even ordinary instruments such as violin, trumpet, flute, etc., are capable of some or much tone-color gradation. The sole common exception is the piano, which is, like the organ, a mechanical instrument but has not the organ's resources. (For verification of this rather con-

troversial point see *Pianos, Pianists and Sonics,* by G. A. Briggs, British Industries Corp., Port Washington, N. Y.) For details on tone-coloring see Chapter 4 and the instrument descriptions.

4. *Duration.* This refers to the length of time a tone lasts once it is produced. The tone of a piano or guitar is of comparatively short and not very controllable duration. That of a bowed violin or organ can be as long as desired. The duration of the tone of a wind instrument depends on the lung capacity of the player. In electronic music we deal almost entirely with tones of indefinite duration which will last as long as we keep the key or switch in operation.

5. *Envelope.* This is most quickly explained by the drawing of Fig. 1-1. (A) is the envelope of a code-practice oscillator. As the key is pressed the tone in the headphones rises from zero to maximum volume almost instantly. This is the *attack* and it is very fast as shown

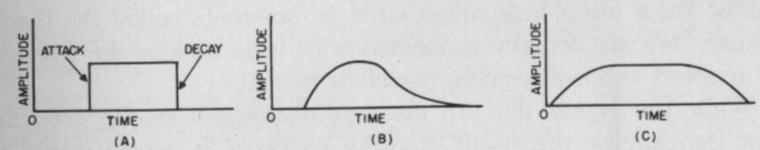

Fig. 1-1. Envelope of a switched tone (A) compared to that of a piano (B) and an organ (C).

by the steep, straight side of the envelope. The tone continues at constant level as long as the key is held down. When it is released the tone falls to zero just as fast as it rose to maximum during the attack. The fall is the *decay*, which is again extremely short.

Such an envelope is unmusical (except in rare instances) and does not please most listeners. The envelope of the piano, shown at (B) in Fig. 1-1, is better. When the key is struck the sound amplitude rises rather quickly but by no means instantly. After reaching highest amplitude it begins to die away and does so very slowly (assuming the damper pedal is held down). Thus the piano envelope has a fairly fast attack and a very slow decay.

At (C) is the envelope of an organ tone. Both attack and decay are fairly slow. They may appear fast to the ear because it is accustomed to hearing envelopes of this kind in music. But just compare the effect with that of a keyed code oscillator and the difference will become very apparent. In electronic music some provision must be made to prevent attack from being instantaneous as at (A). This provision is known as attack delay. Preferably the same should be done for decay but it is less critical. Omission of attack (and decay) delay also causes a click effect because the steep envelope edge contains large high-frequency components.

Chapter 2

Basic Musical Facts

In order to construct an instrument, we must know just a little about what is sometimes called the theory of music. We are actually concerned with little besides the frequencies involved and will confine ourselves to that.

While the human ear can instantly distinguish among 1500 different frequencies, the result of using so many in music would be chaos – not because it would sound bad but because instruments capable of so many different tones would not be practical. As a matter of convention, therefore, it is customary to use only certain selected frequencies comprising the notes in the scale. While we shall discuss the diatonic tempered scale used for most Western music, keep in mind that it is not the only one in use. In China or India, for example, entirely different scales are used, probably with more notes; such scales sound every bit as natural and musical to their listeners as the Western one does to us.

Figure 2-1 is a drawing of a piano keyboard with the full 88 notes. At each note its frequency is given to four significant figures. While the actual frequency is much more precise, four figures is more than enough since any instrument must have adjustable tuning and the final frequency setting must always be done by ear or by comparison with a standard.

The basis of the scale is the *octave*. This is the interval between any note and the next note above or below it which "sounds similar." The frequency ratio between any two notes an octave apart is 2 or ½. The octave is important because of the psychological fact that the human brain senses a peculiar identity between two notes with these frequency ratios, even though their actual pitches are different.

The tones of the scale are identified by letters from *A* through *G*. The letter identifying the eight note of a progression up or down the

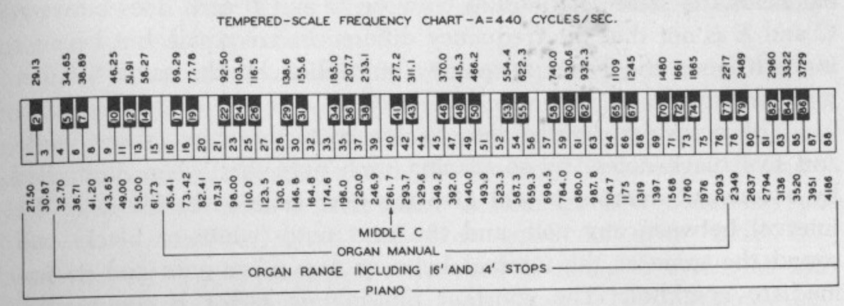

Fig. 2-1. This diagram shows frequencies of all notes.

scale (on the white keys) is a repetition of that of the first. Thus, a scale beginning at *C* includes *D*, *E*, *F*, *G*, *A*, and *B*, and ends at the *C* above the first note played. This upper *C* has a frequency exactly twice that of the first one and since it is the eight note, it is known as the octave.

The black keys are known as accidentals. A black key is a *sharp* if it is above the white key whose letter is used. Thus key No. 41 is *C*-sharp, or as it is commonly written *C#*. A black key is a flat if immediately below the white key whose letter is used. Therefore key No. 41 is also called *D*-flat, or in musical notation D^b. For our purposes it is immaterial what we call a particular black key, since the difference between a sharp and a flat is a matter of musical theory which is artistic rather than physical* and with which we need not be concerned. As a matter of convention and speed in writing we will use the sharp or # throughout this book. There are, then, five black keys – *C#*, *D#*, *F#*, *G#*, and *A#*.

We will also ignore the problem of why the spacing of black notes is irregular within one octave, except to remark that it is fortunate, for it allows us to identify identifical notes in different octaves by noting their positions relative to the black keys. For instance, key No. 32 is *E*. We can see that key No. 44 is also *E*, for it is located just above a group of two black keys. We can also identify identical black keys by their relation to the others.

The relationships of musical frequencies must be thought of in terms of *intervals* rather than frequency differences. The ear perceives pitch as a logarithmic relationship, just as it does loudness. The reason, therefore, for the identity of feeling between C^{40} (Fig. 2-1) and C^{52} is not because there is a frequency difference of so many cycles but because one is twice the frequency of the other. The reason the

* When, as is always true of keyboard instruments, the tempered scale is used.

ear hears the same relationship between *G* and *B* as it does between *C* and *E* is not that the frequency differences are equal, but because in both cases the lower frequency multiplied by the same number gives the upper.

The octave is divided into twelve parts therefore (seven white and five black notes) by so placing each note that when multiplied by a constant number it will give the next in line. In this way the interval between any note and the next note (white or black) will sound the same as the interval between any other note and its immediate neighbor. The constant multiplying factor is the twelfth root of 2 or approximately 1.05946309.

The musical terms referring to pitch or frequency which we shall have occasion to use are few and simple:

1. *Half-tone.* The interval between any note and its immediate neighbor, black or white. The interval between *G* and *G#* is a half-tone, as is also that between *E* and *F* or *B* and *C*. (Note that we say half-*tone,* not half-note, which is a musical indication concerning the duration of a note. We are also careful to use a hyphen between half and tone to avoid confusion with the sort of halftone which is a printed illustration.)

2. *Tone or whole tone.* An interval equal to two half-tones, for example between *C* and *D* or *B* and *C#*.

3. *Octave.* The interval between any note and the note above it whose frequency is double or that below it whose frequency is half – for example between $A\#^{26}$ (Fig. 2-1) and $A\#^{38}$ or $A\#^{14}$.

4. *Fundamental.* The lowest frequency present in a tone which includes a number of harmonic components.

5. *Harmonic.* A frequency which is an integral multiple of another frequency. For example, 400 cycles is a harmonic of 200, and also of 100, 50, and 25, among others. The order of harmonic is obtained by dividing its frequency by that of the fundamental. For example, 400 cycles is the second harmonic of 200 ($400/200 = 2$), the fourth harmonic of 100, and the eighth harmonic of 50 cycles.

6. *Overtones and partials.* We shall avoid these terms as much as possible because they are of little use in electronic instruments. They refer to components of a tone which may or may not be actual exact harmonics. They almost never occur in electronic music unless they are exact harmonics, or are used as harmonics even though not, for some irrelevant reason, precise.

7. *Frequency tolerance.* This refers to the precision with which a tone generator must be tuned. It is determined ideally by the pitch discrimination of the most acute listener, but in practice it is of course

determined by the person who tunes it unless a standard and automatic comparator are used.

8. *Pitch discrimination.* This refers to the ability of an individual to discriminate between two pitches. The greatest pitch discrimination is said to exist in the person who can distinguish between two successively sounded pitches with the smallest frequency difference. The pitch discrimination of listeners determines how accurately an instrument must be tuned and how small its error must be. Average individuals can tell a difference of about 3 cycles at 435 cycles, which is 0.69%. This is 1/17 tone (about 1/17 the interval between middle *A* and *B*, for instance). Sensitive ears can detect intervals of about 1/2 cycle at the same frequency, which is about 0.115%. In a test made on a group of excellent musicians, the best score was 0.1 cycle, or .023%*. For single tones, the allowable tolerance is much greater than this indicates because there is no comparison tone. But in practice we almost always find two tones or more being played simultaneously for purposes of harmony, in which case the same test is effectively made. Since tolerances of a group of notes tend to accumulate, it is advisable to build instruments in which frequency tolerance is kept below 0.25%, preferably down to 0.1%. This includes drift, of course.

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* Seashore, *Psychology of Music,* McGraw-Hill Book Co., New York.

Chapter 3

Requirements of an Instrument

In the last two chapters we have seen a little of the background; in this chapter we shall get down to business and enumerate the factors which go to make up a successful electronic musical instrument.

First, let us define what we are talking about in this chapter and throughout the rest of the book. For our purposes (whether or not a lexicographer would sustain the definition), an electronic musical instrument is a device with which a musician can produce music and in which the technology of electronics plays a major part. It does *not* include so-called electric guitars, for instance, to which an electronic system merely adds volume and perhaps a few special effects, nor does it include electronic chime systems, where the music is produced by actual chimes and simply amplified electronically. But it *does* include such amplified-reed organs as the Wurlitzer because without the electronic system the instrument would be unusable; and it does include instruments in which the tones are produced electromagnetically or electrostatically, such as the Hammond and Compton organs, for the same reason. The most obvious application of the term is, of course, to instruments in which everything from tone generation up is done electronically such as the Baldwin and Minshall organs and others.

Despite popular misconception an electronic musical instrument is not necessarily an organ. Two types of instruments exist, the monophonic and the polyphonic. An organ is a polyphonic instrument, meaning that several tones may be sounded simultaneously. An instrument like the Hammond Solovox, for example, is monophonic, since only one tone at a time can be produced. The Solovox is not, of course, an organ.

Every electronic instrument requires certain sections: a tone-

generator system, a keying system, a volume-controlling or expression system, a tone-coloring system, a general control system, an amplifying system, and a radiating system. Actual construction information for a variety of each will be found in Chapters 16 through 21 and the commercial instruments in Chapters 5 through 14 are described in enough detail to permit full or partial construction by those with the necessary skill and ambition. Here we shall look into the general requirements and considerations.

1. *Tone generators.* The prime requirement for an instrument is that it supply all the necessary tones within the frequency tolerances discussed in the last chapter. A monophonic instrument may have only one or a few generators with variable frequencies. A polyphonic instrument such as an organ must usually have a generator for each tone it is to produce; and this may run to as many as 84 separate generators for an instrument with a five-octave range plus 4- and 16-foot couplers.

The generators may be of any of a number of types. A purely electronic instrument usually employs vacuum-tube oscillators. The oscillators (if more than one) may be separately tuned; or there may be one (in monophonic) or 12 (in polyphonic instruments) master oscillators which are very accurately tuned to the highest notes to be played, with the remainder of the oscillators (or frequency dividers) synchronized with them by some method of signal injection.

Other instruments use partially mechanical generators, such as rotating tone wheels with electromagnetic, photoelectric, or electrostatic pickup. In such cases there is usually a separate generator for each note; the wheels are all driven from a common motor with the geometry of each wheel and its speed (geared up or down) governing its frequency.

The choice of generator ties in with choice of the method of tone coloring, and to some extent with the choice of keying system. For instance, where harmonic synthesis is to be used (see following comments on tone coloring) sine-wave generators are usually wanted. Where formant filters are used, more complex generator waveforms are desirable. The principle effect of the keying system on the choice of generators is concerned with the output impedance of the generators.

2. *Keying system.* There must be some method of selecting the tone or tones desired at any particular time. Most electronic instruments have keyboards similar in kind, though not necessarily in size, to those of the organ or piano. The keys operate switches with sometimes as many as nine poles or sections.

3. *Expression system.* Organs always have so-called swell pedals

which, in acoustic instruments, operate a set of shutters over the open side of the chamber in which the pipes are located. In electronic organs the swell pedal or pedals operate more or less conventional electronic volume controls which are placed at a convenient point in the signal path. There may be a single swell pedal controlling the volume of the entire instrument; or there may be more, each controlling the output of a given section. Similar arrangements for volume control are contained in instruments other than organs. Sometimes a lever is provided to be operated by knee pressure. Only very rarely can volume be controlled by pressure on keys as it is in the piano.

4. *Tone-coloring system.* The tone-coloring system is an outstanding characteristic of electronic musical instruments and may well determine the success or failure of a particular one. The subject is discussed in detail in the next chapter. Meanwhile, we can say that the methods include electronic filters, complex switching systems which automatically mix together various tones when a key is pressed, and others. Sometimes the tone-coloring is a part of the generator proper; in a photoelectric instrument, for example, the rotating wheels may contain several different patterns for each note with a method of selecting which pattern will be transmitted to the amplifier when the key is pressed. Similar means may be used with electrostatic generators, as will be shown in the description of the Compton Electrone.

5. *General control system.* With all the elements in an instrument there must be an adequate method of controlling them. The keys themselves are part of the control system as is the volume control. Other controls include switches which perform the function of organ stop tablets or knobs to control the tone color, crescendo pedals to select in sequence the necessary tone colors and levels to build up gradually to "full organ," combination controls to select one of several groups of tone colors in response to pressure on a single button, speaker and amplifier switches, and so on. All must be designed for maximum convenience and flexibility of operation and in many cases – especially where an electronic organ must stimulate in appearance a conventional console – switches must be made to operate by unconventional means such as draw knobs, flat tablets, and the like.

6. *Amplifying system.* Electronic musical instruments call for fairly conventional amplifiers to raise the level of power so that it can drive one or more speakers adequately. Precise frequency fidelity is not a usual requirement of such amplifiers, but excellent power-handling capacity and low distortion are. Very good bass power response is necessary for organs, since a fundamental tone may go as low as about 32 cycles and must be available at full volume. Higher

power rating is required than is the case for home phonograph amplifiers because an instrument may be used in an auditorium and even in a home it will not give the desired bass power response and "body" of tone without adequate power. Amplifiers under 20 watts are rarely used.

7. *Radiating system.* The prime ingredient of the radiating system is, of course, one or more loudspeakers. The speakers must have adequate power-handling ability and should have excellent low-frequency response, even if at the expense of treble. Multiple speakers are desirable because in a musical instrument the "point source" effect of a single speaker is much more annoying than in radio or record reproduction. It is common to use two or four 15-inch speakers in a typical installation, with a high-frequency speaker sometimes added for brilliance.

The speakers should be mounted in totally enclosed cabinets (all are usually mounted in the same cabinet) with a large interior volume to give the bass full rein. To get away still further from the point source effect, the speakers are often faced toward a smooth, hard wall rather than toward the listeners; this gives a diffusing effect. Pointing them toward a right-angle corner is better yet. Where the installation allows it, a special chamber should be provided, with hard, reverberant walls and just one open side. The sound has a chance to bounce around in the chamber to some extent before emerging into the auditorium. The slight loss of clarity is beneficial and tends to get rid of the "electronic sound" inherent in most instruments.

8. *Vibrato or tremolo.* Vibrato or tremolo is the variation of pitch or volume of a tone which takes place at a rate of 5 to 8 cycles per second. It is deliberately employed by almost all singers and musicians to add interest and color. Tremolo is a variation in volume similar to amplitude modulation. The amplitude of the musical tone, whatever it may be, is varied at the tremolo rate of 5 to 8 cycles. Vibrato is a variation in frequency similar to FM. The tone pitch is made to vary slightly above and below its true value at the 5- to 8-cycle rate. Of the two, vibrato is probably more pleasing to most people. It can be obtained in any electronic tone generator (oscillator), either by placing a reactance tube across the oscillator tank and modulating its grid with 5- to 8-cycle sine wave, or by varying the supply voltage applied to an oscillator tube element at the required rate. Where the frequency of the generators themselves cannot be varied – as is true of non-oscillating generators – tremolo must usually be used. This can be done by modulating in phase both grids of a push-pull amplifier stage with a 5- to 8-cycle sine wave.

Chapter 4

Tone Color

The most distinctive attribute of any musical instrument, traditional or electronic, is its tone quality. There are other qualities, of course, such as its attack and decay envelope, its frequency range, and so on. But the principal reason for such a great variety of instruments is that each has a certain distinctive tone quality.

Let us look at some of the best known. The trumpet has a brassy sound – it blares obtrusively. Yet the French horn, a member of the brass family, has a muted sort of tone, rather smooth and formal. The saxophone (a reed instrument) is much smoother, with hardly any of the whine effect that can be heard in the oboe, another reed instrument. The stringed instruments of the violin family can sound sharp or relatively smooth according to the player's desire; but the violin and the viola, which belong to the same family, look almost the same, and can cover much of the same frequency range. They are easily told apart because of a marked difference in the way they sound.

This variety in sound qualities accounts for the sustained interest of orchestral and ensemble music. Many people who have heard a little about Tin Pan Alley think that the great composers merely wrote piano scores, then hired arrangers to convert the melodies and harmonies to orchestral form. Nothing could be further from the truth (except in a very few isolated instances), for one of the outstanding characteristics of great ensemble music is the artistic use of the various instrumental tone qualities. Each tone quality and each combination of them conveys quite a different impression to the hearer; and since music is really nothing but a series of impressions, the particular instruments are every bit as important as the melody, harmony, and rhythm. To prove this to yourself listen to someone

play a symphony theme on the piano. Your impression of the music will be greatly different (and less favorable) than when an orchestra plays it in all its variety of tone coloration.

In electronic musical instruments, tone quality is a prime factor in making an instrument good, passable, or bad. To be more accurate, an organ-like instrument – even a monophonic one (as compared with the multiple "voices" of a large organ) – must be capable of variation in tone color. It follows, therefore, that a circuit capable of producing the notes of the scale but incapable of producing a number of interesting and pleasing tone colors is of little or no value.

There are, as far as tone is concerned, two general types of electronic musical instruments. The first does not try to imitate the tones of ordinary acoustic instruments particularly; the Hammond organ is an example. The second tries to reproduce as closely as possible the traditional tones of the organ, as does the Baldwin organ. There are several other commercial instruments (which we shall describe in the chapters to follow) which compromise between the two classes – they are imitative to some extent but not entirely.

HOW MUSICAL INSTRUMENTS WORK

Some of the earliest work done in scientific circles to analyze instrumental sounds was carried on by Helmholtz, who worked on acoustic resonance. Helmholtz developed some of the first wave analyzers with which he analyzed sounds and found what they contained in the way of harmonic structure.

He did this by constructing small enclosures, each of which was acoustically resonant at a certain frequency. We can get roughly the same results by blowing at the edge of a bottle and producing that steamboat-whistle sound. We find that we can change the pitch of the sound by filling the bottle with a liquid and that height or lowness of pitch depends on how full the bottle is. We find that as we fill it more and more the pitch becomes higher and higher. This is a scientific experiment which shows us that a container is acoustically resonant at a certain frequency. The frequency depends on the size of the enclosure. It becomes higher as the enclosure becomes smaller.

The resonance effect is exactly analogous to the electrical resonance we obtain from a coil and capacitor or from a piece of wire or transmission line cut to a certain length.

Acoustically, resonance is not limited to enclosures. If we strike a bar or tube of metal, or a bell, we often get a sound of one pitch because the makeup of the material is such that it can vibrate at a certain optimum rate. A stretched string is also resonant at a frequency depending on its length, physical makeup, and tightness, so that when it is plucked or struck it gives off a certain tone.

Every finite object has at least one resonant frequency. It may not be apparent if the object is very large and heavy, for then the frequency is below audibility. Also the frequency is hard to find if the object is made up of a number of different materials each of which has a different resonant frequency; then any one of the tones is hard to distinguish. The latter accounts for a standard test of a piece of good crystal glassware; if it gives a clear, sweet tone when struck it is made of high-quality glass, uniform throughout the piece. If it does not, the glass is inferior, because its structure at various places in the piece has varied in manufacture. The variations in the glass compound all have different resonances and the sound is dull because it is a mixture of unrelated tones.

Just as in electrical resonators, mechanical ones have a Q factor – a factor of efficiency. In a lumped electrical tank circuit the amount of pure resistance determines the Q. The more resistance the lower the Q, for then there are power losses which cut down the sharpness of the resonance. Sound power losses come from absorbent surfaces. A tone can be produced by blowing into a cardboard milk container, but it will be lower in volume than that obtained from a glass milk bottle. The cardboard sides are not efficient as sound bouncers, for their roughness absorbs some of the sound and the resonance curve flattens out.

Acoustic resonance and Q account for the existence of the bathroom baritone. When you sing in the bathroom two things happen. First, the room itself has a low resonant frequency, which boosts the lower tones of your voice and tends to suppress the higher harmonics which cause unpleasant sound. Second, the bathroom has a high Q; its walls and floor are usually of very hard, smooth tile, so the sound is bounced right back at you instead of being absorbed.

The flute (and many organ pipes – which are nothing but king-size flutes) produce tones because of acoustic resonance. Air is blown across a sharp surface so that the stream is set into agitation. The inside of the flute, having a certain volume and shape, is resonant at a certain frequency. The pitch is varied by opening or stopping up holes which increase or decrease the length of the flute, and thereby its resonant frequency. Reed instruments such as the clarinet, oboe, saxophone, and so on, operate in the same way except that the air stream is agitated by the motion of the reed. Brasses are somewhat similar, with the air vibration produced by the player's lips when he gives the musical equivalent of a "Bronx cheer."

WHY TONE QUALITIES VARY

The only instrument which produces a fairly pure tone – one without a very high harmonic content – is the flute. This is because the agitation of the air stream produced by blowing across the sharp edge at the mouthpiece is fairly constant in character. By the resonance of the flute's air column the stream is caused to vary from minimum to maximum but it never shuts off entirely. The tone consists mostly of the fundamental pitch, though there are some harmonics because the air stream does not vary in a real sine manner. This is helped by the fact that the bore of the flute is almost entirely cylindrical, with the same diameter at all points, so that each section of the air column has the same resonant frequency. The division into sections is by no means arbitrary, since resonance is produced not only by an air column of the wave length of the fundamental, but also by columns with lengths of a quarter wave and even smaller divisions. In this respect, an air column is very like an electrical resonant transmission line.

The reed instruments are entirely different in character from the flute. The reed of a saxophone or clarinet as affixed to the mouthpiece leaves a slight opening. When blown, the reed is set into vibration against the lay of the mouthpiece. As the opening increases and reduces, the waveform of the air stream that it produces is almost a true sawtooth – gradual rise and sudden drop. Since a sawtooth wave contains an infinite number of harmonics, with the volume of each proportionate to its ordinal number, obviously the harmonic content of a reed instrument is high.

The brasses have the same type of effect, since the lip movements create the same sawtooth shape.

The sawtooth mode of the air stream in each case is produced by pressure variation. At first, with the lips or reed toward closure, a certain amount of air produces great pressure, since it is restrained by the closure. The pressure makes the lips or reed open. As the opening becomes wider, the air pressure becomes less and the opening action it gives becomes less. When the opening is wide enough, there is not sufficient air pressure to keep it open, so the reed – or with the trumpet player the lips – spring toward closure, continuing in rapid vibration.

The shape and size of the instrument and the materials of which it is made determine just which of the harmonics originally generated are heard and in what proportion. Thus the basic simple conception of tone quality variation by harmonic content variation accounts for most of the tone difference among the instruments.

Helmholtz's resonators were used many years ago to analyze the tones. The instrument to be tested was played into several resonators in turn; by listening the experimeter could get an idea of which resonators showed resonance and from that he could build up a picture of tone content.

The next step was to discover from a practical standpoint what was responsible for the total tone quality of a given instrument.

Let us take the case of the oboe. The air stream is varied in practically a sawtooth form by the reed. The length of the air column determines the fundamental resonance and the fundamental note produced. The fact that the bore is conical rather than cylindrical means that different sections of it have different resonances. (A typical oboe has an inside diameter ranging from 0.186 in. at the reed end of the instrument proper to 0.640 in. at the beginning of the bell flare. At the end of the bell the i.d. is 1½ in.) Therefore for the harmonics of the fundamental the column is divided into several parts and certain harmonics find resonances and are emphasized. The material of which the oboe is made gives a certain Q—we might call it a sound-bounce factor – which determines how much each resonant harmonic is emphasized. This is true because the amount of bounce or absorption is different at different frequencies.

But in addition to all this, the wood body of the instrument has its own resonant frequency because it has a certain mass and makeup.

The resonance of the body emphasizes the harmonics which fall at and around it.

FORMANTS

The action in an instrument which determines its tone color or quality which we have just described is covered by the modern theory of *formants,* a theory which plays a paramount part in the design of most completely electronic organs such as the Baldwin, Minshall, Lowrey, and the like.

A formant may be defined simply as a frequency range in which the harmonic components of a complex wave are prominent relative to the harmonics in other frequency ranges. Any tone color may have more than one formant, due to the physical properties of the instrument which produces it, as our discussion has indicated. The tone color as it appears to the ear is influenced not only by the formant frequency range but also by the amount of emphasis in that range and the width of the frequency band involved. Figure 4-1 is a chart showing the formant frequencies of some typical instruments. The instruments in the chart all have quite pronounced formants; many others have much wider formant ranges with much less emphasis

of the harmonics which fall in those ranges. The generalized difference to the ear is that the latter sound rather bland, while the instruments in the chart sound fairly sharp and are easily identified.

Notice the formant frequencies of the brass instruments; the frequency becomes higher as the size of the horn bell decreases. Note, too, the human voice as shown in the chart. The voice has two formants caused by the size and resonances of the oral cavity (not by the difference between male and female voice pitches, which is accounted for by the difference in fundamental pitch generated as the result of differences in the length and mass of the vocal chords). Speech is intelligible because in speaking we change the shape of the oral cavity to give different formants within the ranges shown. All vowels have a formant in the lower range and most also have one or more in the upper range.

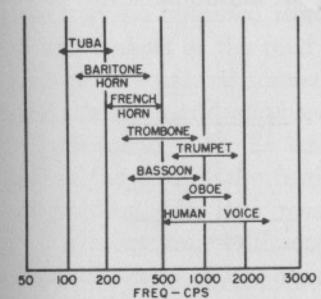


Fig. 4-1. Formant ranges of typical instruments and the human voice.

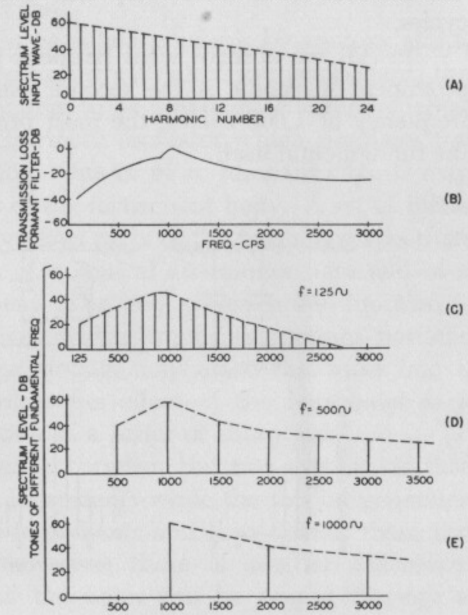


Fig. 4-2. How formant affects the spectrum of an instrument.

Figure 4-2 shows the effect of a formant range on the spectrum of a musical instrument. Let us assume that we wish to imitate a type of tone color whose formant is at and around 1,000 cycles. At (A) appears the harmonic content of any tone produced initially at, say, the reed or the mouthpiece. It may closely resemble a sawtooth, which means that succeeding harmonics will be present with amplitudes inversely proportional to their orders. To duplicate it, therefore, let us set up a sawtooth oscillator of variable frequency.

In (B) we see the desired bandpass characteristic of the following circuits, with a rise in response at 1,000 cycles and a drop in response at each side of 1,000. This can be achieved with a simple *L-C* tuned circuit of moderate Q. In (C) we see what happens when we pass a tone of 125 cycles through the system. The most prominent harmonic, due to the response of the filter, is the eighth, at 1,000 cycles.

In (D) we observe what happens to a 500-cycle note – the most prominent harmonic is the second. And in (E), with a fundamental frequency of 1,000 cycles, the most prominent harmonic is the first – the fundamental itself.

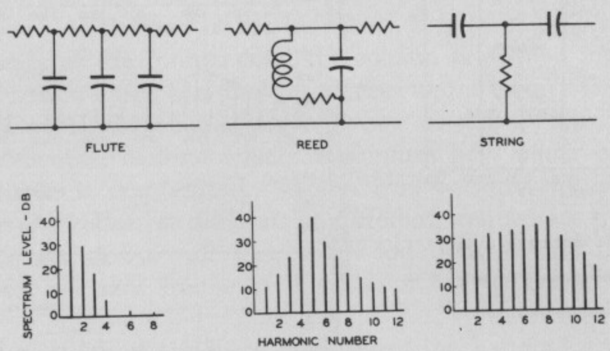


Fig. 4-3. Spectra for typical flutes, reeds, and strings, with basic electrical networks which can produce the same spectra from harmonic-rich electrical oscillations.

This is a simplified example, for actually the "response curve" of the spectrum of an organ stop or other instrument does vary somewhat, depending on the fundamental frequency – the formant shifts to some extent as the instrument is played over its fundamental range. There are, however, very much larger differences between the spectra of some other instruments, so providing a single filter over the entire range of an electronic instrument makes a satisfactory approximation. Figure 4-3 shows typical spectra for flutes, reeds, and strings in the octave above middle *C*, together with the basic electrical filters which will give the correct frequency response to imitate these spectra. In practice, the basic filters shown may be followed by others which help to produce the desired boost and rolloff characteristics. Notice the sharp dropoff in upper flute harmonics, the highly resonant character of the reed spectrum, and the much softer emphasis at the string formant. Strings in general have a number of formants.

ELECTRONIC TONE SYNTHESIS

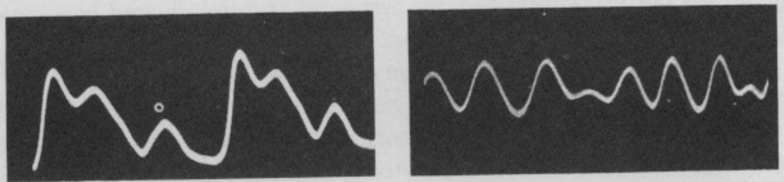
Two distinct approaches are used in electronic musical instruments for producing various tone colors. One, generally known as *harmonic synthesis,* constructs specific tone colors by mixing together sine waves corresponding to a fundamental and the desired harmonics. This is done in the Hammond organ.

The second approach might be called synthesis by anology and is based on formant theory. It analogizes with electrical components the acoustic properties of the instrument to be imitated. The fundamental tone may be obtained from a sawtooth oscillator which simulates the lip movements or the reed action. One or more resonant circuits may simulate the natural resonance of the instrument body. A set of filters may attenuate and emphasize various parts of the frequency spectrum to correspond with the effects of a conical air-column bore and of a bell at the end of the instrument. The bell, whose prime function is air-coupling, also has resonances which emphasize certain portions of the spectrum. Differentiator circuits may make the wave into a series of sharp pulses to simulate the effect of the horsehairs in a bow, which set a string in motion in a series of sharp jerks.

Fundamentally the difference between the two systems is that the first builds a tone from its ingredients while the second generates a wave containing all possible ingredients and then deletes those not wanted. In actual practice, however, there is another difference. With the second approach, all the notes can be passed through a single set of filters for a particular tone quality. Since the formant frequencies and those of emphasis and attenuation do not vary greatly no matter what fundamental pitch is being produced, the waveform of the finished tone is different from note to note. If a formant frequency for a certain quality is 800 cycles, for instance, then a 200-cycle note will contain a large component of its fourth harmonic. A 400-cycle note would have a large second-harmonic content, while an 800-cycle note of the same instrument or stop would tend to approach a pure sine wave (though usually a second, higher, formant takes over at this point). Thus the waveform of the three notes of the same stop would be different; this is the case with actual acoustic instruments and the realism obtained with the system is remarkable.

With harmonic synthesis, as it is used in the Hammond organ, the controls are so set that every note has the same harmonic content. For a given setting, for instance, there may be 50% fundamental, 25% second harmonic, and 25% third harmonic. Then every note of the scale has the same waveform. While this system produces pleasing tones, it does not simulate ordinary instruments (though many of the diapasons can be successfully approximated).

The oscillograms shown in this chapter were taken from the tones produced on a 3-manual Kilgen organ in an auditorium at Oklahoma Agricultural and Mechanical College, Stillwater, Okla. They were made especially for the purpose by Professor Hugh Lineback, to whom

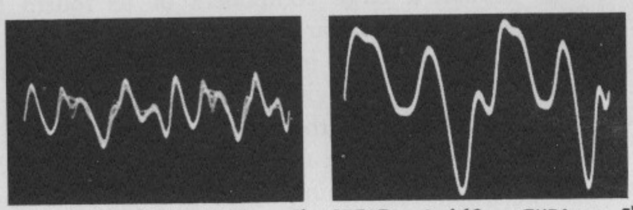


GREAT-A-110-VIOLONE CELLO 8' CHOIR-A-440-VIOLONE CELLO 8'

Fig. 4-4. Waveform of an organ cello stop.

the author expresses great appreciation. These photos show how the waveforms of particular stops are altered at different frequencies.

To make these waveform photographs Professor Lineback placed a microphone in front of the organ chamber and recorded the patterns of most of the stops at two pitches two octaves apart, the *A* just above middle *C* (440 cycles) and the *A* two octaves lower down. Comparison of each pair shows clearly how the harmonic content lessens, more nearly approaching a sine wave, as the 440-cycle tone comes nearer to its formant (which is different for each stop) and to the upper-frequency limit of the harmonics of the pipe. The legends indicate the organ department (group of pipes and corresponding manual), the frequency, the name of the stop, and the register. (The three principal registers are the 8-foot, 4-foot, and 16-foot. When the 8-foot register is used, the tone pitch corresponds to the key pressed. When the tone heard is one octave lower than the normal tone for the key used, the register in use is the 16-foot; when the tone sounds one



CHOIR-A-110- ORCH. OBOE — 8' GREAT — A-440 — TUBA — 8'

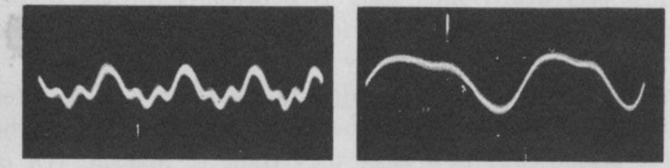
Fig. 4-5. Organ oboe pattern indicates richer harmonic content.

octave higher than the key would indicate, it is in the 4-foot register.)

The waveforms of the 8-foot violin cello stop, Fig. 4-4, are particularly interesting. Notice that at 110 cycles, the main outline of the waveform is a sawtooth. The 440-cycle violoncello pattern is

considerably simplified from a harmonic standpoint, since the formant frequency is being approached.

The oboe is a true reed. Its sound is rather "buzzy," indicating a complex harmonic structure. That structure is amply illustrated by



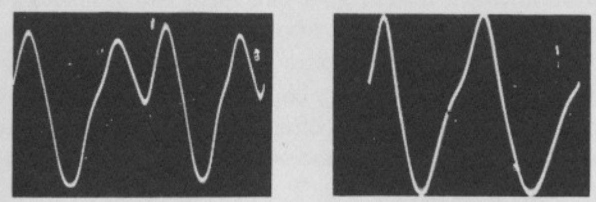
GREAT — A-110 — FLUTE 4' GREAT — A-440 — FLUTE 4'

Fig. 4-6. Flute tones are simple but not sine waves.

the oboe patterns, Fig. 4-5, left. Even without lengthy analysis, it is easy to see from the many undulations and irregularities of the waveform that it is far from a simple sine wave.

The tuba is normally a brass instrument, but in a pipe organ it is approximated by a reed. The tuba waveform, Fig. 4-5, right, indicates a complex harmonic structure and — even more important — its complexity even at 440 cycles shows that its formant frequency must be rather high.

The flute tones of an organ are produced by flue pipes rather than reeds. It is often supposed that a flute tone is almost a sine wave. The patterns shown here, Fig. 4-6, refute that to a large extent. As we said early in this book, a sine-wave tone is musically uninteresting. The flute pipes have harmonic output and formants just like the others. The flute pattern for *A*-110 (it is actually a 220-cycle tone because it is in a 4-foot rank) is obviously well endowed with harmonics, though it is simpler than the brass, reed, and string stops such as



GREAT-A-110-OPEN DIAPASON-8' GREAT-A-440-OPEN DIAPASON — 8'

Fig. 4-7. Two diapason waveforms.

the tuba, oboe, and cello. It's formant is not too high, however, for the A-440 (actually 880-cycle) pattern shown next to the other is beginning to approach sine-wave form.

The diapasons, Fig. 4-7, are heavy flute-type tones which are the foundation of what most hearers think of as true organ quality. The 110-cycle diapason pattern is relatively simple, showing a large fundamental content, with the second harmonic the predominant one. At 440 cycles, the fundamental is very strong. Though there are some higher harmonics, as evidenced by the slight waveshape irregularity, the wave is very nearly sine. Largely because of this the diapason is never used as a solo voice, its principal function being to reinforce other organ stops or to accompany congregational singing.

Readers who plan to build electronic organs will be interested in comparing the waveforms of their own instruments with these to see on the oscilloscope how close they come to realism. (Of course, stops with the same name on different organs vary somewhat.) Actual analysis of these patterns by inspection is extremely difficult, but clues can be obtained by comparing the patterns with the many harmonic combinations pictured in the *Encyclopedia on Cathode Ray Tubes and Their Uses*, by Rider and Uslan (John F. Rider Publisher, Inc.).

Now, beginning with the next chapter, we shall proceed to analyze a large number of successful commercial electronic musical instruments. This should give a better insight into all the ramifications of the subject than we could possibly give by further explanation of theory.

Chapter 5

The Hammond Organ

We begin this section on commercial instruments with the Hammond Organ, the first electronic (or, as its makers prefer, electrical) musical instrument to be commercially successful. It is still successful, outselling all others in the field. Its continuing position of leadership does not indicate that it is the best instrument in any sense; the "best" is a matter of personal preference. It is largely due to the Hammond's early start and its distinctive sound which together made "electric or electronic instrument" and "Hammond" synonymous in the public mind before it had any serious competition.

The Hammond Organ is not actually an imitative instrument, its tone being recognizable without any difficulty, although it can give a reasonable facsimile of a few of the flute-family tones of a standard organ. It is particularly useful for the weird effects for which it is so often used on the radio and its special tone colors and fast attack have made it a favorite of many popular-music performers. Its comparatively low price and long-period reliability are also notable, and many churches have bought it largely for these reasons. Many persons prefer the Hammond tone to that of other instruments which are more imitative of traditional pipe organs. The instrument is now very widely available, being sold through piano dealers, department stores, and the like. There are several models, ranging from a spinet to the concert model shown in Fig. 5-1. They all function in the same basic manner and may be understood (and serviced) with the aid of the information in this chapter.

The block diagram of Fig. 5-2 gives a preliminary over-all view of the Hammond organ. The tone generator assembly generates a sine-wave tone of every pitch required, 61 for the 5-octave range, plus an extra 30 tones for use as harmonics and subharmonics in mix-

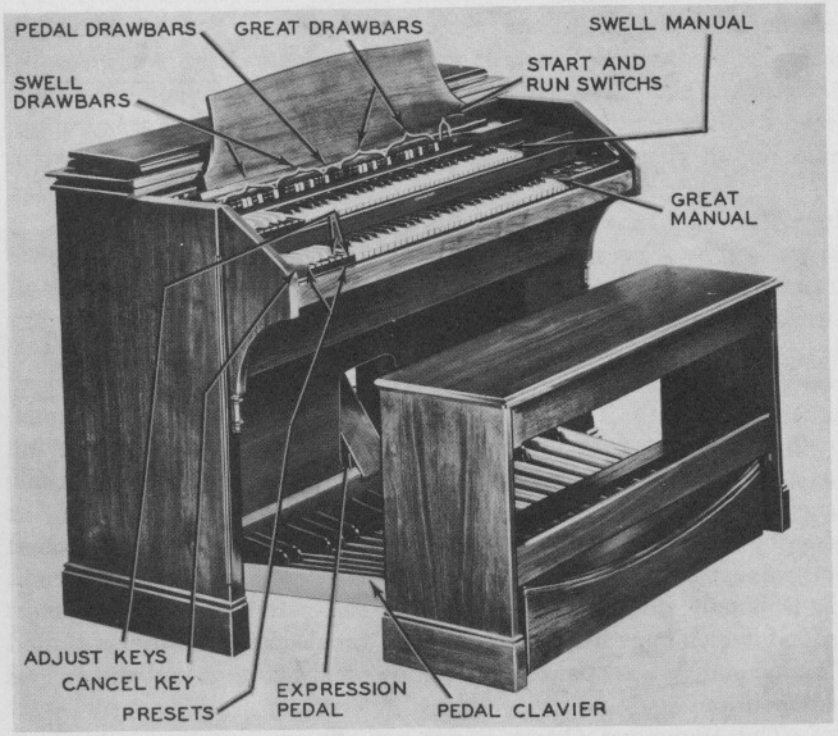


Fig. 5-1. A concert-model Hammond Organ.

ing tone colors, or 91 in all. Nine of these tones are fed to a 9-pole, single-throw switch assembly under each playing key, each pole carrying the tone corresponding to one of the harmonics or subharmonics of the tone represented by the key. The pedals operate 8-pole switch assemblies.

From the playing keys and pedals the tones go (through preset and adjust switches) to sets of drawbars. Each drawbar is an attenuator for one harmonic – the third, for instance. As the drawbar is pulled out step by step it increases the amplitude of the third harmonic of each note played on the manual controlled by that set of drawbars. There are five sets of drawbars, two for each of the two manuals and one for the pedal clavier. Which set shall govern it at a particular time on a given manual is decided by whichever adjust key is selected. To set up a particular tonal "formula" on one manual, the player pulls out the drawbar corresponding to each harmonic to an extent corresponding to the amplitude of that harmonic he desires in the tone.

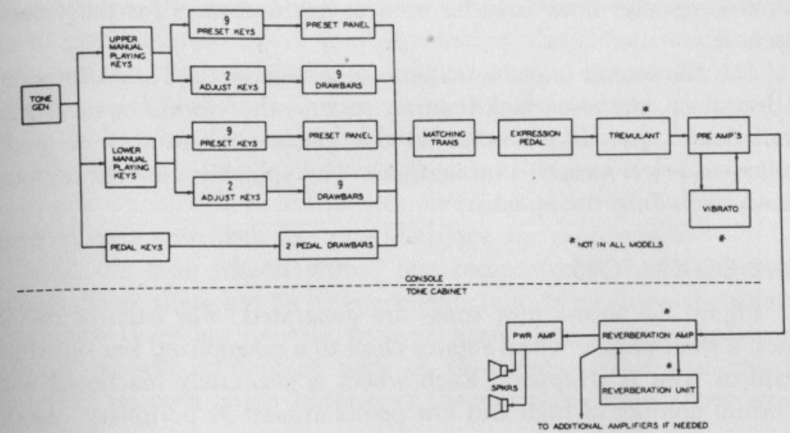


Fig. 5-2. Block diagram illustrates Hammond operation.

PRESET PANEL

A preset panel is provided for each manual. This is an octave of 12 keys at the left of the manual, with colors reversed – "black" keys white, and "white" keys black. The second through twelfth of the preset keys are push switches; push one, it remains down and causes any other key down to spring up. Push the first (the lower C key) and it cancels whatever was down, then springs up itself.

The second through tenth keys (C-sharp through A) control certain preset *combinations* of harmonics. The eleventh and twelfth (B-flat and B) are adjust keys. They switch over to manual drawbar control and select the combination which the player has set up on one of the two sets of drawbars for each manual. Quick changes in registration are easy, requiring only a quick push of one preset or adjust key for each manual. There are no presets for the pedal clavier. Two drawbars are provided, one giving a combination of the lower-frequency harmonics and the other higher-frequency tones.

Tone emerging from the harmonic mixing system is fed to a matching transformer and then to an attenuator or expression pedal. Next it passes through a tremulant, which is a motor-driven variable-resistance device to vary volume at the tremolo rate. Finally the tone reaches a preamplifier, combined with which may be a device which gives a vibrato effect – varying the pitch of the tones slightly at a rate of a few cycles per second.

The output of the preamplifier is fed to one or more loudspeaker units. Each contains at least one power amplifier and one may have a reverberation unit which gives the echo effect found in large halls. As many speaker units may be used as are necessary for the volume required.

The Hammond organ is made in several models. They all operate as described, but some lack features such as the vibrato or reverberation. Some have fewer notes in the generator assembly or pedal clavier, or fewer preset combinations. The spinet is entirely self-contained, including the speaker.

TONE GENERATORS

Figure 5-3 shows how tones are generated. For each of the 91 tones, a steel phonic wheel rotates close to a magnetized bar on which a coil of wire is wrapped. Each wheel is accurately machined with a certain number of high and low points around its periphery. As the

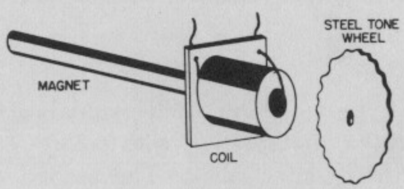


Fig. 5-3. The Hammond tone-generator, a phonic wheel.

wheel rotates, the steel body is effectively brought closer to and farther from the magnetized rod with a frequency depending on the rotational speed and the number of teeth. As this occurs, there is a small change in the magnetic field. The field changes induce a small voltage in the coil at the frequency of the changes. The voltage is a.c. of roughly sine-wave form, and is, in effect, the generated tone.

The long generator assembly contains 48 rotating assemblies, each including two phonic wheels. Since there are only 91 tones, five of the wheels are blanks, included for mechanical balance.

DRIVE ASSEMBLY

The drive assembly has a long drive shaft. At one end of the shaft is the starting motor, a shaded-pole induction unit; at the other end is the synchronous running motor, which holds the speed of the shaft in step with the power-line frequency. Couplings to the motors are resilient so small instantaneous changes in speed due to line frequency variations are absorbed.

Along the drive shaft are 24 brass gears, two each of 12 different sizes. Within the assembly, the space is divided into 24 compartments, each containing one brass gear attached to the drive shaft and four tone wheels. Figure 5-4 shows in cross-section one of the 24 tone-wheel assemblies. At the center is one of the 24 brass gears on the drive shaft. This drives two bakelite gears, one on each side and each on its own shaft. At the two ends of each of the two subsidiary shafts is a tone wheel. There are four in each compartment, and four magnetized rods with coils are provided, one for each wheel.

All the tone wheels within one compartment run at the same speed. Since there are 24 compartments (and brass drive-shaft gears), and since there are only 12 sizes of brass gears, two compartments contain wheels rotating at each of 12 speeds. Thus one speed is provided for each of the 12 tones of the chromatic scale. Tone wheels

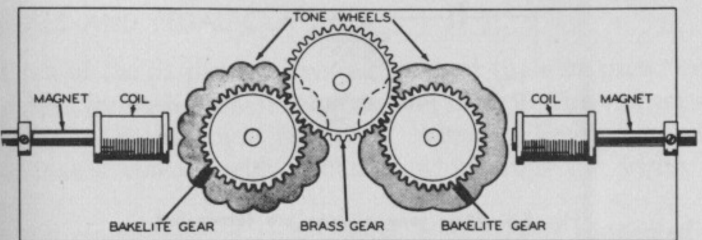


Fig. 5-4. One of the 24 Hammond tone-wheel assemblies.

with 2, 4, 8, 16, 32, 64, and 128 teeth provide the 12 tones repeated at various octave intervals to fill out the complete range of the instrument.

The tone generator assembly is a unit of the highest precision. The steel of the phonic wheels must be homogeneous and exactly the same from one wheel to another. Machining of the wheel teeth must be precise to avoid introducing spurious harmonics and to get a good sine wave. The wheels must be dynamically balanced, and there must be no play in the gears. A synchronous motor actually runs in a series of pulsations and the pulsations must be ironed out. Resilient coupling from motor to drive shaft and similar couplings dividing the shaft itself are necessary.

Lubrication is important. The individual tone-wheel shafts are mounted in bearings of porous bronze. Each bearing is connected to an oiling system by a cotton thread. Capillary action carries oil from a reservoir trough. Proper maintenance includes using the grade of oil recommended, for mixing grades will gum up the cotton threads and prevent oil flow.

SINE WAVE OUTPUT

The output of each tone generator must be as nearly sine-wave as possible. The tone coloring system requires this. The lower-frequency coils have copper rings on them. The eddy-current loss in the ring is low for the fundamental frequency but high for the harmonics. All the coils have simple filters at their outputs. As shown in Fig. 5-5, the first 43 coils are shunted with a resistor. The next five are fed through transformers. The upper coils are fed through the

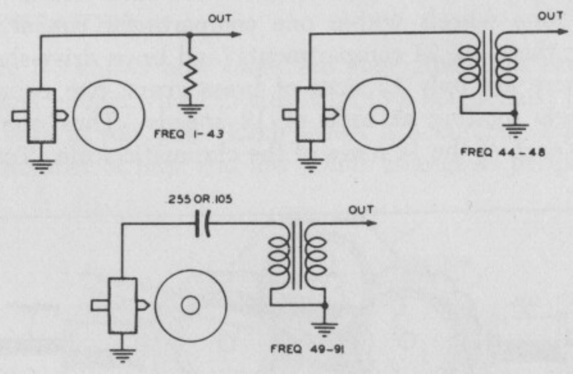


Fig. 5-5. How tone pickups are terminated.

transformers with a series capacitor; the capacitor and transformer make up a tuned circuit at the fundamental frequency. Each transformer is different, with the correct turns ratio and inductance for the frequency.

The entire generator assembly is built as one unit, and a new one can be substituted in case of damage or change to a location with a new line frequency. When the service technician orders a new assembly, however, he must give the model and serial number of the organ, as different generator types are not interchangeable.

Some models of the Hammond organ have a chorus generator in addition to the main generator. The chorus generator is similar to the main generator, and provides frequencies 56 through 91. Frequencies 56 to 67, however, are 0.8% sharp or flat, while the rest are 0.4% from the correct frequency. This slight difference between the frequencies of the two generators gives a chorus or ensemble effect of two instruments playing at the same time. It destroys the undesirable precise tuning of the electrical system.

The precise, perfect tuning of the electrical generation of tones is most unlike, some say, the quality of music produced by ordinary acoustic instruments. These have an imprecision, or variation in pitch

and tone quality, which gives the music warmth, life, and vibrancy. One of the major drawbacks, then, of making music electrically, is that this very quality of spontaneous, live, direct control of the tone, pitch, and quality of the music is not possible in an electric system.

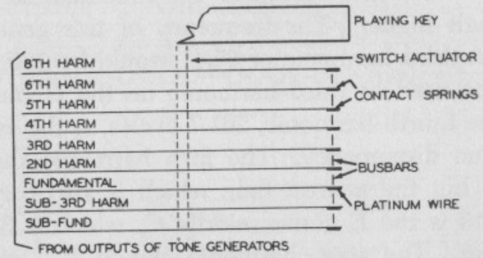


Fig. 5-6. Hammond key contacts.

MANUALS AND PEDAL CLAVIER

Each of the 61 playing keys and each of the nine preset and two adjust keys for each manual actuates nine small bronze contact springs with precious metal points (Fig. 5-6). When the key is pressed, each spring makes contact with a busbar which runs the width of the manual.

Each contact spring (note shape, Fig. 5-7), is connected to the output of one tone generator. Resistance wire is used, the resistance affording isolation necessary because each tone generator is connected

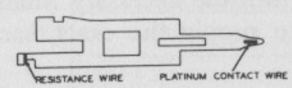


Fig. 5-7. Note shape of contact springs.

to a number of contact springs. This arrangement is the key to the harmonic synthesis tone color system employed in the Hammond electric organ.

The tones provided by the generator are the fundamentals of all the 61 notes plus another 12 below the lowest keyboard note and another 18 above the topmost one. Each of these tones is one of the 12 in the "well-tempered" scale with which we are familiar. In addition to being used as fundamental tones, however, they are also used as harmonics. For instance, suppose the lowest C on the manual is pressed and the fundamental drawbar is pulled out. The note heard will be 65.41 cycles (refer to the frequency chart of Fig. 2-1 on page 7). Now, with the same key pressed and the second-harmonic drawbar pulled, the note heard will be 2×65.41, or 130.8 cycles, derived from the same tone generator which would furnish 130.8 as the fundamental for the C one octave higher.

The third harmonic for the lowest C key is derived from the same generator which also furnishes the fundamental for the G an octave and a half higher. The frequency of this generator is 196.0 cycles. The true third harmonic of 65.41 would be 196.2 cycles, however, so the tone used as third harmonic on the Hammond is really slightly flat. The fourth harmonic, 261.7 cycles, is the same as middle C, so there is no discrepancy. The fifth harmonic should be 327.1 (approximately) but the nearest tone which is also part of the scale as a fundamental is the E above middle C, which is 329.6, so again there is an error. The sixth harmonic should be 392.5, while the nearest available note (G above middle C) is 392.0, another error. The seventh harmonic should be 457.9, but the nearest are 440 and 493.9, neither of which is close enough to do. The seventh harmonic therefore is not used on the Hammond. The eighth harmonic is the following C at 523.3 and is perfectly accurate.

The so-called sub-third harmonic is an octave below the normal third harmonic and has the same error. The subfundamental is simply one octave below the fundamental.

The errors in harmonics are actually slight, the worst being the fifth harmonic. Not many listeners can tell that the harmonic structure is faulty. The effect, if audible, is merely characteristic of the instrument and is neither a fault nor a virtue. It would be economically unfeasible to furnish the necessary fundamental tones plus the additional generators to supply the exact harmonics for each note.

MIXING HARMONICS

The method by which drawbars and presets control harmonic mixture is clever and simple. Figure 5-8 shows the entire tone coloring system. Note the set of busbars marked LOWER MANUAL, for instance. There are nine bars, one for each harmonic and subharmonic. These bars run the width of the console, under the playing keys, the preset keys, and the two adjust keys.

Each of the 61 playing keys of that manual actuates a set of nine vertical contact springs, as in Fig. 5-6. Each spring is wired to an appropriate tone generator and carries that tone continuously. A typical playing key, with its resistance-wire connections to the tone generators, is represented by the vertical set of contacts at the left of the lower-manual busbars in Fig. 5-8. All 61 keys are the same. Any number pressed at the same time make contact with the same set of busbars.

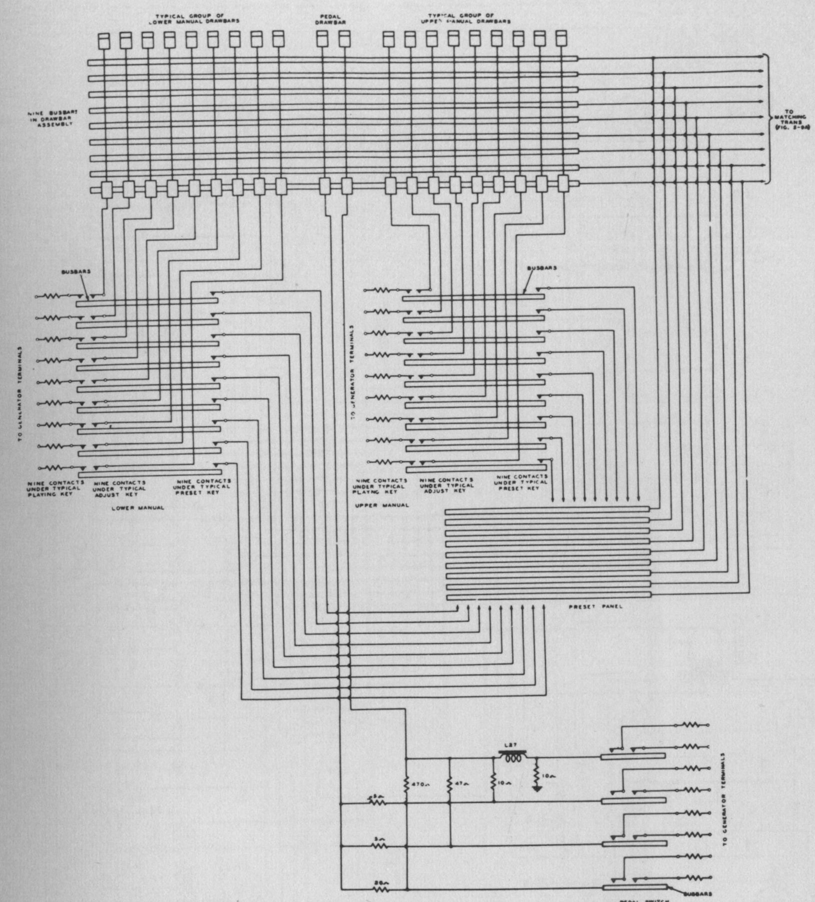
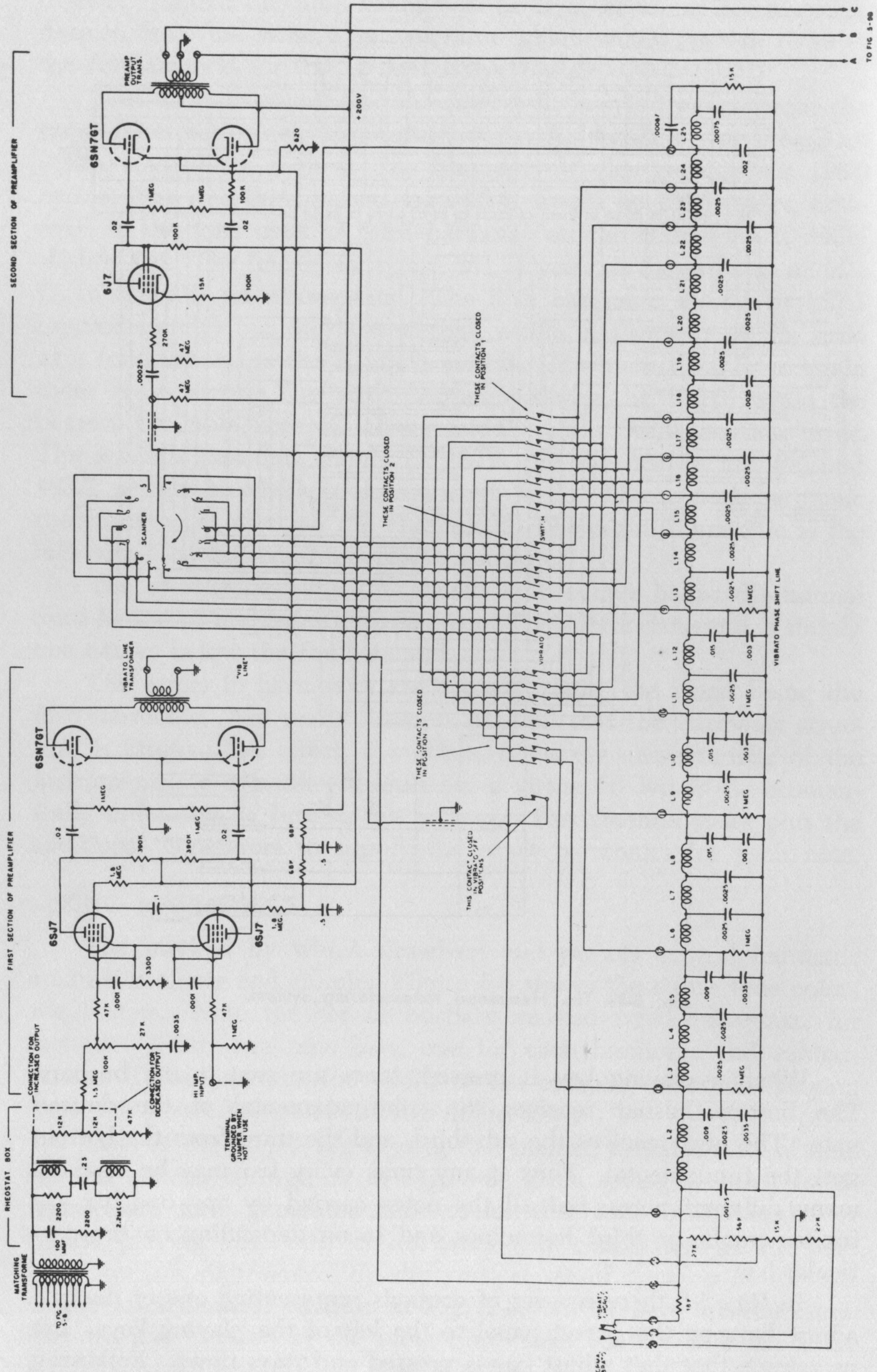


Fig. 5-8. The Hammond tone-coloring system.

When a playing key is pressed, tones are sent to the busbars. The bottom busbar receives the sub-fundamental of the desired note. The next receives the sub-third, and the third from the bottom gets the fundamental. Thus at any time, every bar may be carrying many different notes; but all the notes carried by any one bar are fundamentals, or third harmonics and so on depending on the particular bar.

In Fig. 5-8 there is a set of contacts representing one of the two adjust keys on the preset panel to the left of the playing keys. Let us assume that that adjust key is pressed and stays down. Following



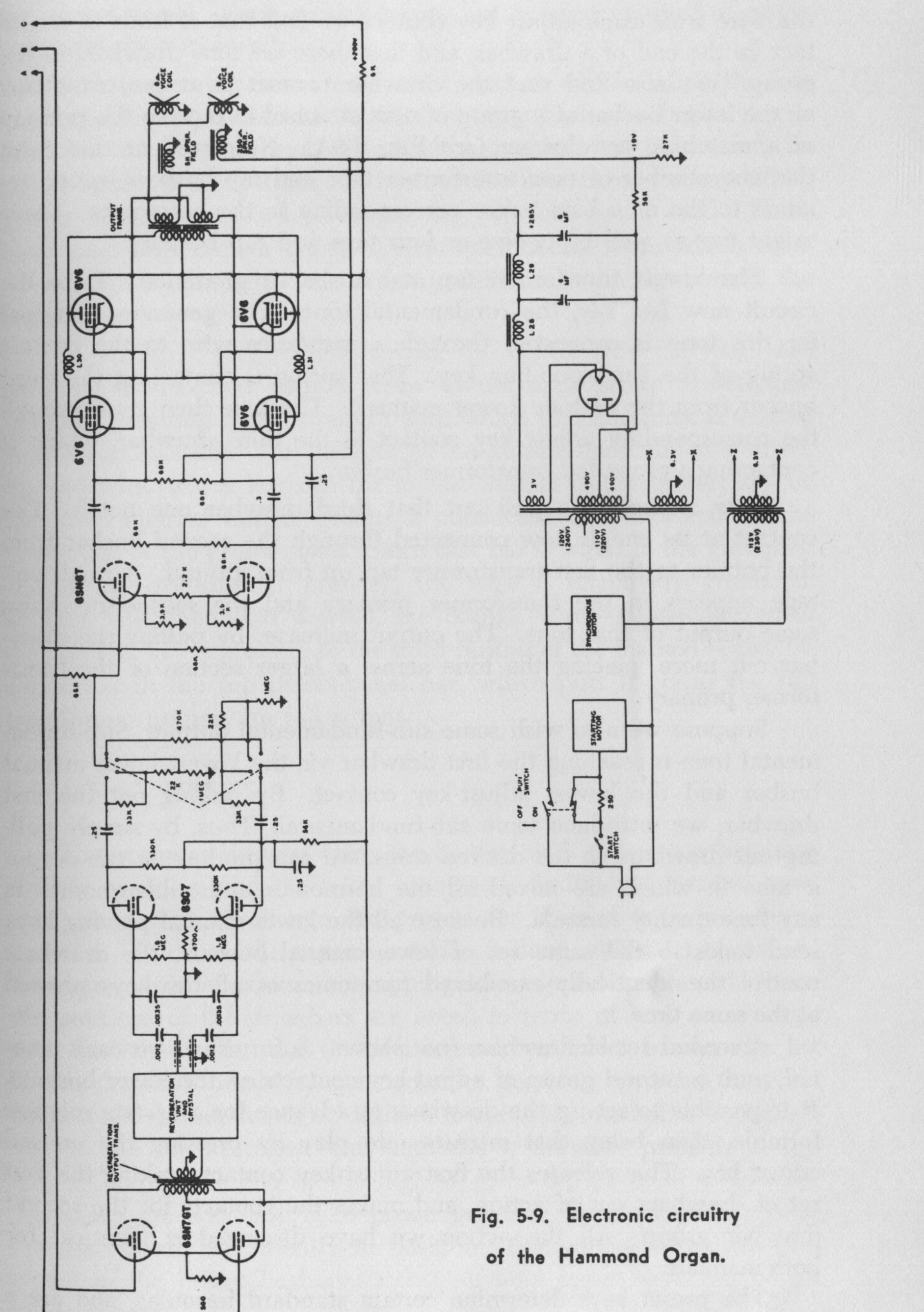


Fig. 5-9. Electronic circuitry of the Hammond Organ.

the wire from each adjust key contact we find that it leads to a contact on the end of a drawbar, and that there are nine drawbars in the group. We also find that the drawbar contact is at present sitting on the lower busbar of a group of nine attached to taps on the primary of a matching transformer (see Fig. 5-9A). Note well at this point that the number of nine transformer taps and tap busbars has no relation to the nine bars below corresponding to the harmonics. There might just as well be twelve or four taps and tap busbars.

The lowest transformer tap and busbar is grounded. Trace the circuit now for, say, the fundamental tone. The generator terminal for this tone is connected through a resistance wire to the contact spring of the corresponding key. That spring is contacting the third busbar from the bottom (lower manual). The tone then goes through the corresponding adjust key contact to the third drawbar, which is contacting a grounded transformer busbar.

Now suppose we pull out that third drawbar one notch. The contact at its end is now connected through the second busbar from the bottom to the first transformer tap up from ground. A small voltage appears in the transformer primary and the secondary shows some output of that tone. The output increases by pulling the drawbar out more, placing the tone across a larger section of the transformer primary.

Suppose we also wish some sub-fundamental output. Sub-fundamental tone is reaching the first drawbar via the lowest lower-manual busbar and the lowest adjust-key contact. By pulling out the first drawbar, we introduce some sub-fundamental. Thus, by simply pulling out drawbars to the desired steps, we can produce in the output a tone in which are mixed all the harmonics and subharmonics in any tone-quality formula. Because all the lower-manual playing keys send tones to the same set of lower-manual busbars, the drawbars control the identically numbered harmonics of all the keys pressed at the same time.

A second set of drawbars (not shown) is furnished for each manual, with a second group of adjust-key contacts on the lower busbars. It is possible to set up the drawbars in advance for a certain mixture formula, then bring that mixture into play by pressing the unused adjust key. This releases the first adjust-key contacts, taking the first set of drawbars out of action, and makes the contacts for the second drawbar group. All the action we have discussed is identical for both manuals.

The preset keys determine certain standard formulas, and are a part of the same system. As Fig. 5-8 shows, one lead from each transformer tap is brought down to an auxiliary set of busbars known as

the preset panel. Each of these bars has a large number of screw terminals along its length.

Each preset key has nine contact springs which, when the key is pressed, make contact with the same group of nine busbars as is used for the playing keys. Thus, when a preset key is pressed, its nine contacts pick up from the busbars the nine tones corresponding to the harmonics of whatever playing keys are being used. These nine tones appear in a group of nine color-coded wires which come up through a hole below the preset-panel busbars on the back of the organ case.

To set up a preset combination, the coded wire carrying each harmonic is attached with a screw to the appropriate preset-panel bar corresponding to the strength with which that harmonic is desired in the combination. For instance, if no sub-fundamental is desired, the red wire (which leads to the lowest contact of that preset key on the lower-manual busbar group) is connected to a terminal on the lowest bar of the preset-panel. Since that bar is wired to the grounded end of the transformer primary, the wire is effectively grounded. If loud fundamental is desired, the orange wire – connected, when that present key is pressed, to the third busbar of the manual – may be connected to the top preset-panel bar, which puts it across the full transformer primary for full output.

CHANGING PRESETS

The organ is shipped from the factory with preset combinations already made up and the wires connected. Individual owners, however, often wish to change the presets for their own favorite tonal varieties. To do so, the experimentation is done with a set of manual drawbars, with, of course, the appropriate adjust key pressed. Then the positions of the drawbars are noted in terms of how many steps each has been pulled out. With a screwdriver in his hand and a list of the drawbar numbers, the owner or service technician may reconnect the preset wires as desired, keeping in mind that a connection to the lowest preset bar simulates a drawbar pushed all the way in and connection to the highest gives the effect of a drawbar pulled all the way out. The preset panel is, in effect, nine sets of drawbars fixed permanently in position. Enough terminals are provided on the preset busbars so that all nine preset keys can be set up as desired.

Table 5-1 gives the color coding of the preset key wires.

TABLE 5-1

Harmonic	Wire color
Sub-fundamental	Brown
Sub-third harmonic	Red
Fundamental	Orange
Second	Yellow
Third	Green
Fourth	Blue
Fifth	Violet
Sixth	Grey
Eighth	White

PEDAL DRAWBARS

The pedal drawbars work on the same principle as the manual drawbars but the wiring is different and there are no presets. In some models fundamental through eighth harmonic tones are used; in others the tenth and twelfth are added. In each case, however, two harmonics are fed to each pedal busbar, as indicated in Fig. 5-8. The outputs of all the busbars are mixed with a resistor system and two outputs are derived, one predominant in low and one in high harmonics. These are fed to two drawbars which control pedal tone quality in a much less exact and selective manner than for the manuals. This is justified because the pedal is ordinarily used only as a "foundation" tone to add a solid bass to the ensemble, and it is rare to play two pedal notes simultaneously.

ELECTRONIC SECTION

The only electronic sections of the Hammond organ are those following the tone-generator and harmonic-mixing circuits. Because the instruments do not contain vacuum tubes (except for audio frequency amplification), Hammond prefers to call it an *electric* rather than an *electronic* musical instrument.

Figure 5-9 is a schematic of the entire Hammond electronic circuit. Parts of the circuits differ somewhat from model to model but this diagram gives a good over-all picture.

The secondary of the matching transformer (Fig. 5-9A) is connected to a rheostat box. It contains a rheostat which is operated by

the foot pedal to control expression or volume; the shaft is connected to the pedal by a rod, which can be seen in Fig. 5-10. The rheostat is across the transformer secondary, in series with a capacitor and coil. This makes it impossible to lower the volume to zero. The rheostat box ends with a three-resistor voltage divider which has two taps

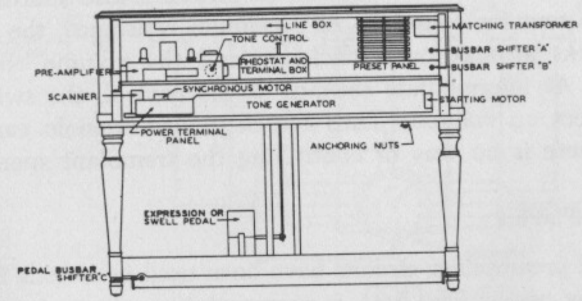


Fig. 5-10. Rear of Hammond, showing layout of parts.

brought out to terminals. When the organ is installed, the proper tap to give the desired amount of output signal is selected.

Figure 5-11 shows one of the rheostat boxes used in models which have a tremulant. The tremulant varies the *volume* at a slow rate, approximately 6.33 cycles. It is in effect a rheostat in series with the high side of the circuit and it is mechanically varied from minimum to maximum resistance and back at the tremolo rate. It is actually a

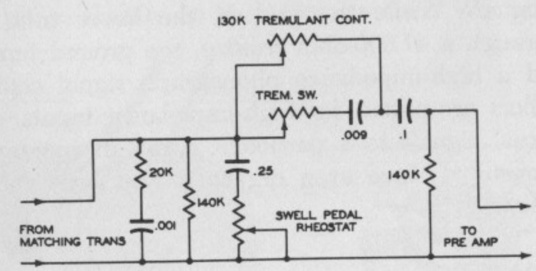


Fig. 5-11. Schematic of a rheostat box.

series of five resistors ranging in value from 15,000 to 450,000 ohms with their junctions connected to contacts. A laminated bakelite strip is positioned so it can slide between the contacts, one after another, and break them. The strip is alternately pushed in and out of the contact assembly by an eccentric geared to the shaft of the tone-generator motor. The gearing ratios are such that the action takes place at slightly over 6 cycles per second. This varies the net resistance of the tremulant switch and the output volume from the rheostat box.

The 130,000-ohm tremulant control is directly in parallel with the varying tremulant switch. When the control rheostat is shorted (by adjusting the knob on the console) the switch is also shorted out and the volume remains constant. At maximum resistance, the tremulant switch works at maximum efficiency and the volume changes are maximum. At intermediate settings of the control, the switch has a smaller effect on volume. Thus the degree of tremolo can be controlled. There is no way of controlling the tremulant speed.

THE PREAMPLIFIER

Several preamplifier circuits have been used in various Hammond models. The one in Fig. 5-9A is representative. At its input there is a tone-control circuit, actually a mild bass booster, which is adjusted at installation. It compensates for room acoustics.

The input stage of the preamplifier is a cathode-coupled phase inverter. The organ signal goes to the grid of the upper tube. The cathodes of both tubes are grounded through a common 3,300-ohm resistor. The signal on the upper tube causes a voltage drop across the cathode resistor, as in a cathode follower. The same cathode voltage then appears between the cathode of the lower tube and its grid, which is effectively grounded through a 47,000-ohm resistor. Thus both tubes are excited equally and 180 deg. out of phase by the one input signal. While the grid of the lower tube is normally grounded through a 47,000-ohm resistor, the ground jumper may be removed and a high-impedance phonograph signal connected. Not all preamplifiers are suited for high-impedance inputs. Before connecting external signals to a particular organ its instruction booklet should be consulted since even organs of the same model number may differ slightly.

VIBRATO SCANNER

In models without vibrato, the output of the 6SN7 which immediately follows the 6SJ7's is transformed to low impedance and a line is run from the transformer secondary to the power amplifier in the tone cabinet. Figure 5-9A, however, shows the circuit of later models, many of which have a unique vibrato circuit. Since there was no good way of varying the frequencies of the tone generators at the vibrato rate, the problem was solved by an interesting phase-shift system.

Figure 5-12 is a block diagram of the vibrato circuits. A signal from the 6SN7 preamplifier is fed to the input of a delay line. A frequency-discriminating circuit shifts the phase of the signals going through it. A delay line is a compound low-pass filter made up of L-C components. The filter shifts the phase of signals passing through

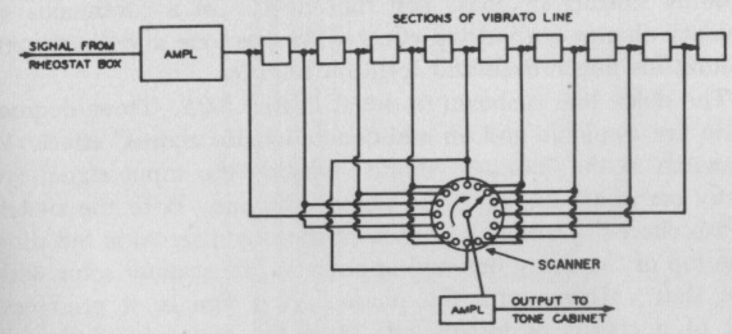


Fig. 5-12. Vibrato-circuit block diagram.

it by an amount depending on the frequency of the signal and the upper cutoff frequency of the filter. Phase shift is 180 deg. at the cutoff frequency.

A continuing shift in phase is effectively the same as a shift in frequency. While the phase is changing, frequency effectively changes as well, the amount of apparent frequency change depending on the *speed* of the phase change and the *total* phase shift.

Returning to Fig. 5-12, the signal passing through the delay line is shifted in phase slightly by each filter section. We have, then, the

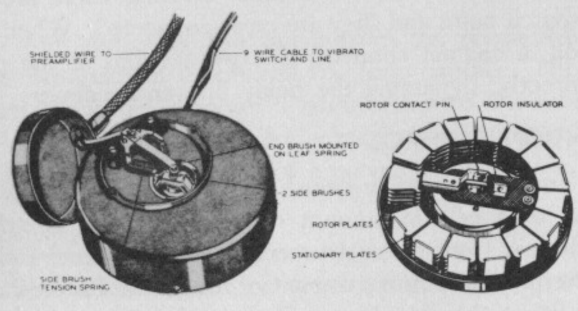


Fig. 5-13. The vibrato scanner.

same signal at the junction between each pair of sections, but at increasing phase differences. A rotary scanner, Fig. 5-13, has a set of capacitor plates rotated at the end of an arm by a shaft. Around the housing perimeter are stator assemblies, each being a set of capacitor plates through which the rotor plates pass. Each stator is connected to one point along the delay line.

As the rotor goes around, it picks up signal capacitively from each stator in turn. The rotor is connected to the input of the second section of the preamplifier. The 6J7, therefore, is fed the organ signals constantly shifting in phase and the effect is of a continuous small frequency change – a pulsing effect – like the tone a violinist gets by vibrating his finger back and forth on a string.

The delay line is shown in detail in Fig. 5-9A. Three degrees of vibrato are available and an additional "vibrato chorus" effect. With the switch in the "normal vibrato" position the input signal is fed directly across the input to the phase-shift line. With the switch in "vibrato chorus" position, a portion of the organ signal is fed directly to the top of the delay line and appears on the scanner rotor without phase shift. Mixing with the phase-shifted signals, it produces an effect of a chorus of instruments since the same signal then goes through simultaneously with two different and constantly changing phase relationships.

The three degrees of vibrato are produced by scanning less than the entire delay line. They are selected by closing one of three 8-pole, single-throw switch assemblies, as shown in Fig. 5-9A. For wide vibrato (position 3), eight points of the filter are scanned between points 14 and 2. In position 2, the scanner picks up signals only between points 11 and 1; in position 1, only points between 8 and 1 are scanned. As can be seen by the numbering of the scanner stators in Fig. 5-9A, although the rotor rotates continuously the scanning is carried on in a back-and-forth manner because there are two rotors for every switch point and they are cross-connected. When the vibrato switch is off, a contact connects the two sections of the preamplifier together directly, bypassing the delay line and scanner.

REVERBERATION CONTROL

"Live" music is usually heard in large halls, so most people are accustomed to a certain amount of reverberation caused by sound reaching the ear from the instrument and from echoes reflected from walls and ceiling. A limited amount of reverberation produces a more interesting effect than single-source music for the same reason that a chorus of instruments playing in unison is more interesting than a single instrument.

Hammond organs are often used in homes and small halls where the room is either acoustically "dead" (sound is absorbed by rugs and draperies) or too small to allow long-duration echoes. Hammond therefore created an artificial reverberation control unit, Fig. 5-14.

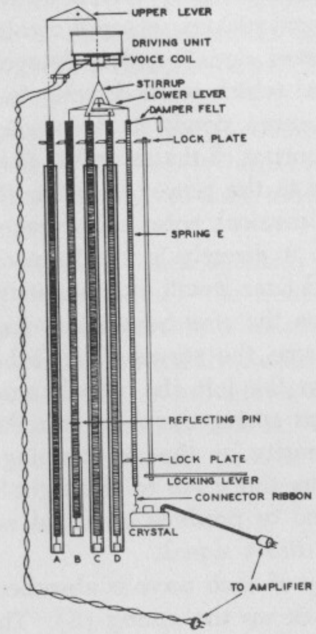


Fig. 5-14. The Hammond reverberator.

The reverberation control is an electro-mechanical device which introduces multiple echoes by reflections within a network of coil springs. The unit is about 4 x 5 inches in cross-section and about 4 feet high. It is concealed within a tone cabinet such as the JR-20 shown in Fig. 5-15.

As Fig. 5-9B indicates, the signal from the preamplifier output is fed to the two grids of a 6SN7 in the tone cabinet. The plates of the 6SN7 are connected through a transformer to the reverberation unit driver. The same preamplifier output signal is also fed to the input grids of the 6SN7 preceding the power amplifier.

REVERBERATION ACTION

The reverberation driver is a moving coil assembly similar to a dynamic speaker without a cone. The audio-frequency coil movement is transmitted to the stirrup directly under it (Fig. 5-14). The two enclosed springs, (C), (D), under the stirrup hold it in position but permit it to move freely up and down; the spring at the far left (A) balances the pull of the others. These three springs are almost entirely immersed in oil so they act largely as dampers to stabilize the response of the driver and prevent undesirable reflections.

A sound wave from the stirrup travels down the open spring (E) at the far right to a crystal pickup, where it creates an electrical signal. This is the "first reflected signal" and is delayed about 1/15 second from the original signal which went directly to the power amplifier, because sound travels more slowly in the spring than in an electric circuit or in air. The output of the crystal is fed through a 6SC7 and a three-step attenuator to the power amplifier (Fig. 5-9B). Thus, for a single, short, sharp musical note, the speaker will first emit the sound which comes to it directly in the form of an electrical wave, then about 1/15 second later it will again emit the same sound which came to it, this time via the reverberation spring.

The same wave from the stirrup (Fig. 5-14) also travels down the second spring from the left (B), which enters a short oil tube. From the bottom of this spring the wave is reflected back along the spring, reduced in intensity by the oil damping. At the stirrup the horizontal level transfers the wave to the right-hand spring (E) and it goes on to the crystal to produce a second reflected signal about 3/15 second after the direct signal.

Little of the energy of each wave is absorbed by the crystal, and the rest is reflected back up the spring (E). There it is transferred by the lever to the spring (B) in the short oil tube. It goes down that spring, is reflected up, and again goes down the crystal spring. The process continues over and over, giving a series of signals about 2/15 second apart until finally all the vibration is dissipated by the oil friction in the short tube. Just above the short oil tube a reflecting pin attached to the spring causes partial reflection and helps make the over-all response uniform.

It is interesting to note that the amount of oil in the short tube varies the amount of energy loss at each reflection, and therefore changes the total length of time during which the reflections keep going. Adjusting the level of oil is a simple way to simulate rooms of different sizes and reverberation characteristics. A reverberation selector switch operates a pair of balanced potentiometers to select the amount of reverberation signal to be added to the direct signal. The switch is in the tone cabinet and must be adjusted (usually at the time of installation).

POWER AMPLIFIERS AND SPEAKERS

The tone cabinets contain the loudspeakers and power amplifiers, as well as reverberation units in some models. Several tone cabinet models are available, differing in power output, type of speakers, size, shape, and furniture style. Two power outputs are used, 20 and 40 watts. The power amplifiers are similar and conventional; most

employ push-pull parallel 6V6's as output tubes. The 40-watt units usually contain a pair of 20-watt amplifiers, sometimes on the same chassis. The type JR-20 tone cabinet shown in Fig. 5-15 is a 20-watt unit with a reverberator.

Another tone cabinet, the DR-20, contains a 20-watt amplifier and two 12-inch speakers directed toward the ceiling. It may also contain the reverberator. In the 40-watt tone cabinet, the treble is produced by two 12-inch speakers directed upward. Bass tones are produced by a bank of nine 10-inch speakers mounted on a vertical baffle and projecting sound through the front grill.

The ideal solution is to provide a tone chamber for the loudspeakers. If the chamber is very rigidly constructed, with reverberant walls made of concrete or tile, it provides a more desirable type of reverberation than the reverberation control. The chamber should be as large as possible, at least 800 cubic feet in volume for best results.

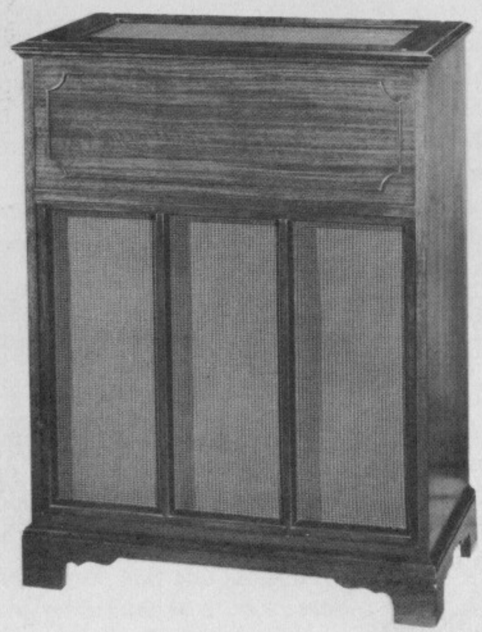


Fig. 5-15. Model JR-20 tone cabinet.

Chapter 6

The Baldwin Electronic Organ

The Baldwin organ, built by the Baldwin piano people, is the first completely electronic instrument which we shall describe. With its vacuum-tube tone generators and its tone-coloring system operating on the formant principle described in Chapter 4, its tones and resources are so like those of a pipe organ that lay hearers are usually deceived. It is used not only in churches but for recital work of a serious kind; in fact, a number of prominent organists and conductors prefer it to a run-of-the-mill pipe organ. Its console conforms in appearance, measurements, and operation to the American Guild of Organists standards and the instrument may be played without any special instruction by anyone familiar with ordinary pipe organs.

Two models are manufactured. Model 5, pictured in Fig. 6-1, is the first Baldwin organ made and is still the largest seller. It has standard 5-octave great (lower) and swell (upper) manuals and a 32-note radial pedal clavier. Vibrato (FM type) affects both manuals and pedals and is adjustable. There are 25 stops and two couplers. The console is 52½ inches wide and 29 inches deep. Model 5 is the subject of this chapter.

Model 10 is a newer organ with additional stops and couplers and with pistons to select present combinations. It operates almost exactly like Model 5. Baldwin also introduced a model which operated on the photoelectric principle, but this model has been withdrawn from the market.

GENERAL FUNCTIONING

The Baldwin organ is an all-electronic device with no moving parts except keys, pedals, and stop tablets or "tabs." It generates a series of sawtooth waves which are passed through fixed, selectable

Fig. 6-1. The Model 5 Baldwin organ.

filters to modify the waveshapes and reproduce various tone colors.

Figure 6-2 is an over-all block diagram. The generator assembly generates 73 different tones. A vibrato oscillator modifies the tone frequencies at a vibrato rate when the vibrato switch is operated. The generators operate continuously. When a key on the swell manual or pedal clavier is pressed, two gradual-contact switches connect two tones to the busses for that keyboard. On the swell, for example, if the key for A-440 is pressed, a tone of A-440 is connected to the swell 8-foot bus and a tone of A-880 is connected to the swell 4-foot bus.

Each bus is connected in parallel to the input ends of a number of tone filters. As shown in Fig. 6-2, for example, the swell 8-foot bus goes to five filters. and the swell 4-foot bus goes to three filters. At the output of each filter is a stop switch. It is normally open, but when closed it connects the output of its filter to the common bus leading to the mixer amplifier. Thus, to set up a certain combination of tone colors, the player presses the required stop tabs, connecting the outputs of the desired filters to the amplifying system. The pedal-

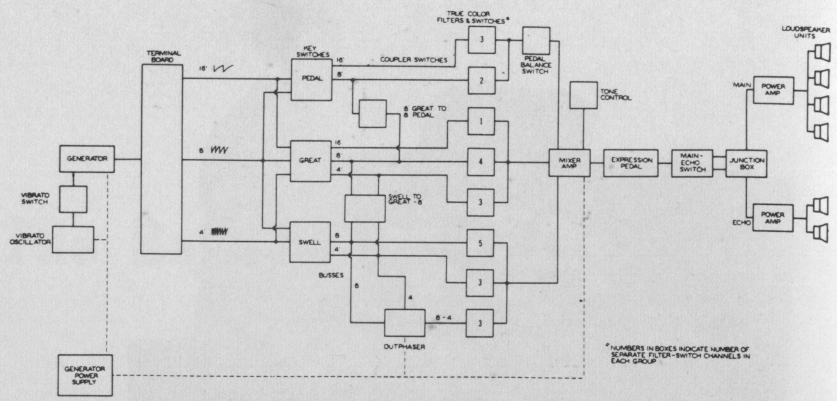


Fig. 6-2. Over-all block diagram of the Baldwin.

balance switch adjusts the outputs of the pedal-tone filters for best balance with the manual tones, depending on room acoustics, at the time of installation.

The expression pedal operates a volume control which affects the entire instrument. The main-echo switch transfers the output of the console to either or both of two amplifier-speaker assemblies located at different places in the auditorium. Tone cabinets containing power amplifiers and speakers produce the music.

TONE GENERATORS

Figure 6-3 shows the rear of the organ with the wooden back removed. The metal frame holds 13 subchassis, the first 12 of which are generators and the last the power supply. Each generator chassis generates a single alphabetical note of the scale, six notes per chassis similar in tone but an octave apart. The chassis, when in place, are connected to the system through bus bars and plugs.

Figure 6-4 is a schematic diagram of one generator chassis. The left triode of the left 6SN7-GT is a self-controlled, grid-tuned L-C oscillator tuned to one note of the octave between 1109 and 2093 cycles. Oscillation is caused by positive signal feedback from plate to grid through the transformer, which has a turns ratio of 6:1. The trimmer across the grid winding is used for tuning. The output of this oscillator, signal 1, furnishes the top-octave tone for the instrument. It is taken from the voltage divider R to furnish a correct load impedance for the oscillator and a good source impedance for the following keying circuits.

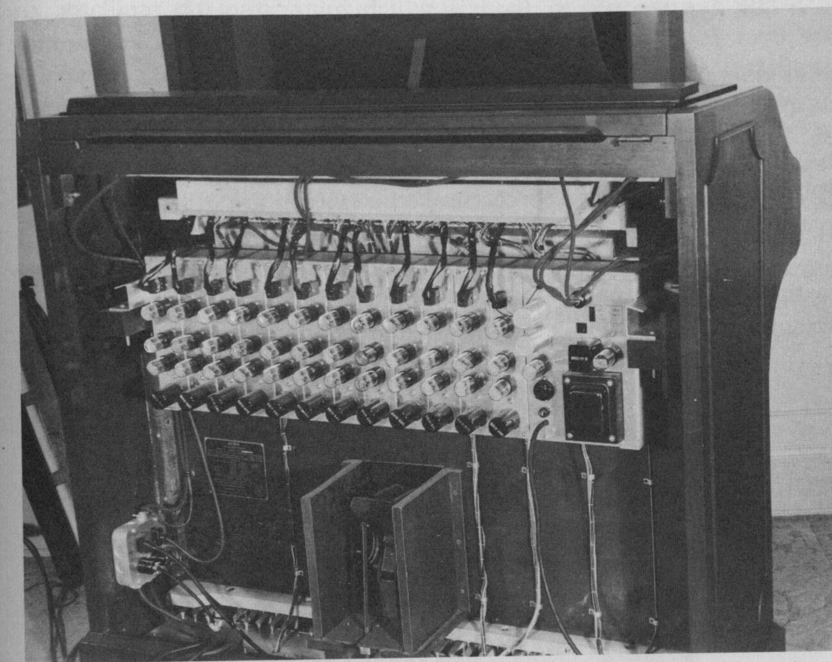


Fig. 6-3. Rear of the Baldwin shows tone generators.

R_2 passes part of the oscillator signal to the plate of the first frequency divider and also provides plate voltage for the top oscillator. The low side of the grid winding of the top oscillator is grounded through the vibrato oscillator grid circuit. The vibrato oscillator, located on the power-supply chassis, furnishes a signal at about seven cycles which serves to vary the top-oscillator bias at the same rate. Since the bias affects the frequency, there is a small frequency change at the vibrato rate.

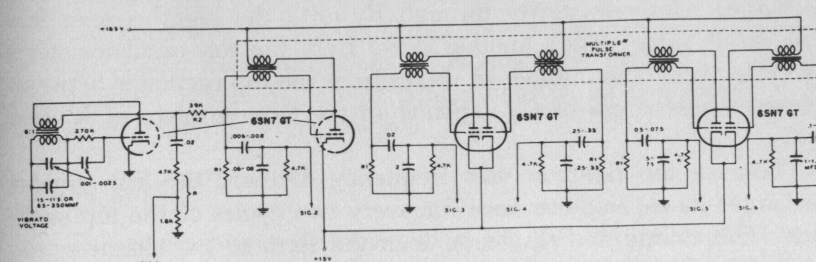


Fig. 6-4. Schematic of a generator chassis.

The five frequency dividers which follow the top oscillator are identical in principle, all being blocking oscillators. Like the top oscillator, each is a tuned-grid oscillating circuit. It has no tuning capacitor, however, and its natural resonant frequency, dependent on distributed capacitance of the transformer windings, is around 100 kc. It has a high L-C ratio and a very high grid-leak resistance R_1. (Values for R_1 and the unmarked capacitor are carefully adjusted at the factory.)

The plus 15 volts applied to the divider cathodes biases them so that they will not normally oscillate. The first wave transmitted through R_2 to the first blocking oscillator supplies enough extra volt-

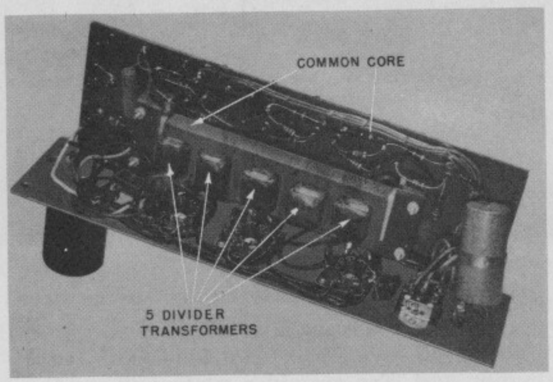


Fig. 6-5. Photo of a generator chassis shows the special multiwinding transformer.

age to start the divider oscillating at its high resonant frequency. Within a single cycle of this oscillation, however, the grid draws so much current through the high-value grid leak that the triode is biased beyond cutoff, and it stops oscillating. The resulting charge on C_1 discharges relatively slowly through R_1 until the bias becomes less than cutoff value. Then another wave from the top oscillator starts the cycle over. The number of waves from the top oscillator between successive discharges of C_1 depends on the time constant of R_1 and C_1-C_2.

Because the Baldwin uses frequency division, the first divider discharges its capacitance once for every two cycles of the top oscillator. The charge and discharge is in the form of the characteristic sawtooth, which is the wavefrom appearing across C_1-C_2. The two capacitors are used as a voltage divider, with the output signal voltage taken from the junction. The output, signal 2, is 1/10 of the voltage

appearing across the combination and is at a frequency just half that of the top oscillator.

The five divider transformers are all wound on a common core, as can be seen in Fig. 6-5. Thus, while there is no conductive connection between the dividers, the top-octave signal fed through R_2 into the multiple transformer affects them all in the same way. Each divider must, of course, lock at some submultiple of the top octave.

Figure 6-6 shows simultaneous wave shapes at various points in the generator circuits. The output of the top-octave oscillator is not very complex, but this is unimportant for the later tone-shaping circuits because the fundamentals are so high that most of the higher-

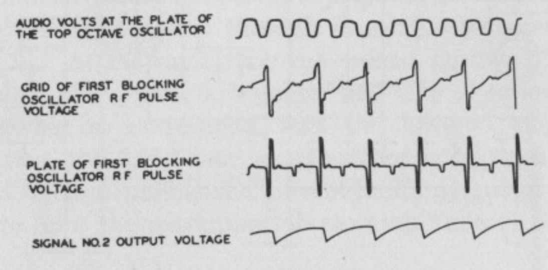


Fig. 6-6. Waveshapes at various points in the generator.

order harmonics would be inaudible anyway. The waveform at the plate of the first blocking oscillator shows the effect of the synchronizing waves supplied from the top oscillator – large pulses of high-frequency oscillation. The grid waveform of the blocking oscillator shows the resulting pulse of grid voltage caused by the grid current, followed by the gradual fall of voltage as the capacitance discharges. The signal 2 output voltage taken from the capacitive voltage divider is a simple sawtooth because of the filtering action of the capacitors.

The schematic of Fig. 6-4 shows for several components a range of values rather than a single one. Components so marked are dependent for their values on the particular note to be generated by the chassis in which they are placed.

POWER SUPPLY

The power supply diagrammed in Fig. 6-7 also holds the vibrato oscillator and an extra tone generator for the lowest C of the instrument.

The primary of the power transformer is provided with three taps, one of which is selected in accordance with the average a.c. line voltage in the location. The d.c. power is supplied by the 5Y3 rectifier and a filter consisting of choke L and capacitors C_2 and C_3. The filtered 260-volt line is fed to tubes in the tone-color box, where there are additional filter sections.

Resistors R_2, R_4, and R_5 form a voltage divider to supply plate and cathode voltages to the generator chassis. The B-plus, 165 volts, comes from the junction of R_2-R_4, and the cathode voltage comes from R_4-R_5. The ratio of 11:1 between the plate and cathode voltages

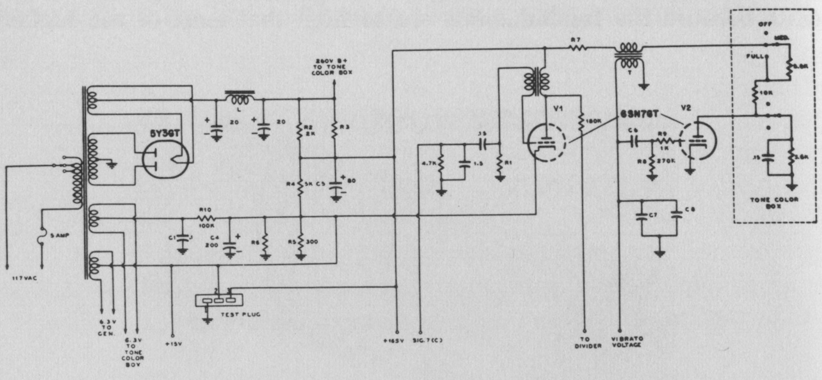


Fig. 6-7. Power supply for the entire organ.

must be maintained exactly to keep the oscillators dividing correctly. C_5 provides additional filtering for these two voltages. C_4 provides still more cathode filtering.

There are three different filament windings on the power transformer. The first is a 5-volt winding for the 5Y3. The next supplies 6.3-volt power for the 37 tubes (see next paragraph) in the generator assembly. The center-tap of that winding is connected to the +15-volt line so that the cathodes and heaters of the divider tubes are at the same potential and heater-cathode leakage is minimized. The last furnishes power for the tubes in the tone-color box. The center-tap is connected to +15 volts to minimize hum, and R_{10}-C_1 filters the connection so that no signal from the generators is transferred to the tone-color tubes.

The left section of the 6SN7, V_1, (Fig. 6-7), is the blocking oscillator for the lowest C of the instrument. Each tone-generator chassis has three tubes and can put out only six tones, one for each octave of the organ. The C series must have an additional note (there are seven C's in six octaves), so that each manual will go from C to C. The divider is identical to the others except that it receives synchro-

nizing signals from the plate of the lowest C on the C chassis through a 180,000-ohm resistor rather than through the coupling of a common transformer core. The output tone is obtained as in the other dividers.

V_2 is the vibrato oscillator. The feedback circuit includes the transformer T, C_6, and R_8 and R_9. The vibrato frequency is largely controlled by the inductance of T and the combined capacitance of C_7 and C_8. C_8 is selected at the factory as a final frequency adjustment, to fix vibrato rate. The grids of all the oscillators are grounded through the grid winding of the vibrato transformer.

The depth (amount of frequency swing) of the vibrato and the on-off control are combined in a three-position switch in the tone-color box. With the switch in the off position, the vibrato oscillator plate is disconnected from the power supply. In the center or medium position, plate current passes through R_7 and other components to the plate of V_1. At the same time the second portion of the switch shunts the plate of V_2 with 5,600 ohms and 0.15 μf to lower the frequency somewhat to correspond with the lowered amplitude. In the full position, the 5,600-ohm series resistor is bypassed and there is no shunt. R_7 is a permanent current-limiting resistor selected at the factory to limit the maximum vibrato amplitude.

THE KEYING SYSTEM

One of the most interesting features of the Baldwin electronic organ and one which contributes greatly to its outstanding performance is the keying system. Every designer of an instrument using continuously-running tone generators – and that means all instruments with synchronized octave-generator chains of any kind – is faced with the problem of key switching.

The chief cause of the trouble is key clicks. The average voltage represented by any recurrent a.c. waveform is zero, and, unless there is another source of d.c. involved, the a.c. voltage has no d.c. component. Keying a source of a.c. should therefore create no sudden rush of plate current in a following class-A amplifier stage, since the average plate current in such a stage does not vary (ideally) with a change in input signal level.

The fact is, however, that there is no way of knowing at what part of the a.c. cycle the key will close. And if it closes at any part of the cycle except one of the two instants when it is passing through zero, there is a sudden instantaneous change in the tube plate current from its resting value to some higher or lower value. This is illustrated in Fig. 6-8. Note that at the instant of key closure the plate current rises in a very short time and the vertical line looks like the leading edge of an excellent square wave. This very short rise time is in the nature of a part of a pulse with very small rise time containing a large number of harmonics at high amplitude; the result is a loud click in the speaker. The noisiness of ordinary switching in electronic music is further aggravated by any dirt or corrosion on the contacts or any lack of positiveness in the contacting. Either of these results in making and breaking the contact several times in quick succession, each time with a click.

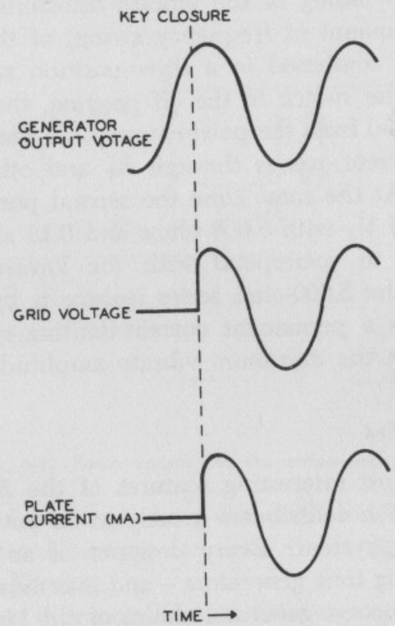


Fig. 6-8. Ordinary switching produces sharp transient.

The loudness of the click depends on the rise time of the plate current and the amplitude of rise, being greater as each of these increases. There are three possible solutions. The first, an impractical one, is to see that the switch closes only when the voltage is passing through zerc. The second is to place a low-pass filter after the switch somewhere in the system, eliminating the high harmonics which create the click (and effectively lengthening the rise time of the pulse). This second method is used in the Minshall organ described in the next chapter. The third method, especially useful when the waveform of the audio to be switched is sharply peaked and causes high-amplitude plate-current excursions, is to use a device which acts as a volume control and closes the circuit gradually. Then the initial plate-current excursion is very small and clicks are

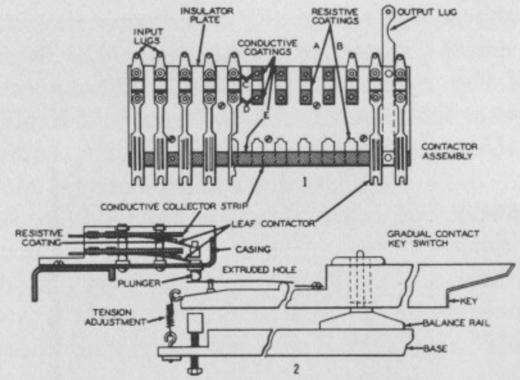


Fig. 6-9. Drawing shows construction of the Baldwin key switches.

inaudible, while the waveform, once it has come to maximum amplitude, is unimpaired. Such a "volume-controlling" or gradual-contact switch may be either mechanical or electronic.

A second and equally important advantage of a device of this kind is that it duplicates to some degree the action of conventional acoustic musical instruments, the attacks of which are (except for a few plucked- and struck-string instruments) gradual.

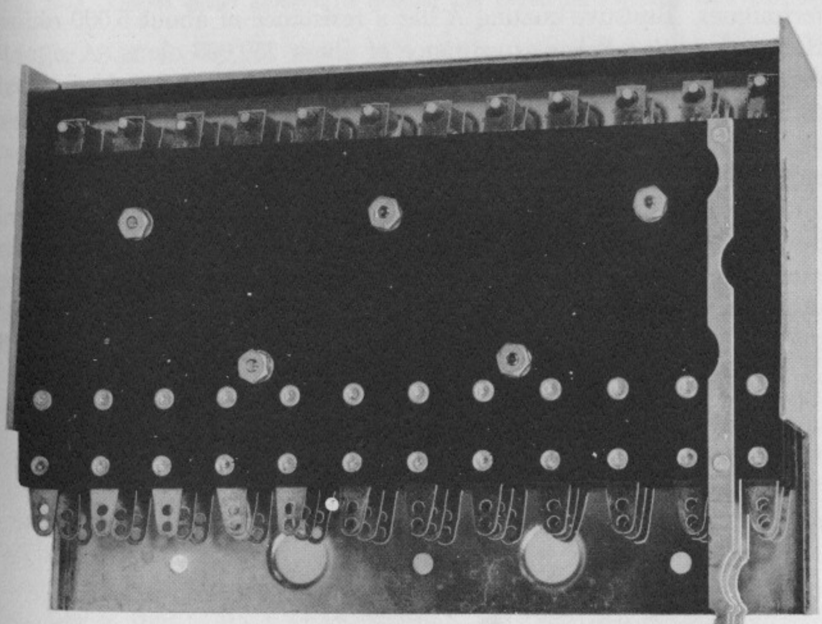


Fig. 6-10. A l-octave key-switch section for the great manual.

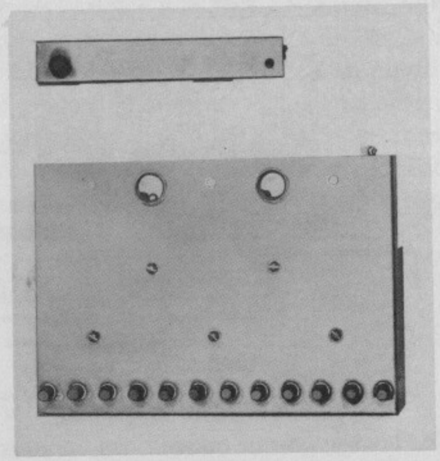


Fig. 6-11. Bottom of the switch assembly and a single switch. Rear of key presses wooden button, which closes switch.

The special switches employed in the Baldwin for gradual attack are drawn in Fig. 6-9 and pictured in Figs. 6-10 and 6-11. Note first the contactor assembly at 1 in Fig. 6-9. This is a flat insulator plate with conductive and resistive coatings applied by printed-circuit techniques. Resistive coating *A* has a resistance of about 5,000 ohms. Resistive coating *B* has a resistance of about 250,000 ohms. A signal-input lug is fastened to conductive coating *C*, and one end of a leaf contactor is secured to conductive coating *D*. A third conductive coating *E* contacts all the resistive coatings *B* and is connected to an output lug.

The scheme is diagrammed in Fig. 6-12. With the silver-plated beryllium-copper leaf springs in the resting position as shown, no signal from the generator passes to the following circuits. As pressure is exerted upward on the end of the spring, the spring first contacts the left end of resistive coating *B*, and gradually passes along it until

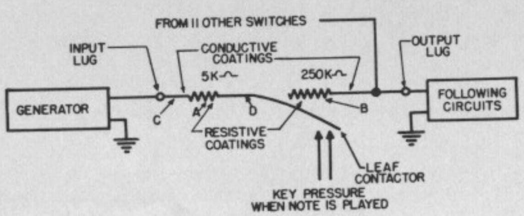


Fig. 6-12. Diagram of a single switch circuit.

it contacts conductive coating *E*, the circuit is then complete except for the permanent 5,000-ohm isolating resistance. Under these conditions, the plate currents of any following tubes will look like the waveform in Fig. 6-13. The steep initial rise is still there, at point *X*, but its amplitude is very small and the a.c. output builds up smoothly to the maximum, as the envelope shows.

The details of the gradual-contact switch are shown in 2 of Fig. 6-9. There are actually 24 or 36 of them on each assembly, depending on whether the assembly is to be used in the great or swell manual. This is in two or three horizontal lines of twelve each. We shall explain the reason for the two or three groups later. The photograph

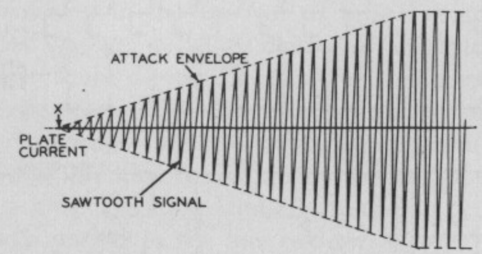


Fig. 6-13. Rise of sound envelope is gradual and causes no high-amplitude transient.

of Fig. 6-10 shows the upper side of the key switch assembly, with the 36 input lugs (this is a three-stack assembly for the great manual) and the three output lugs. The ends of the leaf contactors, with their wooden actuators, can be seen under the insulator plates at the top. The two extruded holes at the bottom are for cables. Figure 6-11 shows the bottom of the assembly, with the actuator buttons coming through the extruded holes. Above the twelve-key switch assembly is a single-key assembly consisting of two gradual switches, one above the other, used, as will be explained later, in the stop switches and for the pedal clavier.

THE KEYING CIRCUIT

The generator assembly provides 73 tones, from the third C below middle C to the third C above middle C. Figure 6-14 shows the switching schematic for the 8-foot register on one of the manuals. (This means that when the middle-C key, for example, is struck, the tone switched is actually middle C, 261.7 cycles, not an octave above or below that.) Since a manual includes only five octaves of keys, only five of the six octaves of tones generated are used here, octaves 2 through 6.

The tone from each generator is brought to the upper end of the 5,000-ohm isolating resistive coatings, thence to the leaf-spring contactor. Any of the tones of octave 4 which are switched in by pressing a key in that octave go to the common octave 4 collector strip. The collector strips of all the 8-foot switch assemblies go to a resistive network. The purpose of the network is to attenuate the tones from each octave in increasing degree. At the output in Fig.

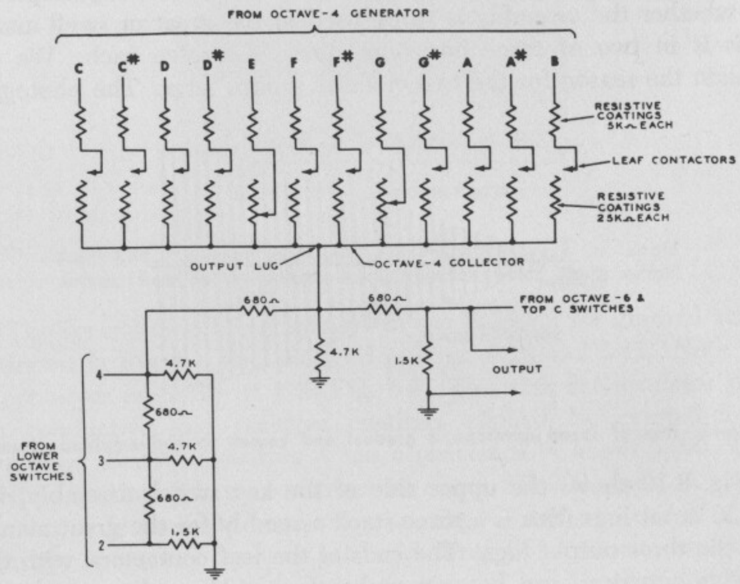


Fig. 6-14. Diagram of key-switching system for one register of a manual.

6-14 the highest tones have the largest amplitude and the lowest tones the smallest. The reason for this will become apparent when we discuss the tone-coloring system.

Each manual has a switching system like this, each system using one bank of twelve gradual-contact switches for each octave (and a single switch for the uppermost C). The great manual has two more of these switching systems, using the other two banks of switches in its octave assemblies. One system is a 16-foot register, meaning that when middle C is keyed, the tone actually heard is one octave below middle C, and so on. The 16-foot system makes use of the tones generated in octaves 1 through 5. The third switching system is for the 4-foot register, and uses octaves 3 through 6. Since there is no seventh octave, normally necessary to provide tones one octave above the highest keys on the manual, the upper octave of 4-foot switches

simply repeats the tones of the octave below in the 4-foot register; i.e., the highest octave sounds in the 8-foot register.

We thus have three separate switching systems in the great manual, though all are actuated every time a key is pressed. Each system has an output so there are three outputs in all (plus an extra 4-foot output taken from the low-frequency end). The swell manual has only 8- and 4-foot registers, with a total of two outputs. The pedal clavier has only 16- and 8-foot registers, making a total of two outputs. All these outputs are fed to the tone-color box and each output provides sawtooth waves.

TONE COLORING SYSTEM

Tone coloring in the Baldwin is built solidly on the theory of formants (see Chapter 4). Two classes of organ tones are important, in addition to those discussed earlier – the diapason family and the "stopped" tone colors. Both have the peculiarity of having prominent odd harmonics; while even harmonics appear with some effect in the diapasons, they are almost completely missing in the stopped colors. These tones have a hollow kind of sound like those produced by a clarinet when played in the low register.

To produce the stopped tones it is necessary not only to pass them through formant filters like other types of tones but to begin with a generator waveform which has almost entirely odd harmonics. The sawtooth has both odd and even, and an ingenious system is used in the Baldwin to eliminate the evens. It is done by mixing octave-related sawtooth waves to produce a square wave which, being symmetrical, is composed almost entirely of odd harmonics which create that pleasing tone effect.

The mixing process is illustrated graphically in Fig. 6-15. When a given key on the swell manual is pressed, an 8-foot and a 4-foot tone are switched into their respective networks and emerge through the network outputs. They have approximately the same amplitude, a sawtooth waveshape, and are in phase.

A special outphasing circuit is employed for the mixing, and (A) of Fig. 6-15 shows how the waves are mixed. X is the 8-foot, or lower-frequency, wave. Y is the 4-foot wave, one octave above wave X. Wave X has been reversed in phase before the mixing and wave Y has been reduced to half the amplitude of X. Now, by simple graphical analysis, it can be seen that the resultant is a square wave, as in B of Fig. 6-15.

The mid-point or average value of instantaneous values of waves X and Y, when added graphically or algebraically, produces or defines the resultant square wave. The reader can prove this for himself.

At each of several points in (A) of Fig. 6-15, place a point at the resultant value of voltage. This point is half-way between the individual values of the two waves. When all the points are connected the result will be a square wave.

Let's look at it in another way. We start with an 8-foot tone of, let us say, 1,000 cycles. Its second, third, and fourth harmonics are 2,000, 3,000, and 4,000 cycles. We mix with it a 4-foot tone of 2,000 cycles, in phase opposition. Each harmonic of the second tone will buck out any harmonic of the first tone whose frequency coincides, since the two are in phase opposition. The coinciding frequencies are all the even harmonics of the lower-frequency tone, leaving only

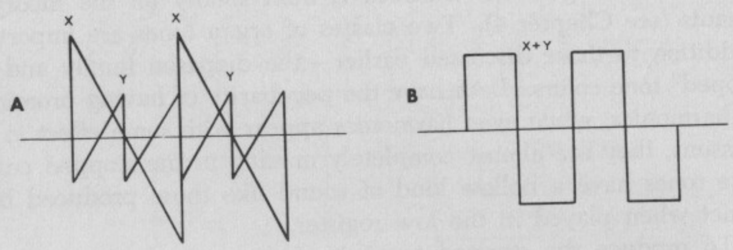


Fig. 6-15. How the two sawtooth waves mix to produce a square wave.

the odd-numbered frequencies, 1,000, 3,000, 5,000, and so on. The resultant is a practically square wave of the 8-foot fundamental frequency. The circuit which does the outphasing appears in Fig. 6-19, to which we shall refer later.

Figure 6-14 showed the keying circuit with which all the tones of the generators are keyed and channeled through successive attenuator networks to their outputs. Now we shall see what happens next to these tones.

THE TONE-COLOR BOX

The tone-color box is shown in Fig. 6-16. It is a shallow metal chassis in the form of a tray containing all the *R-C-L* filters which form the tone colors for the various stops in accordance with the formant principle. The box is just under the top cover of the organ console. Its components can be reached by first removing the wooden top, then unscrewing and removing the metal top of the box itself.

The tone-color box also contains three tubes – one for the outphasing circuit and two for the preamplifier which follows all the filters. Along its front edge are ranged all the stop switches; these

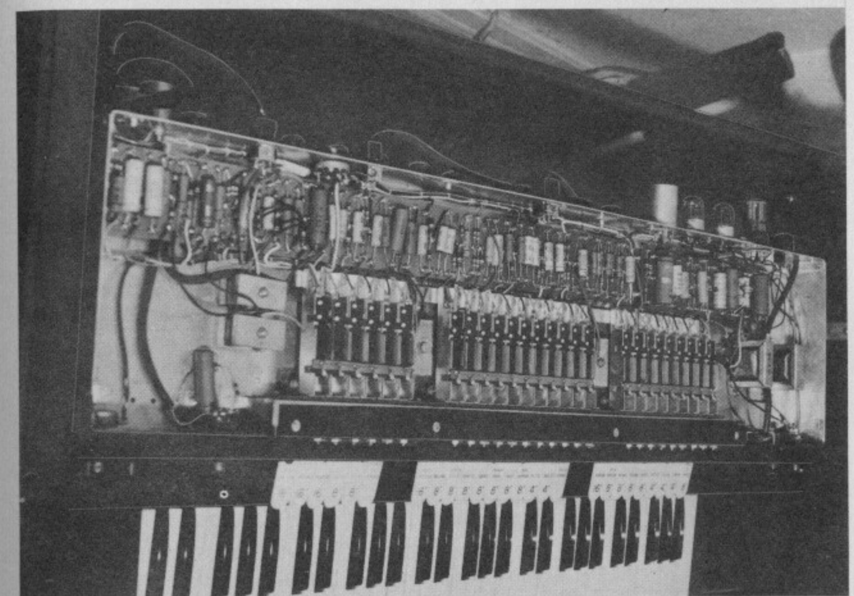


Fig. 6-16. The tone-color box.

are gradual-contact switches of the same type used for keying which follow each filter. For simplicity they are shown as ordinary s.p.s.t. switches in the diagrams. The resistor *R* preceding each one is the 5,000-ohm printed-circuit resistance between the input terminal of each switch and the leaf contactor.

Figure 6-17 shows the tone-coloring system for the pedal clavier. There are two outputs from the pedal switching system – 8-foot and 16-foot. The pedal 8-foot FLUTE stop is a soft-voiced one, used to add clarity to the pedal in soft music. Let us analyze the filter which produces it to show how the system works.

The filter is made up of three L-section *R-C* filters in cascade, each giving, theoretically, an attenuation of about 6 db per octave above its turnover frequency. The 47,000-ohm resistor and .01-μf capacitor have a turnover frequency of about 320 cycles. The next two sections have turnovers of about 160 and 65 cycles, respectively. The resultant curve of the entire filter has low-pass action, with a slope carefully engineered to produce the most lifelike and pleasing tone. The spectrum of this curve agrees closely with the typical flute spectrum shown in Fig. 4-3, Chapter 4.

Most of the stop filters do not have attenuation curves as steep as that of the 8-foot pedal FLUTE, and the curves vary widely. In general, however, most do have low-pass action. While this attenu-

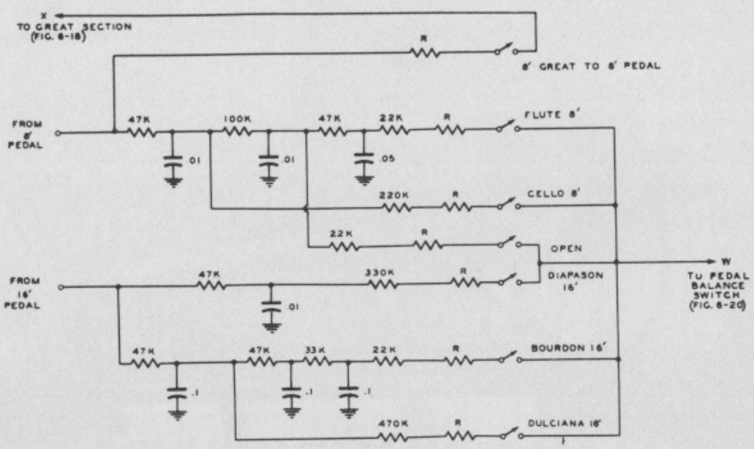


Fig. 6-17. Pedal tone-color filters.

ates harmonics of each generated tone to form the correct spectrum characteristic, it also has the effect of attenuating the fundamentals of the generated tones as the tones become higher in pitch. The attenuator network following the switches, shown in Fig. 6-14, partially offsets this. The outputs of the switching networks are taken from the high-frequency end, which means that the higher tones are initially louder. If this were not done, some of the higher notes would be barely audible. As each of the stop qualities is described below, examine the filter (Fig. 6-17) and note how the effect is achieved.

The 8-foot pedal CELLO is a string-type stop, moderately voiced and somewhat like the orchestral cello. The diapason family is the backbone of the organ, not imitative of any orchestral instrument. The pedal 16-foot OPEN DIAPASON is voiced rather loud. Notice that it has a fairly large harmonic content, with only a single-section *R-C* filter from the 16-foot sawtooth source. Note, too, that it has a certain amount of 8-foot tone added. As with a few other stops, a dual switch is required to avoid disturbing the busses. The loudness of a particular tone quality is governed by the resistor which follows the filter, just preceding *R* (as well as by the filter itself, of course).

The 16-foot BOURDON is a softly voiced stop of the flute family, with great depth and clarity. The 16-foot pedal DULCIANA is as the name implies, a very soft tone, and belongs to the diapason family. The output lead of the pedal department (marked W) goes to the preamplifier.

The great manual is the lower one and its filter schematic appears in Fig. 6-18. The 16-foot GREAT BOURDON is a soft flute stop which adds body to an ensemble.

Note that the 4-foot CLARION filter is fed from the *low-frequency* end of the 4-foot great switching network. This is because it is a high-pass filter and deals with high-pitched tones. If it were fed from the high-frequency end, the higher notes would be too pronounced and the lows would almost disappear.

The remainder of the great stops have two inputs each. One is from the 8-foot or 4-foot great switching network, as shown, but the other is from lead X in the pedal department, where one of the stop switches is labeled 8′ GREAT TO 8′ PEDAL. This is a coupler, and when the switch is closed the tones coming from the 8-foot pedal switching network pass not only through the pedal stops but also through whatever great stops have been selected. Thus the resources of the great can be made available to the pedal as well. The great also has a coupler (Fig. 6-18), SWELL TO GREAT 8′. This coupler sends 4-foot great tones to the 4-foot swell stops and 8-foot great tones to the 8-foot swell stops. Thus when playing on the great manual the swell stops can be used in addition to the great stops. The swell has no couplers and when playing on the swell manual the player can use only the swell stops.

The great 4-foot CLARION is a very keen reed tone of great brilliance. It has a definite formant range in which fundamentals and harmonics are greatly emphasized, due to the resonant L-C filter. The great 4-foot VIOLINA is a string-type stop with a high-pass characteristic. The OCTAVE is a diapason tone in the 4-foot register.

The 8-foot TRUMPET is a loud, heavy-voiced reed, again with a resonant filter. It is a surprisingly good imitation of the orchestral trumpet when played in certain ways. The 8-foot great DULCIANA is much like the pedal DULCIANA, but in a higher register. The 8-foot MELODIA is a soft flute-type tone. The 8-foot OPEN DIAPASON possesses the tone quality of which the average person immediately thinks in connection with organs. It is extensively used to accompany a choir. It is the basic organ tone, heavy enough to give a good, solid foundation, and bright enough (note that some 4-foot-register tone is added to it) to make a melody stand out. The OPEN DIAPASON is overused by inexpert organists who count on sheer heaviness and volume for effect.

The great manual is ordinarily used for accompaniment and for full-bodied playing. The swell, the stops of which are diagrammed in Fig. 6-19, is often used for solo playing and includes all the solo stops. Frequently the swell is used as a monophonic instrument, a

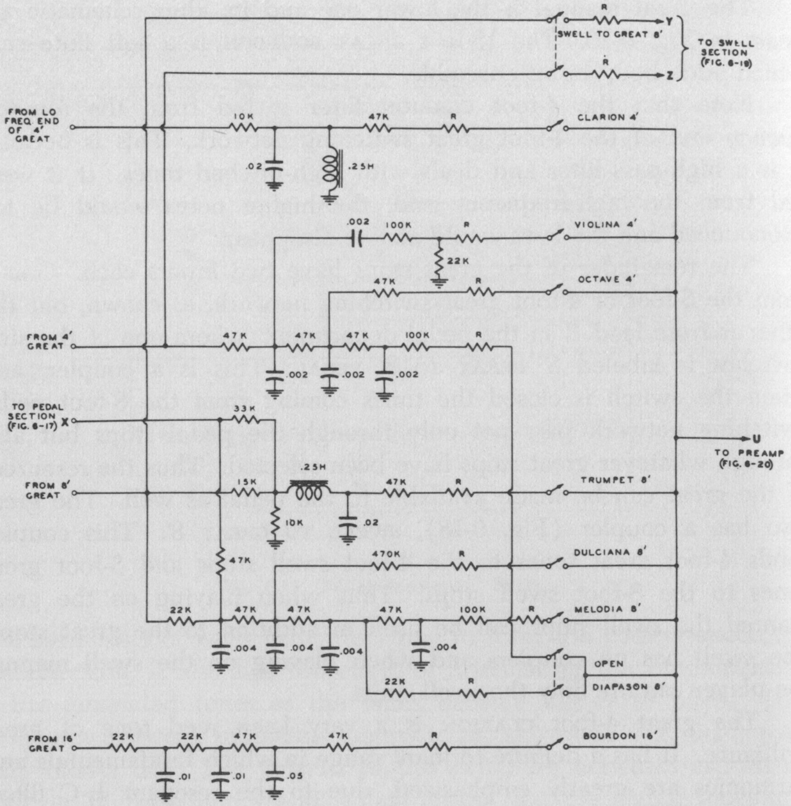


Fig. 6-18. Baldwin great tone-color filters.

"one-finger" melody played on its OBOE stop, for instance, with an accompaniment on the great MELODIA or DULCIANA and a foundation of 16-foot DULCIANA on the pedals.

Most of the swell filters have two inputs, as have the great. The second input in each case carries tones from the great coupler so that stops selected on the swell may be heard when the great is played.

The DOLCE CORNET is a reed stop rich in upper harmonics because of the high frequency of the resonant filter (above 2,000 cycles). The 4-foot SALICET filter is a high-pass unit giving a very keen string tone; it is softly voiced, however, by the 220,000-ohm series resistor. The 4-foot FLUTE is a soft tone without very much harmonic content. The 8-foot OBOE is one of the finest solo stops on any good pipe organ, and it is particularly good on the Baldwin. It is of the reed family but rather mildly voiced. It is an excellent imitation of the orchestral oboe, with its rather nasal tone but without enough sharpness to be irritating.

The FRENCH HORN sounds much like its orchestral namesake; it is very useful for solo work. The TROMPETTE is a moderately voiced reed stop somewhat like the TRUMPET but not like any particular orchestral instrument. (Organ terminology calls certain stops "reeds" even though their orchestral counterparts are horns, because reeds produce these tones in most pipe organs.) The SALICIONAL is the basic string

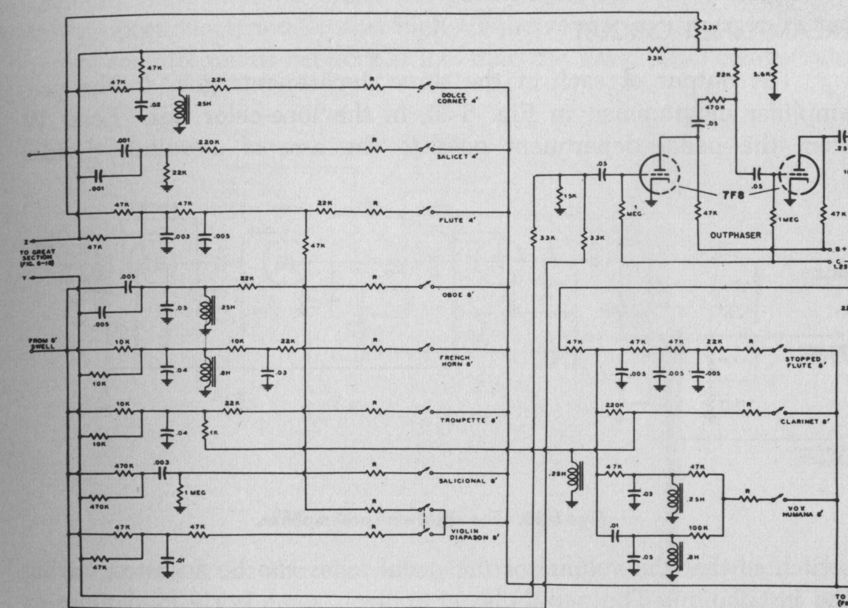


Fig. 6-19. Tone-color filters for swell stops.

stop of the swell department. The VIOLIN DIAPASON is a moderately voiced diapason tone with some string quality added for increased brightness.

The remaining three swell stops are not fed directly from the swell key switching network but from the outphaser whose action was described earlier and pictured in Fig. 6-15. Eight-foot tones are fed to the input of one 7F8 triode, both from the swell key network and from the swell-to-great coupler. At the output of the first triode the phase is reversed. The tones are then fed through the second triode, along with 4-foot tones mixed in, at the proper strength, between the two stages. The output, consisting of 8-foot tone with even harmonics very much attenuated, is fed to the next three stop filters.

The STOPPED FLUTE is a fairly heavy flute tone. Since only odd harmonics are heard, it has a hollow sound peculiar to woodwind instruments and stopped organ pipes (pipes with one end closed.

The CLARINET is much like the orchestral instrument, again with the typical hollow, woodwind sound. The VOX HUMANA, originally conceived in organ tradition as imitating the human voice, is a reed stop of unusual quality. In the Baldwin this is produced by the out-phaser and by the fact that it has two distinct formants, one around 600 cycles and the other in the neighborhood of 2,000 cycles.

PREAMPLIFIER CIRCUIT

The output of each of the three departments goes to the preamplifier diagrammed in Fig. 6-20, in the tone-color box. Lead W from the pedal department goes to the arm of a voltage-divider

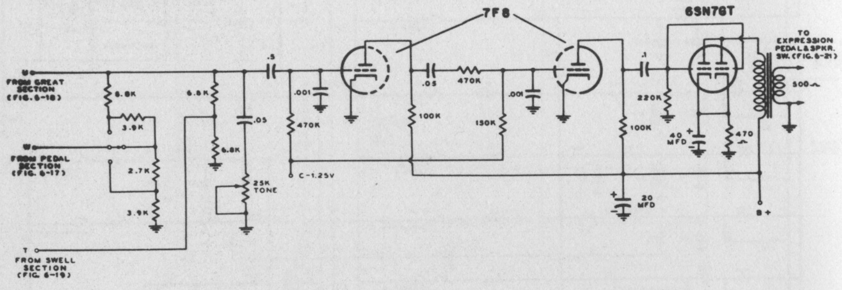


Fig. 6-20. The Baldwin preamplifier.

switch so that the volume of the pedal tones can be adjusted during the installation. The pedal clavier controls some fairly loud tones as low as 32 cycles. At this low-frequency, sound diffuses extensively and windows, fixtures, and furnishings may tend to vibrate. The setting of the pedal balance switch depends on the acoustics of the room or hall in which the organ is located.

Lead T from the swell section enters the preamplifier through a simple isolating network, while lead U from the great department enters without any isolation resistor of its own, the isolation of the other inputs being sufficient. The .05-μf capacitor and variable 25,000-ohm resistor between preamplifier grid and ground make up a tone control. With the resistor at maximum resistance, the capacitor has least shunting effect and tone is most brilliant. As the resistor value is reduced, more and more of the higher tones and harmonics are shunted to ground. The setting of this control, which is on the front of the console panel, depends somewhat on the acoustics of the room or hall, but is often varied during playing. It is seldom left at the full brilliant position, especially when, as in some cases, special speaker systems with high-efficiency tweeters are used. The preamplifier terminates in a 500-ohm transformer.

Two additional controls appear on the organ console. The first is the usual expression or swell pedal which controls volume. The pedal operates a potentiometer (Fig. 6-21), which is connected to the preamplifier output and returns through a tone-compensating and limiting network to ground. The 22- and 220-ohm resistors and the 10-μf capacitor constitute a lower net impedance for high frequencies than for lows. When the potentiometer arm is at the bottom the proportion of middle and high frequencies appearing at the output of the attenuator network is less than the lows. This compensates

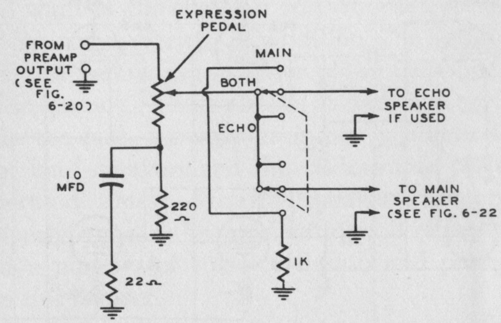


Fig. 6-21. The organ output network.

to some extent for the characteristic of the human ear which hears less bass when volume is low. It is, in effect, a simple loudness control, or compensated volume control of the type used on many receivers. The attenuator output goes to a panel switch with three positions.

In pipe organs one set of pipes known as the echo organ, is sometimes located at a distance from the others so that the effect is that of having tones float in from a distance. The effect is especially marked in a very large church with strongly reflective walls. This effect can be produced in the Baldwin by adding another power amplifier and a separate set of loudspeakers at some distance. The switch of Fig. 6-21 connects the attenuator output to either the main speakers or the echo speakers, or to both. When only one set is being used, a 1,000-ohm resistor shorts the line to the other set preventing an open-ended line.

TONE CABINETS

Several different styles of tone cabinets containing loudspeakers and power amplifiers are available. At least two 15-inch loudspeakers are used in all of them. It is possible, however, to employ any amplifier and speaker combination desired and some installations are very elaborate indeed. Two amplifier models are supplied, one with 20-watt maximum output and the other with 40-watt. The 40-watt amplifier is diagrammed in Fig. 6-22. It is simple but quite adequate. It has only two stages, an input and phase-inverter stage combined, and a paralleled push-pull output, with inverse feedback from output transformer secondary to the input triode cathode. So-called

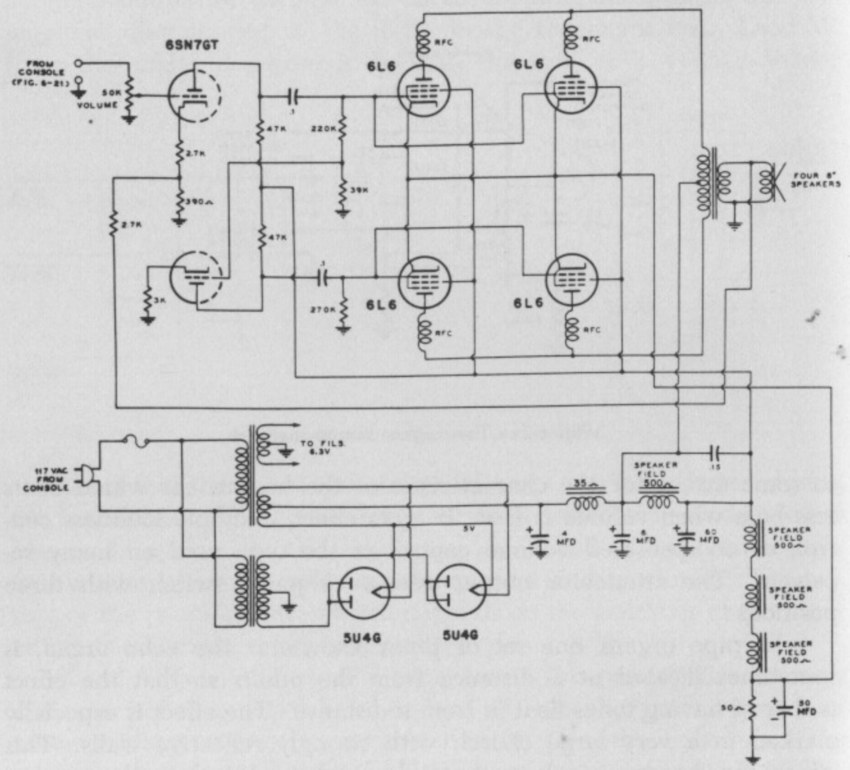


Fig. 6-22. Diagram of a Baldwin 40-watt amplifier.

high fidelity is not a requirement in an organ from a frequency response standpoint, although freedom from intermodulation is. There is, however, a very marked difference in the way the tone colors come out when various amplifiers and speaker systems are used. This is not surprising in view of the formant principle, in which any element of the system which affects the spectrum or frequency-response characteristic alters the tone quality.

Fixed bias of a sort is used in the amplifier; the speaker fields

across the power supply form a voltage divider with the 50-ohm resistor at the bottom. The potential appearing across this resistor, 19 volts, is fed to the 6L6 cathodes.

A bleeder current of about 200 milliamperes flows through the speaker fields. This is roughly equivalent to the current drain of the push-pull paralleled 6L6 output tubes and serves to stabilize the bias on these tubes.

As with any musical instrument, the acoustics of the installation are of paramount importance. The ideal installation includes a tone chamber for the loudspeaker units. The speakers can be faced toward one of the hard walls of the chamber so that when the sound emerges it has diffused to some extent. In addition, the multiple reflections from the walls and ceiling cause some phase shift among the tones, changing with frequency. These effects, as well as a certain amount of reverberation in the hall, are very desirable in eliminating the point-source effect of loud-speakers and the undesirable perfection of electronically generated tones. Straight speaker cabinets, when they are placed properly can also be used with good effect. The company has established a number of rules of thumb and procedures to help the installers in this respect.

When correctly installed, the Baldwin organ is a fine musical instrument – entirely aside from the ingenuity which has gone into its design. It is suited not only for church work, but has been widely accepted for concert performance in both the baroque and modern styles and is equally suitable for theater and radio music. While not superior to a really good pipe organ, it is better than most. It represents a truly happy wedding of the art of the electronic engineer and the music-maker.

Chapter 7

The Minshall Organ

One of the principal problems which manufacturers of organs have is economy of design consistent with good performance. When the market was confined almost exclusively to churches this problem did not loom so large because churches, while most of them are by no means wealthy, can at least, in effect, pool the funds of many individuals to pay for an instrument. The home organ buyer, however, is using only his own money and now that he is an important customer prices must be scaled down. This is a problem because organs are inherently somewhat expensive. They must, for example, provide at least one separate tone generator for each of at least 60 notes, and this alone leads to volume consumption of tubes and other components.

One of the most economical and interesting designs is that of the Minshall electronic organ (made by Minshall Organ, Inc., Brattleboro, Vt. — formerly Minshall-Estey but now having no connection whatever with Estey). The Minshall tone generators employ only half a tube, 4 resistors, and 3 capacitors per note. Similar economies occur throughout the instrument. Yet it produces a great variety of tones which are imitative of pipe-organ sounds.

The company is one of the youngest in the field and this was its first commercial realization. Since then, however, a good deal of change has been made to overcome faults which extensive field reports brought to notice and to increase very greatly the quality and variety of tonal resources. In this article we shall see the organ and its components and point out improvements.

There are several standard Minshall models. Top of the line is the LC shown in Fig. 7-1. This is a full 2-manual unit with 25-note pedal clavier. It is distinguished from the Model L, shown in Fig. 7-2, by the fact that the LC has a second 4-octave set of tone generators which may be tuned slightly sharp or flat of the main generator set and can be switched in to give a "celeste" effect. The two models are

Fig. 7-1. The largest Minshall organ, Model LC, which has a chorus generator.

Fig. 7-2. Model L, described in this Chapter.

Fig. 7-3. The spinet, Model S, has abbreviated manuals and pedals.

Fig. 7-4. The Minshall Chord Organ may also be played without using the chord buttons.

otherwise identical. Figure 7-3 shows the spinet Model S, which has two offset 44-note manuals and 13 pedals; the S has about the same registration as the larger models despite its smaller size and lower price, which makes it one of the most versatile spinets offered. The Chord Organ (no longer made) is shown in Fig. 7-4. This is a single-manual unit. It may be played in the standard way, but for chord use ten chord buttons are provided. These are used with the lower-octave keys of the manual to produce ten chords in each of the 12 keys, 120 in all. In this article we shall describe the Model L in detail. All the other models are essentially similar except for the chord organ which would require a separate article.

OVER-ALL SCHEME

A simple block diagram in Fig. 7-5 gives a general idea of the layout of the organ sections. Each of 12 tone-generator chasses produces five notes separated by octaves, so that 60 notes are generated in all. These are wired to a key-switch assembly for each manual. Each key-switch assembly has three output busses, one each for 4-, 8-, and 16-foot tones. Each bus goes to an amplifier, after which the tones are fed through tone filters. The original tones are roughly sawtooth in shape and contain considerable harmonics; the tone filters, selected by

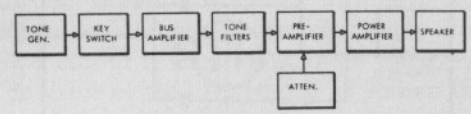
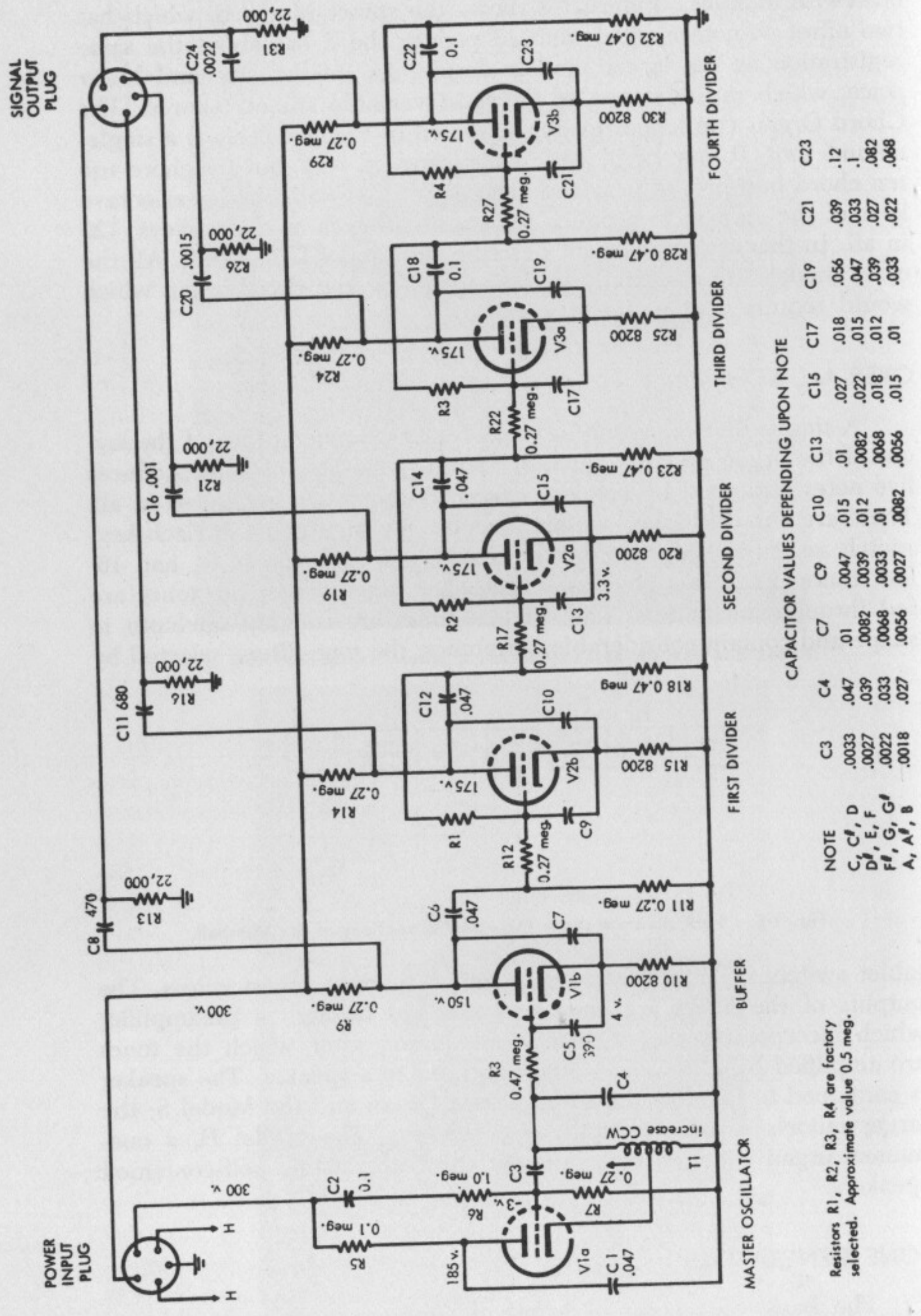


Fig. 7-5. Block diagram shows the essential sections of the Minshall.

tablet switches, shape the tones to imitate various organ voices. The outputs of the filters are combined and fed through a preamplifier which incorporates the swell-shoe attenuator, after which the tones are amplified by a power amplifier and fed to a speaker. The speaker is contained in the console of the Chord Organ and the Model S; the larger models employ separate tone cabinets. The Model H, a one-manual organ which is being discontinued, also uses a self-contained speaker.

TONE GENERATORS

The basic distinction of Minshall electronic organs has always been the tone-generator system. The principal virtue of the circuit is that it is inexpensive and yet reliable.



NOTE	C3	C4	C7	C9	C10	C13	C15	C17	C19	C21	C23
C, C#, D	.0033	.047	.01	.0047	.015	.01	.027	.018	.056	.039	.12
D#, E, F	.0027	.039	.0082	.0039	.012	.0082	.022	.015	.047	.033	.1
F#, G, G#	.0022	.033	.0068	.0033	.01	.0068	.018	.012	.039	.027	.082
A, A#, B	.0018	.027	.0056	.0027	.0082	.0056	.015	.01	.033	.022	.068

Fig. 7-6. The new tone generators are similar to the old ones but have several improvements.

The complete circuit of a tone-generator chassis appears in Fig. 7-6. Each of the frequency dividers requires only one-half of a 12AX7, with no transformers or other expensive components. The resistors and capacitors used are all standard 10 per cent tolerance components, none having to be specially selected except for R_1 through R_4. The frequency ranges over which a generator with one set of values will operate and divide properly is over half an octave. There are four dividers on a chassis, however, and the useful ranges overlap somewhat so that the total range for the chassis may possibly be less than five semitones. For this reason, the required total 12-semitone range for all the generators has been divided into four sub-ranges of three semitones each and four sets of values are used to cover the total. The values for these are all shown in the table at the bottom of Fig. 7-6.

The master oscillator which may be tuned to set the organ in tune is a well designed Hartley. To allow for vibrato creation by variation of d.c. element voltages, the time constant of the grid-leak network R_7-C_3 has been made slightly smaller than would be required for optimum stability and an a.c. path has been added between plate supply and grid consisting of C_2 and R_6. When the plate supply voltage is varied at a rate between 5 and 8 cps, the frequency varies at the same rate. The use of the L-C oscillator is a part of the new models and is an improvement over the old design in which R-C phase-shift oscillators were used. The R-C oscillators tended to be somewhat unstable over a period of time because of change in tube plate resistance.

The frequency-divider stages are effectively voltage amplifiers in which the plate output is used to charge a capacitor between plate and cathode. The value of the capacitor is so chosen that it can charge only at a rate in the neighborhood of half the input frequency. The fundamental component of the plate voltage is then fed back to the grid in such phase as to cause the tube to cut off during every other input cycle; this causes the alternate positive peaks of the input wave to make the tube conduct and produce plate-current pulses at half the input frequency. The action, like that of most feedback systems, is hard to describe in a few words; complete details can be obtained elsewhere.

The plate output is the result of a capacitor which is charged and discharged, and therefore takes the waveform of a sawtooth, as illustrated at (A) in Fig. 7-7. As can be seen, the flyback time of the sawtooth is quite large — at least 20 per cent — and the harmonic content is not very great. This was a fault in earlier models and made it impossible to secure any really bright tone qualities or, indeed, to have any really satisfactory variety of tone colors. In addition, the high-impedance key-switch system had to be of the shunt type because of leakage through the capacitance of open switches.

In the new circuit the plate outputs are not used directly. They are first passed through differentiators, which may be looked on as high-pass filters. They consist of C_8-R_{13}, C_{11}-R_{16}, etc., in Fig. 7-6. They have two functions. First and most important, they change the harmonic structure of the waves, making the harmonics much more prominent with respect to the fundamental, as may be seen in the resulting waveform of (B) in Fig. 7-7. With this improvement, the new models

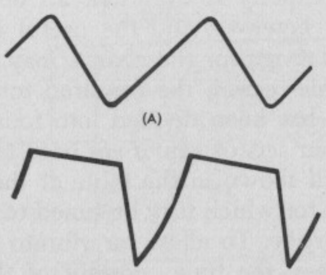


Fig. 7-7. Waveshape at (A) is that given by the generators proper. The shape at (B) has higher harmonic content and is obtained by differentiation.

have very satisfactorily bright and interesting reeds and strings and a very good variety of colors. The second function, incidental but useful, is in voicing. By selection of the capacitor elements of the differentiators, the over-all level of the higher notes is made greater than that of the lower ones. When all tones are passed through the later formant filters which are mostly of the low-pass type, the total scale tends to have more even loudness from top to bottom than if all incoming tones to the filters were of the same level. This is the same job done in the Baldwin organ by networks between octaves in the keying bus outputs and in the Schober Organ Kits by varying-value resistors in series with each key switch.

There are two additional improvements in the new Minshall generators. The first is that a cathode-bias resistor has been added in each

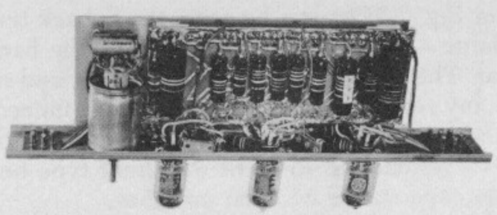


Fig. 7-8. Side view of a tone-generator chassis shows the printed circuit which holds most of the components.

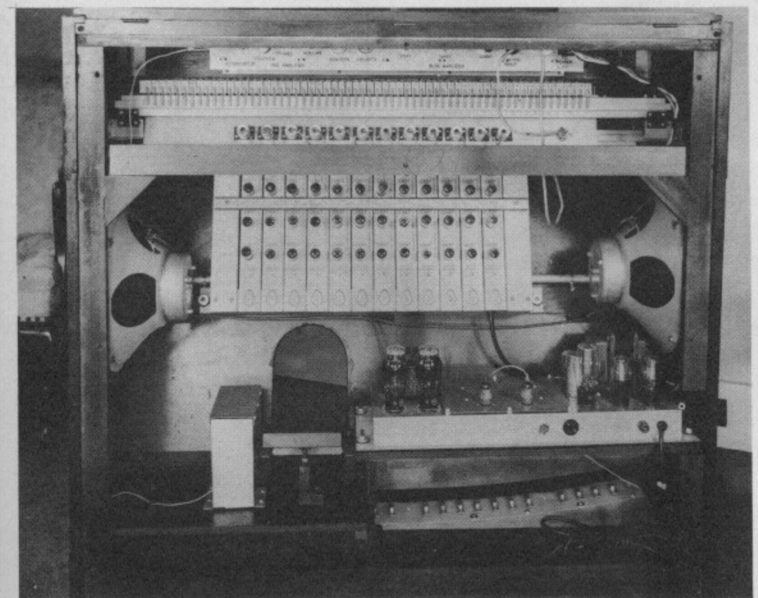


Fig. 7-9. Rear view of the Model L, showing the tone generators, power-supply and power amplifier, and tablet-board chassis.

divider stage. There is now less possibility than before that a change in some component or voltage will cause a misfunction, since an unbypassed cathode resistor tends to be a compensating factor, holding the tube at about the same operating point despite changes in other factors. The second improvement is that all the divider grids are direct-coupled. In the former design the grid impedances were extremely high and weather variations would sometimes cause trouble. The new arrangement brings grid impedances down to normal equipment values and this problem is eliminated.

Figure 7-8 shows how a tone-generator chassis looks. Notice that almost all the resistors and capacitors are mounted on a printed-circuit panel. The printed circuit makes for neat and inexpensive production and easy servicing since each component may be lifted or removed without disturbing others and may even be put back again without harm if found to be good. Figure 7-9 shows the rear of the organ. To remove a generator, the divider strap is loosened and the generator is simply pulled out; it is held in place only by the power and output plugs on its ends.

The key-switch circuitry of the new models is much changed from the old system, both electrically and mechanically. The high impe-

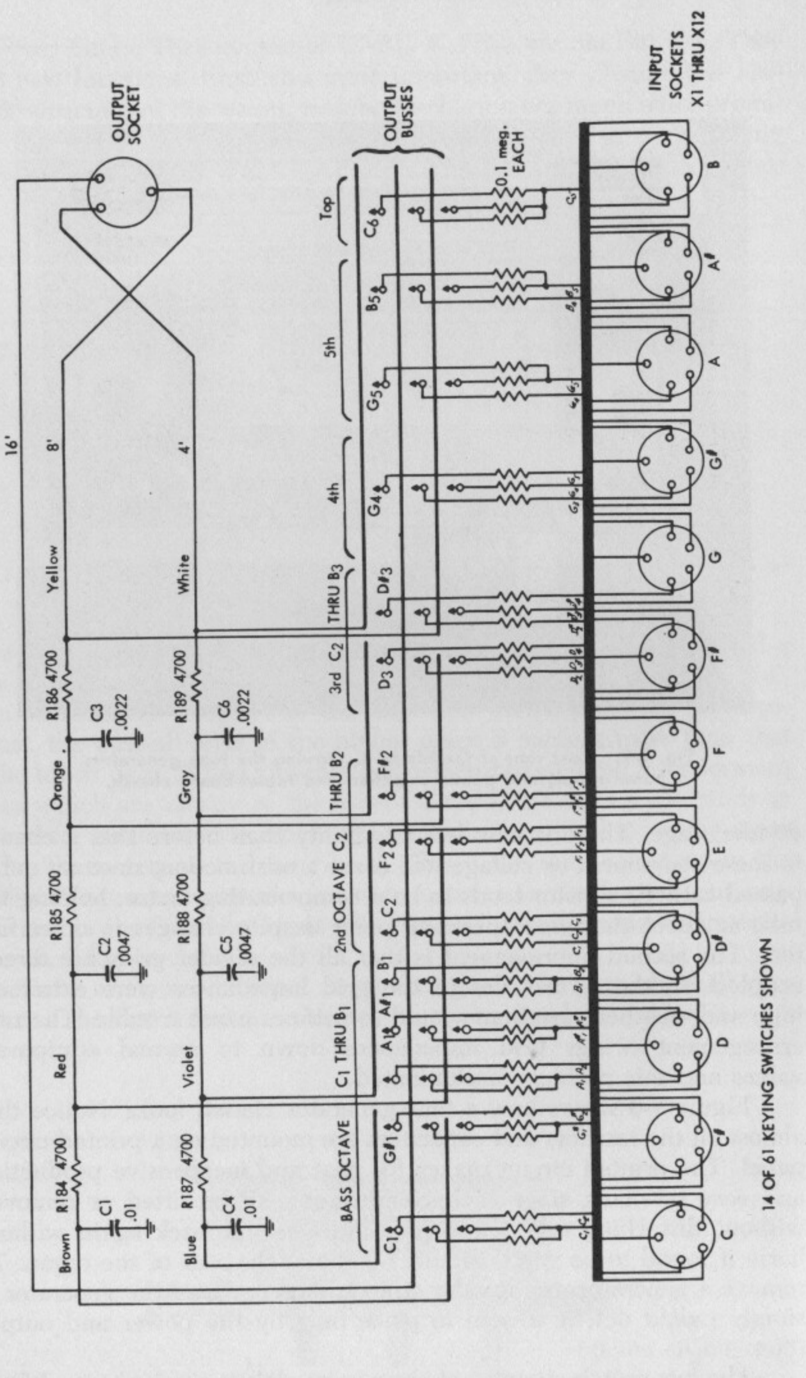


Fig. 7-10. Key-switching diagram for the swell manual. Great wiring is the same, with addition of plugs and cables to carry tone up to swell.

dances of the old generator outputs made shunt keying mandatory; that is, switches were normally closed, grounding undesired tones. Not only did this require more components and more labor, increasing the cost, but a switch with nonfunctioning contacts would cause a "cipher," a continuous sounding of the tone. Inevitably any switch will once in a while fail to work, but it is far less disturbing to have a failure of a tone than to have it sound continuously.

The new system, drawn in part in Fig. 7-10, employs the series keying method, with normally open switches. Each tone is brought to the appropriate switch through a 0.1 megohm isolating resistor. When a key is played the switch closes, passing the tone to output busses which run the length of the assembly. The switches are actually three-circuit ones, so that when a key is played tones of 4-, 8-, and 16-foot pitches are brought to the respective busses. The resistor-capacitor

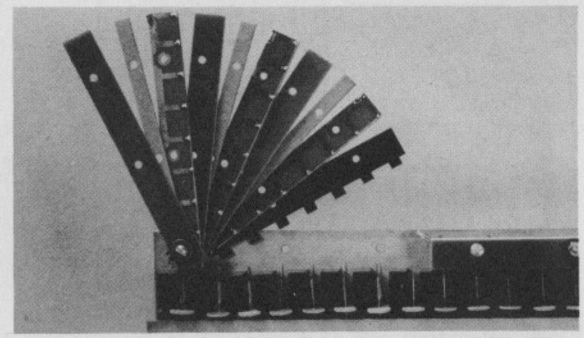


Fig. 7-11. One end of a key-switch assembly, showing the switch blocks, printed-circuit ouput busses, and separators.

networks to which the busses are connected are key-click filters, tailored in values to the specific ranges which they cover, so that clicks are almost eliminated but minimum harmonic structure of the tone is affected.

The key switches themselves are no longer the flat blade type. When used for series switching these units have too large a capacitance between the opened blades, and there is leakage of unkeyed tone into the rest of the system; the result is an annoying whine in the background. Part of a key-switch assembly, opened and fanned for view, appears in Fig. 7-11. The switches themselves are blocks of phenolic in which three silver-alloy fingers are set. A small phenolic actuator is placed over the fingers and when the key comes down it hits the actuator, which forces the fingers down. Each finger strikes a gold bus wire running at right angles to it. Because the entire switch consists only of two thin crossed wires capacitance across an open switch is negligible and there is no audible leakage whatever. The combination of materials results in trouble-free contacting over a long period.

PEDAL TONE GENERATOR

The main tone generators provide only five octaves of tones equivalent to the 8-foot range of the standard 61-note manuals. The 4-foot tones for the upper manual octave are repeats of the upper-octave 8-foot tones and the lower-octave 16-foot tones are repeats of the 8-foot lower-octave tones. In the earlier models 16-foot pedal tone was derived from a "resultant bass" arrangement which mixed the lowest 8-foot tone with its musical fifth and produced a beat note an octave below the fundamental. The actual 16-foot component of this system was small and the idea unsatisfactory from every angle but cost.

In the new models actual 16-foot tone is derived from a special pedal generator which requires only two tubes for the entire pedal section. It is actually not a generator at all but a wide-range frequency divider.

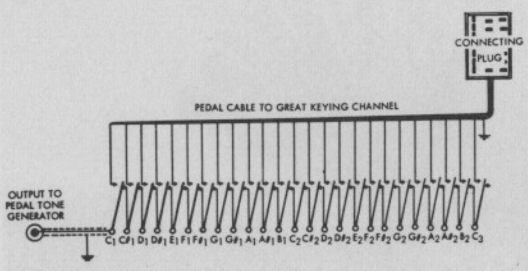


Fig. 7-12. Diagram of the pedal-switch assembly. Tones from the main generators are fed through the large connector to the switches.

Figure 7-12 is a schematic diagram of the pedal keying switches. Each switch is a single-pole double-throw unit and the diagram shows the switches in normal (unkeyed) position. Eight-foot tone from the main generators is fed through the connector and cable to each of the switch contacts as shown. When any pedal is pushed, its switch changes positions and the corresponding 8-foot note is brought to the output. Because of the switching arrangement, only one tone at a time can appear at the output, the lowest of any number played simultaneously.

The pedal generator is shown schematically in Fig. 7-13. The 8-foot tone from the main generators through the pedal switches goes to the first grid of a 12AX7 amplifier and from its plate to the second half of the tube which is a wave shaper giving the tone the proper shape to trigger the following flip-flop circuit. The second 12AX7 is an aperiodic flip-flop which is nonfrequency-sensitive over a wide range. The 16-foot output is taken from one plate circuit so that for

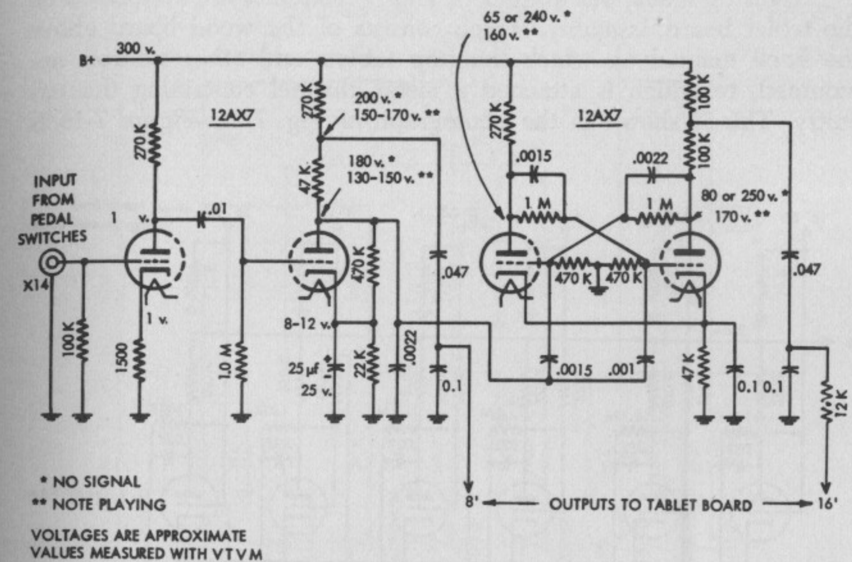


Fig. 7-13. The pedal generator, consisting of amplifier, wave shaper, and flip-flop.

every two input cycles there is one output cycle. The 8-foot output is taken from the second plate circuit of the first tube. Both are fed to the tone-color section. This is an extremely neat, inexpensive, and effective method of deriving real 16-foot tone.

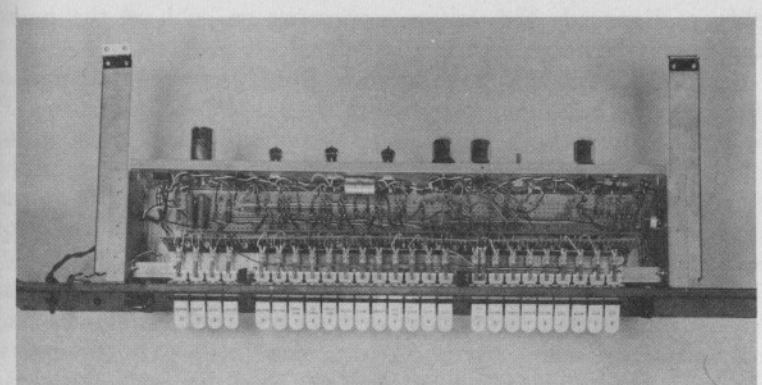


Fig. 7-14. The tablet-board holds the tab switches and the chassis containing bus amplifiers, tone filters, and preamplifiers.

TABLET BOARD CIRCUITRY

The stop filters, bus amplifiers, and preamplifier are all located on the tablet board assembly, which consists of the wood board above the swell manual on which the stop tablets and other controls are mounted, to which is attached a metal channel containing the circuitry. This is shown in the photograph of Fig. 7-14. Figure 7-15 is

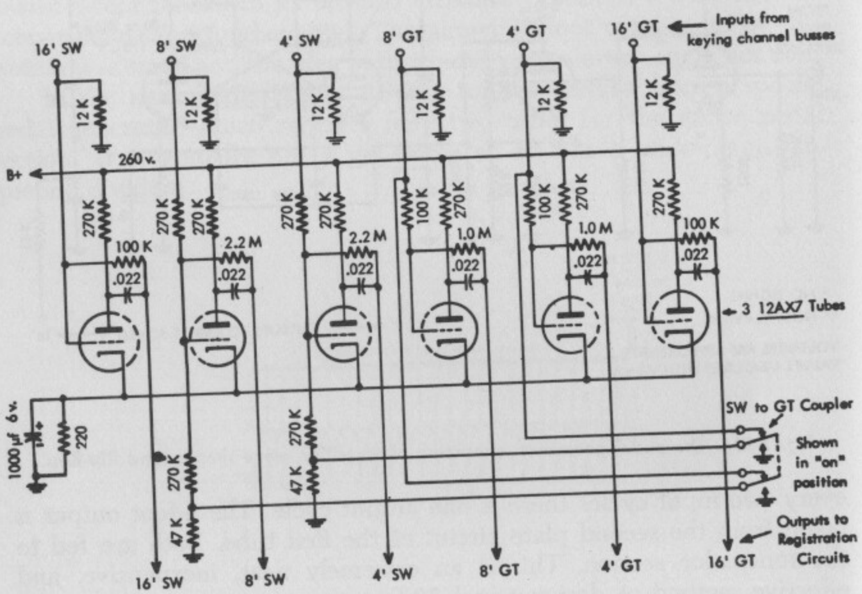


Fig. 7-15. Bus amplifiers and coupler system.

the schematic diagram of the bus amplifiers, whose function is to amplify preliminarily the voltage appearing on each of the manual keying output busses before it is applied to tone filters.

Each of the triode voltage amplifier grids is fed signal from one keying bus, the bus being terminated by a 12,000-ohm resistor. The 16-foot tones from both manuals go through to the grids without isolating resistors, but the others have resistors between bus and grid. Each triode has voltage feedback, a capacitor and resistor from plate to grid; the purpose of this is to give an effectively low output impedance.

There is one coupler on the organ, a Swell to Great. This means that when the coupler switch is closed, as it is in the diagram, 8-foot and 4-foot tones keyed on the great manual will pass through the 8-foot and 4-foot filters associated with the swell manual. This means, of course, that they must be mixed into the swell busses. Note how the 4-foot great tones are handled for this purpose. Tone from the keying

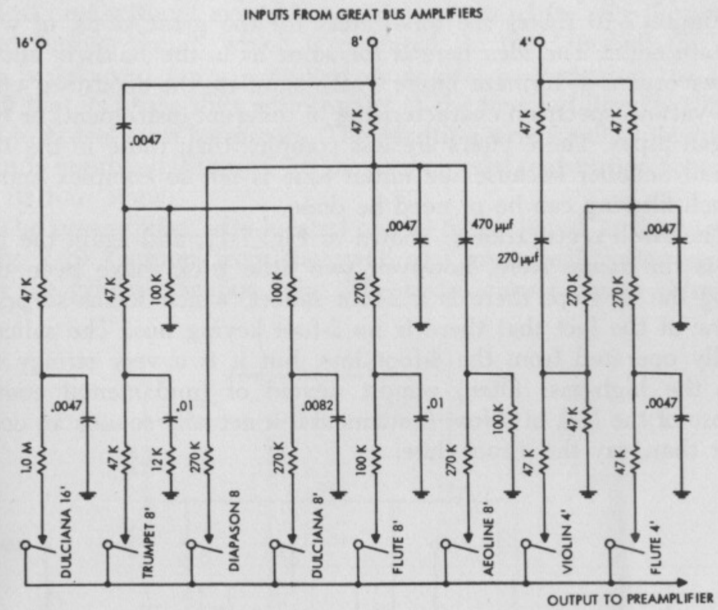


Fig. 7-16. Tone filters and tab switches for the great.

bus is fed to its tube through a resistor and to the coupler switch which, when closed, injects 4′ great tone into the grid circuit of the 4′ swell amplifier tube. An important point in this process is that the 4-foot swell output from its amplifier and the 4-foot great output from its amplifier must not change in level with operation of the coupler switch. Effects on the 4-foot great amplifier are prevented by taking the coupling line directly from the bus ahead of the isolation resistor and making sure that when the switch is closed this point is shunted by nothing which would be comparable to 12,000 ohms.

Preventing some effect on the 4-foot swell tone is not so easy, but it is done here in a very neat way. The output of the 4-foot swell amplifier depends on the magnitude of the feedback. This depends on the total value of resistance between grid and ground, since this resistance is the shunt leg of a voltage divider for feedback. The value of the 4-foot swell voltage reaching its grid is also dependent on the total value of the series resistors since these are the shunt leg of a signal voltage divider. When the coupler switch is closed, the 4-foot swell voltage at its grid is decreased because of the lessening of the total resistance between the grid and ground. However, this decreases the feedback voltage fed around the 4-foot swell tube, so that its gain rises just enough to restore the former output level, now with the addition of the 4-foot great tone. The same process applies to the 8-foot coupling.

Figure 7-16 shows the tone filters for the great stops, of which there are eight. The idea here is the same as in the Baldwin and the Schober organs — formant filters which simulate the acoustical effects of the various spectrum characteristics of different instruments or types of organ pipes. These filters are less complex than those in the Baldwin and Schober because the initial tone is not so complex and not as much filtering can be or need be done.

The swell registration is shown in Fig. 7-17, and again the principle is the same. Here, however, two little tricks have been used. Among the 12 stops there is a 2-foot salicet, which seems surprising in view of the fact that there is no 2-foot keying bus. The salicet is actually operated from the 4-foot line, but it is a very stringy tone (note the high-pass filter) almost devoid of fundamental content. Because of the lack of 4-foot fundamental it actually sounds an octave higher than, say the 4-foot flute.

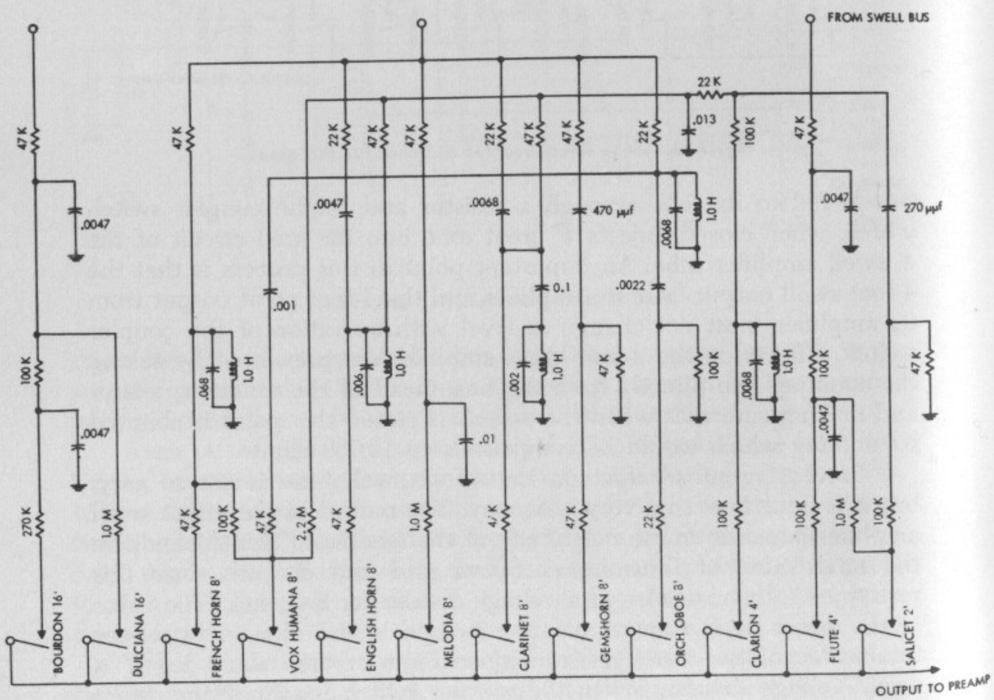


Fig. 7-17. The swell registration circuit.

The second trick is in the clarinet filter. The normal clarinet tone is almost devoid of even harmonics. In the Baldwin organ the symmetrical tone without even harmonics is obtained for this purpose by an outphasing circuit, which is patented. In the Minshall the same effect is achieved (over a limited range) by so designing the tuned circuit that its phase shift with respect to the tone fed directly through tends to cancel even harmonics. The resulting tone is not quite authentic but is improved. Figure 7-18 shows the pedal registration schematic with its four stops.

The preamplifier, also located on the tablet board, is diagrammed in Fig. 7-19. Outputs from the swell and great registration sections go to the grid of the first tube. The pedal output passes through a

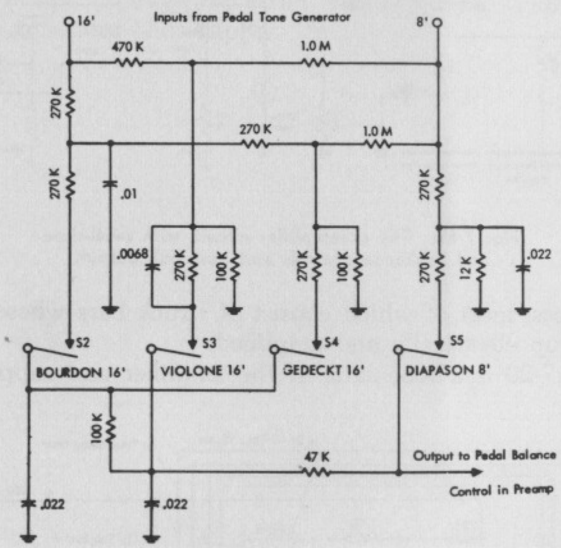


Fig. 7-18. Circuits of the four pedal tone filters.

"balancing" section so that the pedal level can be adjusted for any set of auditorium conditions with respect to the manual levels. This first triode has feedback around it. Volume of the organ is controlled between the first two stages by a swell-shoe control which varies the impedance of the shunt leg by a voltage divider, being compensated for loudness by the capacitor network which raises the comparative level of the bass as volume decreases. A brilliance control in the plate circuit of the second stage is simply an old-fashioned tone control. The third stage is a cathode-coupled phase splitter and the preamplifier output stage is push-pull with feedback around each half. The "chimes" input is to be used with any of the commercial electronic

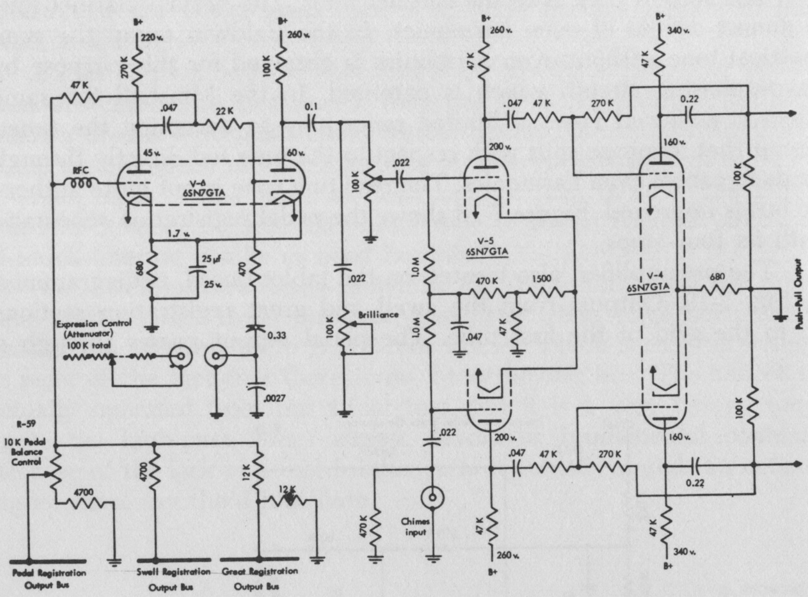


Fig. 7-19. The preamplifier circuit, with swell-shoe and brilliance controls and push-pull output.

chime devices, most of which consist of struck bars whose vibrations are picked up electrically and amplified.

Figure 7-20 is a schematic of the amplifier and supply sections.

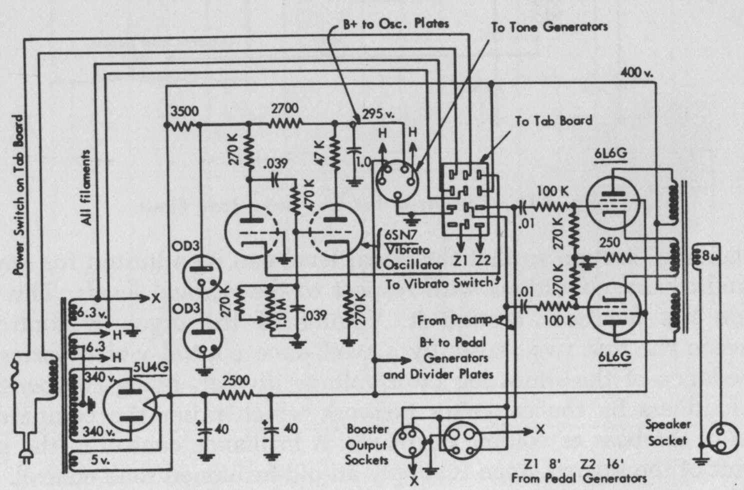


Fig. 7-20. Schematic diagram of the power amplifier and power supply with vibrato oscillator.

The amplifier, rated at 15 watts output, consists of a pair of 6L6's with cathode feedback from the transformer secondary. The power supply is standard, furnishing B-plus and filament voltage to generators and tablet board through connectors as shown. The tablet-board connector also carries preamplifier output to the power amplifier grids and to a pair of sockets into which lines to booster amplifiers may be plugged.

The vibrato oscillator, a 6SN7, is also in the power supply. This is a simple feedback oscillator the frequency of which can be adjusted over a range of about 5 to 8 cps by switching different resistors across R_9. This is done at the tablet board and a line from the cathode for this purpose is carried up through the large connector. Regulated B-plus is applied to the vibrato-oscillator plates (note the VR tubes) and the "vibrated" voltage is carried to the master oscillator plates through the generator plate supply.

Chapter 8

The Conn Electronic Organ

C. G. Conn Ltd. of Elkhart, Ind., perhaps justifies the slight aura of pretentiousness the "Ltd." gives its name by its many decades of outstanding work in the production of band instruments. For a fewer number of years it has also been in the electronic organ business producing a line of instruments called Connsonatas. For a reason which escapes this writer, it has recently abandoned the pleasant and rather clever elision of "Conn," "consonant," and "sonata" which resulted in the name of the Connsonata, and the new line of instruments comprises three principal models of what is known simply as the Conn Organ. The loss of euphony on the nameplate, however, is more than made up by the gain in nearly every musical and electronic element consonant with organ design; the new models make beautiful music and may be had at prices well in line with industry norms despite the inherently greater expense of the design approach.

The most elaborate of the new models is the Classic or 800 Series pictured in Fig. 8-1. This one has three complete sets of generators for Great, Swell, and Pedal, 19 stops, 14 couplers, and a 32-note pedal clavier. The Minuet or Series 500 model is shown in Fig. 8-3. It is a spinet along the lines of the current trend, with two 44-note offset manuals, 13 stops, 7 couplers, and a 13-note pedal clavier. Even this small model has independent generators for the two manuals and a monophonic generator for the pedals, a welcome relief from the discontinued models of not very long ago which used a single generator set for everything.

The middle unit, which is described in detail herewith, is the Artist or 700 Series organ which appears in Fig. 8-2. The Artist model, which has two full manuals, 25 pedal notes, 16 voices, and 9 couplers, is very similar in function to the others. It has two complete sets of tone generators for the manuals and a monophonic generator for the pedals.

Fig. 8-1. The Conn "Classic" model, Series 800.

DESIGN PRINCIPLES

There are two principal schools of thought concerning organ design. One, typified by Hammond, says that there should be separate generators for the notes and that tone colors should be obtained by harmonic synthesis — using the generated tones above the fundamental in use at the moment as harmonics of the fundamental to build up harmonic structures as desired. The other, of which Baldwin is presently the major exponent, employs locked frequency dividers to produce tones rich in harmonics, then passes them through filters which remove the harmonics not wanted to achieve a particular tone color. Both have certain advantages, although there is no doubt that the latter system, known as the formant principle, yields tone colors which are much more closely imitative of acoustical instruments like the pipe organ and others.

The Conn Organ, like the Allen, employs a combination of the two principles (with emphasis on formants), but is much simpler schematically and physically than the Allen while achieving much the same results. It contains a separate oscillator for every note (and for each manual); every oscillator yields not only a sine wave but also a sharply pulsed wave which is passed through formant circuits. In addition, the inter- and intra-manual couplers operate like those on a unified pipe organ to give octaves and mutations which may to a certain extent be used like the harmonics obtained with Hammond's drawbars. The actual number of formant voices and couplers on the Conn does not represent the maximum available with this company's design approach any more than the actual specification of any of the major organs represents the maximums available with their own peculiar approaches. Specifications are the result of educated guesses which attempt to balance musical versatility with cost in an effort to purvey the largest possible number of organs to the largest possible number of as satisfied as possible customers.

Figure 8-4 is a block diagram showing the principal functional components of the Conn Artist model. Seventy-three oscillators are available for the swell manual, comprising notes 13 through 85. The note numbering system is based on the unison pitches for a 5-octave manual, with a 16-foot octave below and an 8-foot octave above. Thus the unison manual notes are 13 through 73 (remembering that there is an extra top C on the manual), with the 4-foot octave 73 through 85. There is no 16-foot octave 1 through 12.

The swell oscillators are keyed by special under-key switches, of

Fig. 8-2. The Conn "Artist" model, Series 700, is the subject of the technical description.

Fig. 8-3. The "Minuet" model, Series 500 is designed for the smaller home.

which more later, which control plate voltage, and also by couplers, which control sounding of the note at unison pitch and/or in other registers or with mutations. Oscillator outputs are sent to an output bus which passes the tones to so-called mixer circuits. These are actually triode amplifiers equipped with filters for the formant circuits and attenuators for the flute tones, with the filters and attenuators controlled by stop-tab switches. Outputs of the mixers are summed and led to a preamplifier containing a swell-shoe circuit, thence to an output connector from which a cable leads tone to the amplifiers mounted in the speaker units (or tone cabinets, as most makers call them).

The setup is the same for the Great except that the notes generated are 13 through 73. The pedal circuitry employs a solo-type generator consisting of only two actual oscillators. Each is used for one of the two octaves covered in the pedals and has variable frequency so that only one note can be played at a time in each octave.

The tone generators are individual oscillators of the grounded-plate Hartley type, as diagrammed in Fig. 8-5, each using half of a 12AU7. The cathode is grounded through low-value summing resistors in the string-bus circuit; R_3 and R_4 are merely scaling resistors whose

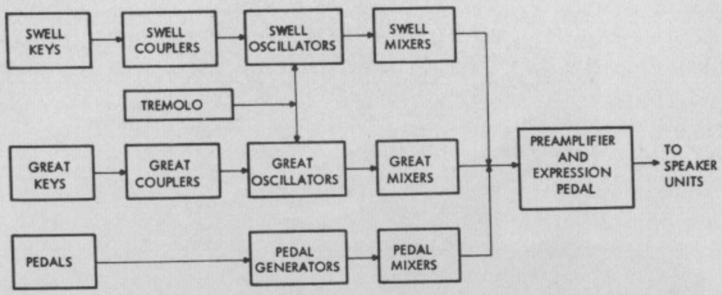


Fig. 8-4. Simplified block diagram of the Series 700 organ.

values vary — R_3 from note to note and R_4 from octave to octave. The tone appearing at the cathode tap of the coil is nearly sine in shape and it is used for flute-type stops; that appearing between the lower terminal of the tuning coil and ground (across the terminating resistor shown in a later circuit) is sharply pulsed and is called string tone. Each oscillator is keyed by applying 75 volts positive to the plate through a 100,000-ohm time-constant resistor R_2 which operates in conjunction with time-constant and bypass capacitor C_3. Limit values are shown in Fig. 8-5 for those which vary. The note is tuned by varying the position of the iron core in L with a screw and spring arrangement.

Twelve notes — one octave — are mounted on each chassis. Figure 8-6 shows the rear of the organ with all the electrical components except the key switches in view. At the lower right are the first

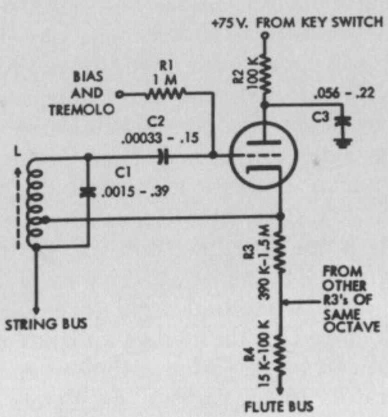


Fig. 8-5. Schematic of one tone generator; a separate generator is used for each note of each manual in the organ.

four Great generator chassis, with the fifth left of the swell shoe. Above are the six Swell chassis.

PREAMPLIFIERS AND TREMOLO

The chassis to the left of the great generators in Fig. 8-6 holds the pedal generators, flute and string preamplifiers for the great division, and the great tremolo generator.

Figure 8-7 shows the division preamplifiers. String tone from the string bus shown in Fig. 8-5 is fed to a 27-ohm resistor and a 100-ohm

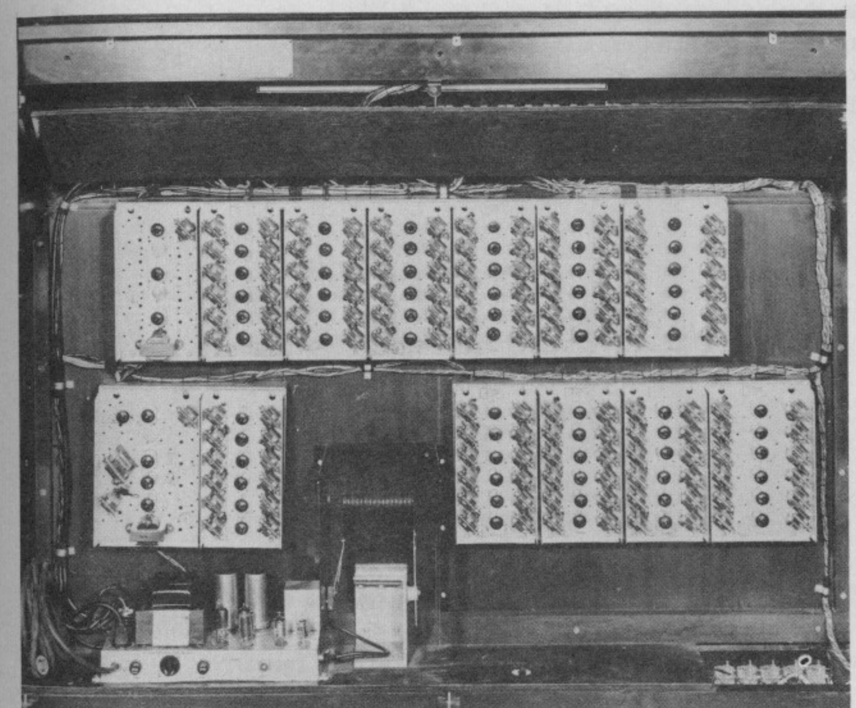


Fig. 8-6. Rear view of the organ to show the placement of the tone generator chassis — twelve notes being accommodated on each.

potentiometer in parallel. The potentiometer adjusts the over-all string level by shunting the resistor. This resistance is what completes the cathode-ground circuit for d.c. in Fig. 8-5. The string tone is low in level and is fed through two cascade amplifier stages, thence through a shielded line to the voicing or mixer circuits. Flute tone is fed directly to the grid of a single amplifier stage, thence from the plate to the mixers. No blocking capacitor is necessary between oscillator cathode resistors and preamplifier grid since the cathode d.c. voltage is very low due to the shunt of less than 27 ohms to ground afforded by the string-bus circuit. A similar set of preamplifiers appears on the chassis to the left of the swell generators in Fig. 8-6 and is used for the swell flute and string buses.

The great vibrato generator, shown in Fig. 8-8, employs a 12AU7. The two triodes are cross-coupled by the 1-megohm resistors and 0.1$\mu\mu$f capacitors between the plate of each and the grid of the other, so that positive feedback is produced around a loop composed of the two triodes. A tuned transformer is connected between the plates, and this smoothes out the waveform. The secondary feeds vibrato signal to a series resistor network to which the bias-input R_1 of each tone oscillator of Fig. 8-5 is connected. This apportions the vibrato signal so that each oscillator "vibrates" the optimum amount for best musical results despite differences in tuned-circuit Q's due to differences in component values and operating frequencies. Since neither end of the transformer secondary is grounded, the tone oscillator grid-ground

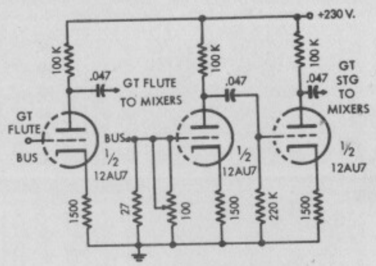


Fig. 8-7. Schematic of the Great bus preamplifiers.

circuit for vibrato is completed by the string-bus connection shown. The network resistors, in conjunction with the 56,000- and 2700-ohm resistors coming from the 75-volt line, divide the B-voltage properly to afford some positive bias for the generator tubes.

Both speed and amplitude of tremolo (actually it is vibrato or frequency swing) are controlled by the three tab switches shown in the cathode circuit. Tremolo generators for Great and Swell are identical (there is none for the pedal as pedal vibrato is undesirable) though the tone-generator resistor networks are somewhat different in detail. Lines from the paired cathodes of each generator go to the tab switches as shown in Fig. 8-8. When the TREMOLO F (meaning full) switch is closed both sets of cathodes are grounded through 2200-ohm resistors. With the TREMOLO M (medium) switch closed, another 10,000 ohms of resistance is placed between cathodes and ground, lowering the plate current and reducing the tremolo-voltage output of the transformer secondary. When TREMOLO L (low) is closed another 4700 ohms in the cathode circuits reduces tremolo still further. With all switches open over 1 megohm is in the circuit, effectively causing

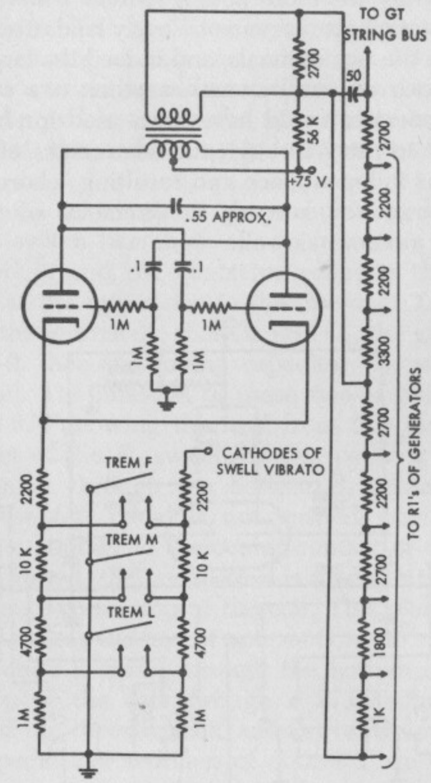


Fig. 8-8. Schematic of the Great tremolo circuit. A similar unit is required for the Swell manual.

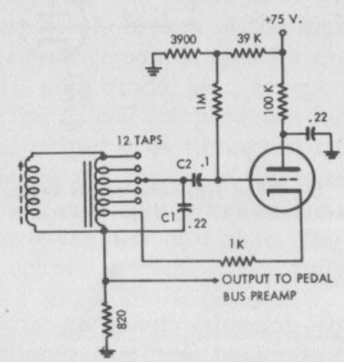


Fig. 8-9. The pedal generator is monophonic and requires only two triode sections for 25 notes.

oscillation to stop. With more than one switch closed, the tremolo will be that given by the more potent switch.

This tremolo or vibrato system obviously lends itself well to separate use of tremolo on the two manuals, and indeed the larger Classic model has GREAT TREMOLO OFF and SWELL TREMOLO OFF switches. While a single tremolo generator could have been used for both manuals, the designers wished to carry through the advantages of separate oscillator ranks as far as independence and resulting "chorus" or "ensemble" effects are concerned by keeping the tremolos of the two manuals independent and random as well.

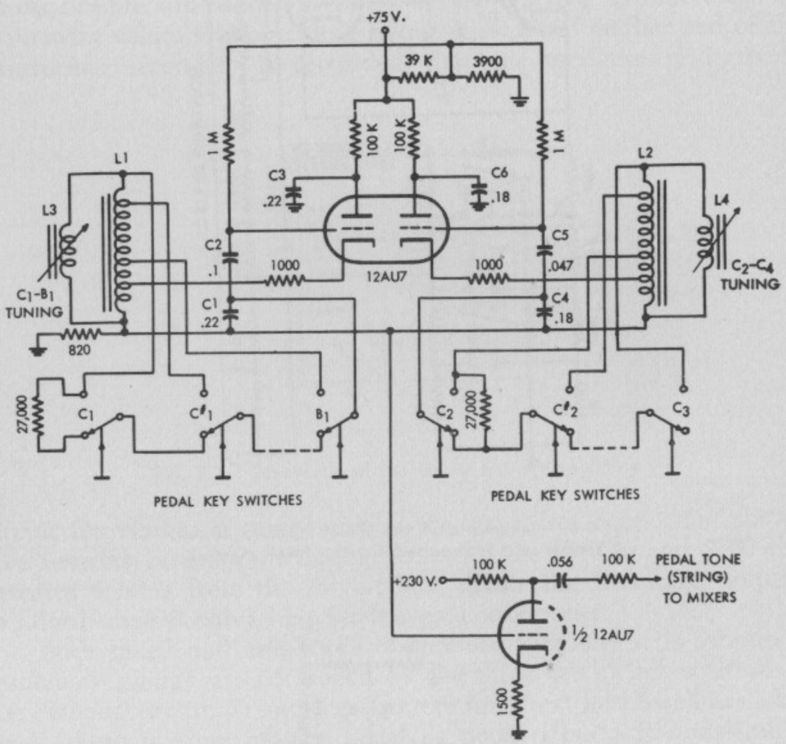


Fig. 8-10. Simplified schematic of the keying and the connections of the pedal generator.

PEDAL GENERATORS

The complete pedal generator circuit appears in Fig. 8-10 but Fig. 8-9, which should be examined first, is a simplification showing the oscillator itself. It is a grounded-plate Hartley, and like the manual generators, a string tone is taken from across a resistor in series with

the tank circuit. No flute tone is taken from the pedal division. A positive bias is placed on the grid with the aid of a voltage divider and isolating resistor. The oscillator is tuned over the 12 or 13-note range required by using an inductor with taps. In parallel with the tapped inductor is a trimming inductor which has variable inductance. In effect, using the pedals causes the appropriate tap to be used for each note so that the remaining inductance tunes the oscillator correctly.

Figure 8-10 is a schematic of the complete pedal division. Each pedal operates a switch like those shown in such a manner that the plunger is pushed in and the contactor connects the center contact with that shown as the upper one in the drawing. Tracing the circuit from the grid of the left triode, there is first C_2, the grid-leak capacitor shown in Fig. 8-9, then the tuning capacitor C_1, which goes to the bottom of the coil. The junction of these two is switched, as can be seen from Fig. 8-9. Following the lead from this junction, it goes to the center contact of the B_1 switch. If this pedal is not pressed, the connection continues through the bottom B_1 contact to the center $C\sharp_1$ contact. If the $C\sharp_1$ pedal is not pushed, connection continues through its bottom contact to the center contact of C_1. If we assume that pedal C_1 is pushed, the connection is made to the top contact of the C_1 switch, thence to the top of the coil. This places the entire coil in the circuit and causes the lowest note — C_1 — to sound. If no pedal is pushed, connection is made through the bottom contact of the C_1 switch to the top of the coil through a 27,000-ohm resistor which destroys the Q of the tuned circuit and prevents any oscillation.

Only three pedal key switches of octave 1 are shown, but there are, of course, twelve, all connected in the same way, with the top contact of each going to a tap on the coil. Therefore, when any pedal is pushed, the connection from the junction of C_1 and C_2 is made to the appropriate coil tap and the right note is heard. If two pedals are pushed simultaneously, only the higher note will be heard since the line from switch to switch is broken at the higher note point.

Exactly the same arrangement is used for the second pedal octave with separate coil L_2 and triode, etc., except that since the pedal clavier has 25 notes, the L_2 coil has provision for 13 notes rather than 12. The pulsed pedal output goes through a triode preamplifier like the manual output buses and thence to the mixers.

Up to now we have said simply that each note is keyed by applying a positive 75 volts to the oscillator plate. The actual conditions are complicated by the coupler system, whose function is to play more than one note when a single key is pressed.

When all couplers are in off position, pushing the C_3 key will supply plate voltage only to the C_3 generator and that will be the only one to sound. When playing on the Swell and the 4′ rocker tablet at the left of the manual is pressed, both C_3 and C_4 will sound. With the

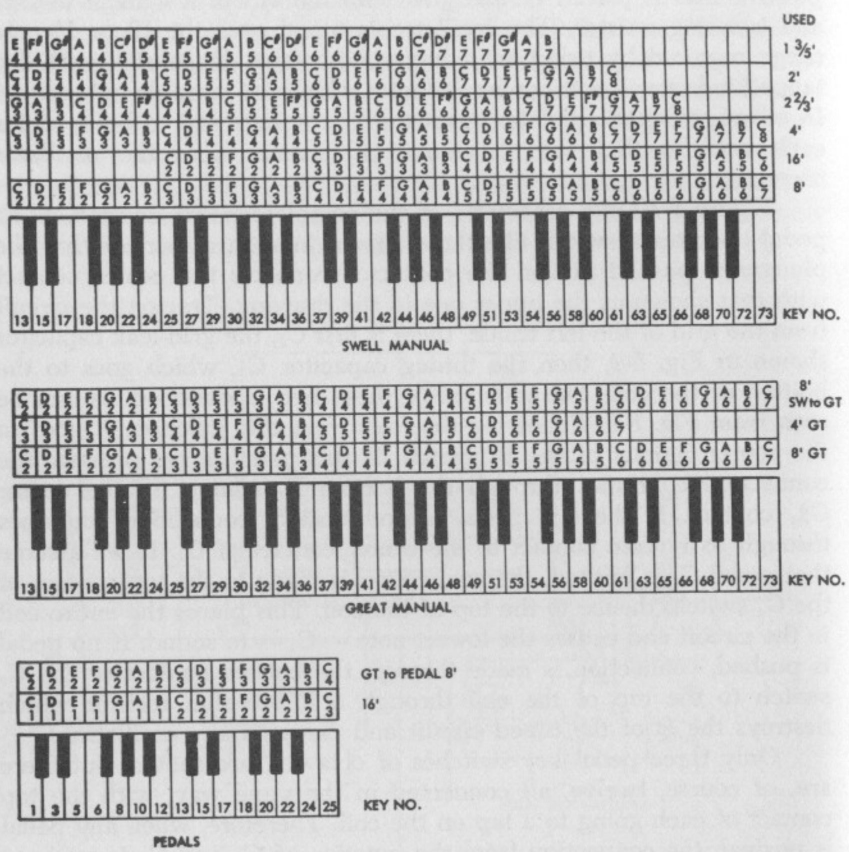


Fig. 8-11. Interconnections of the coupler circuits in the three sections, simplified by labeling only the white keys.

2⅔′ tab pressed, G_4 will sound. (This is approximately the third harmonic of C_3, equivalent to a 2⅔′ rank on a pipe organ.) This process continues with the remainder of the coupler tablets in the manner shown in the keying chart of Fig. 8-11.

Obviously, then, the keying circuits must provide for as many switches per key as there are registers and couplers, and for a method of making registers speak or remain silent at will. The mechanics of the system used by Conn are new and are shown to some extent by Fig. 8-12. In this photo, the manual in position is the Great; note the two coupler tabs, SWELL TO GREAT 8′ and GREAT TO GREAT 4′ at its left. Above this, the Swell manual has been swung up to show the key switches on its underside; note that the white undersides of the swell

keys can just be seen at the top.

There are six registers in the Swell, and for each there is one switch wire or finger. All fingers are molded in a lower plastic dowel (or rear dowel, as it would be with the manual in playing position). The fingers are also held in a second dowel which moves downward when a key is pressed (they move outward toward the reader with the manual up-ended in the photos). The movement of the dowel bends the fingers and causes the free end of each to contact one of the six keying rods which run lengthwise of the manual. These rods carry +75 volts. Since each finger is connected to an oscillator plate, the contact keys the oscillators.

Let us move at this point to Fig. 8-13 and look at the drawing of the Swell keyboard switches. The mechanism here is shown in playing position. A key pressing down on the actuator causes it to move downward as indicated by the arrow. It causes all the keying fingers to move down so that each finger touches the rod beneath it which is running at right angles to the fingers (through the page as viewed). Notice the peculiar cross-section shapes of the keying rods. Each consists of a metal extruded or drawn bar in a D shape with a projection from the straight part of the D. The thick black outline around

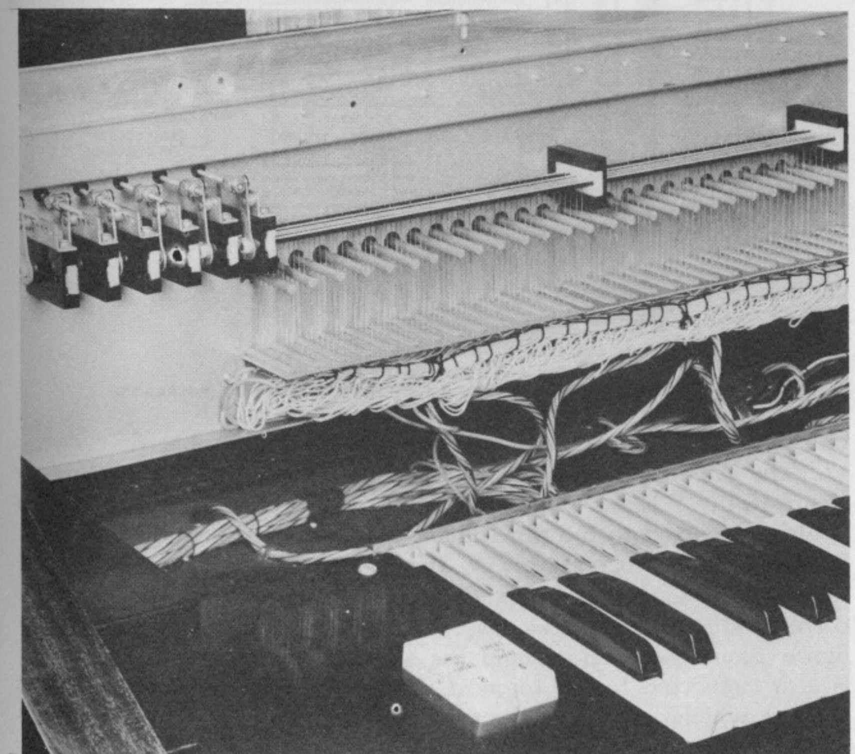


Fig. 8-12. Swell manual hitched up to show keying springs and coupler operation.

everything but the projection represents an insulating sleeve. When the bare projection points up, a finger hitting it makes electrical contact. When the bar is rotated so that the projection points to the right, the finger strikes the insulating jacket and no contact is made. The bars are rotated by the rocker coupler tablets at the left of the manual, so that the position of a tablet determines whether its bar shall or shall not contact its row of fingers. The bar rotating mechanisms, six of them for the Swell, are plainly shown at the upper left in Fig. 8-12.

On the Swell manual the second finger from the top under every key is connected to the plate of the oscillator which generates the unison note of that key; for instance the second finger from the top under the E_4 key is connected to the E_4 generator. This finger is therefore the 8-foot or unison finger and the rod which it contacts is the 8′ rod. Whenever the 8′ tab is positioned so that the bare projection

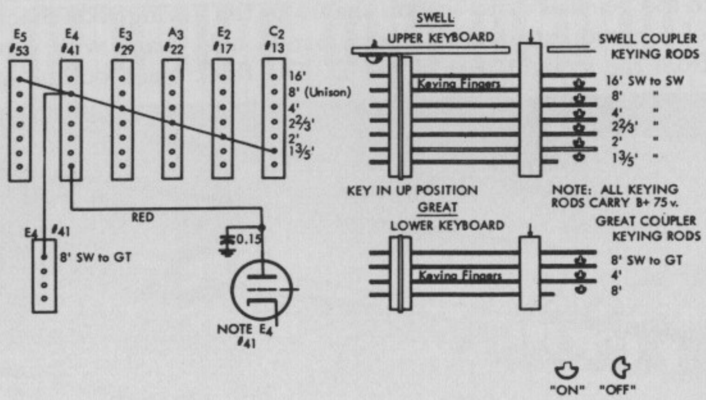


Fig. 8-13. Diagram of coupler action applied to keying mechanism.

on that rod points up, the 8′ note of every key will be heard.

The same E_4 is also the 4′ tone for key E_3; since, according to Fig. 8-13 the 4′ finger is the third from the top, the E_4 oscillator plate is also connected to the third finger from the top under the E_3 key. It is also the $2\frac{2}{3}'$ tone for key A_3, the 2′ tone of key E_2, and the 1-3/5′ tone of key C_2, as well as the 16′ tone for E_5, so it is connected to the corresponding finger under each of these keys, as indicated by the figure. Since the great manual has a SWELL TO GREAT 8′ coupler, E_4 is also connected to the topmost spring under key E_4 of the Great manual, which operates in the same way but has only three couplings including its own unison.

The keying chart shows in detail all of these connections; they

are made by the wiring harness which can be seen under the finger assembly in Fig. 8-12. It should be noted that the 100,000-ohm time-constant and isolating resistor shown in Fig. 8-5 in the oscillator plate circuit is actually located on each finger strip, but has been omitted in Fig. 8-13 to avoid confusion. This assembly is ingenious and effective, and takes the place of solenoid-operated devices which have for many years added to the cost and complexity of pipe organs which employ this kind of coupling as well as earlier organs made by Conn.

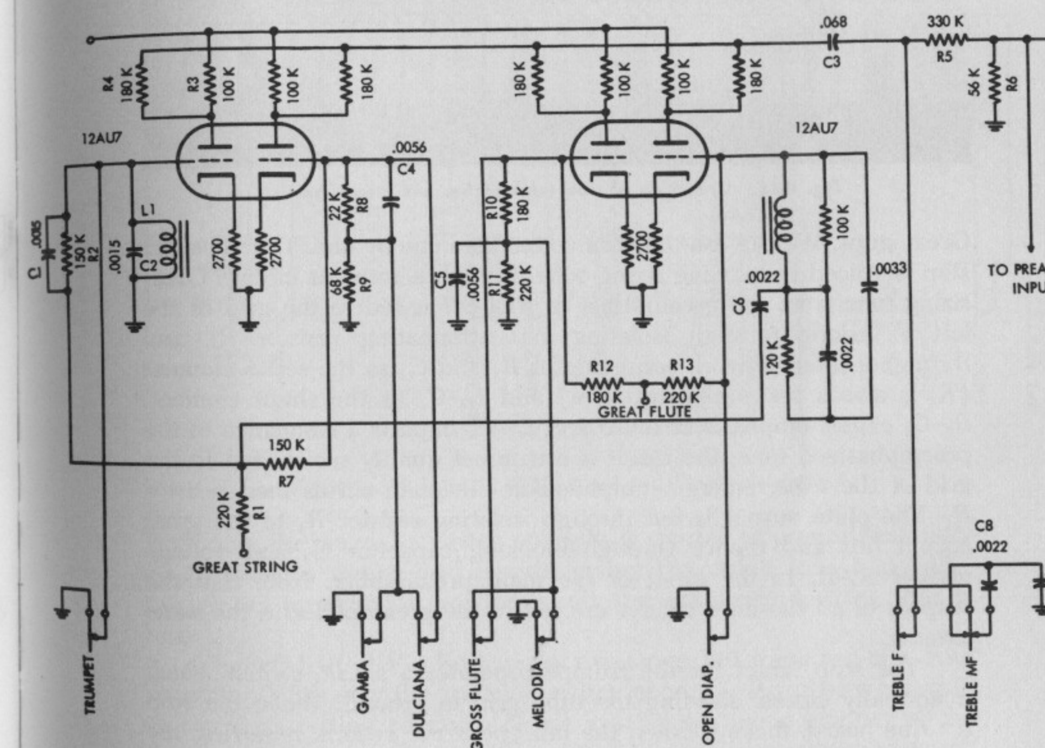


Fig. 8-14. Schematic of the Great mixer network and filters.

MIXER CIRCUITS

The mixer circuits of the Conn Organ are filters and filter amplifiers which modify the spectrum characteristics of the passband and impart formants to the pulse signals. In addition, they operate both as formants and as level controls for the flute signals, which are not quite sine waves and thus contain harmonics. Lastly, they do, for certain stops, mix quantities of both flute and string or pulse tone.

Figure 8-14 is a complete schematic of the mixer circuits for the

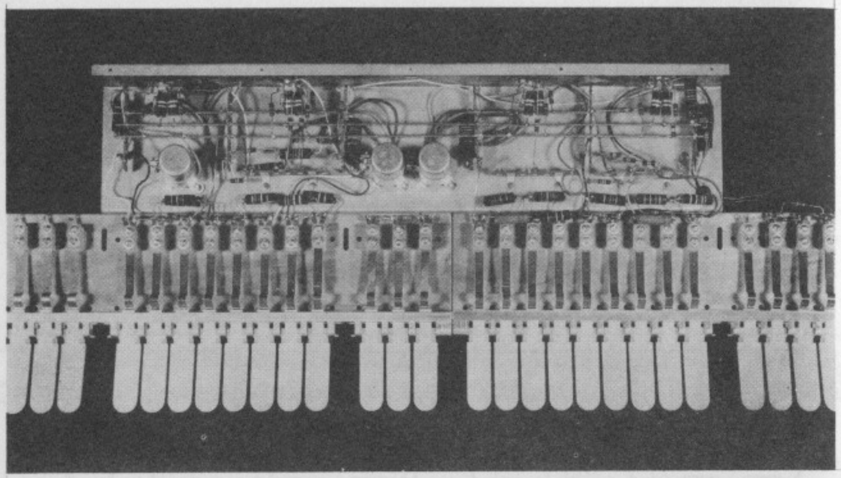


Fig. 8-15. Underside of stop tab switches and mixer chassis.

Great stops. We can examine the voices here one by one. The TRUMPET stop is voiced by passing string tone through a formant circuit. Great string tone from the preamplifier of Fig. 8-7 is fed to the grid of the left V_1 triode through isolating and attenuating resistor R_1, and through a filter network consisting of R_2 and C_1 as the series element (R_1 is also a series element here) and L_1–C_2 as the shunt element. R_2–C_1 causes emphasis of highs and L_1–C_2 imparts a resonance to the preemphasized tone; the result is a trumpet quality which, fed to the grid of the tube, emerges amplified at the plate across load resistor R_3. The plate output is fed through isolating resistor R_4 to the great output bus and thence through blocking capacitor C_3 and voltage divider R_5–R_6 to the input of the main preamplifier. Note that the outputs of all the filter triodes are fed to the preamplifier in the same manner.

The stop tablet for the trumpet operates a s.p.s.t. switch which is normally closed, shorting the tube grid to ground. When the stop is to be heard, flicking down the tab opens the switch, removing the short. The tabs, switches, and some other components are seen in the photo of Fig. 8-15, which shows the underside of the mixer assembly.

The second triode of V_1 handles two stops, GAMBA and DULCIANA. Great String tone is fed through both R_1 and R_7 to the grid, these two acting as the series leg of a voltage divider, with R_8–C_4–R_9 as the shunt leg. With the GAMBA and DULCIANA switches on (open), the tone at the grid has a slight rolloff due to the shunting effect of C_4, but not a great deal because of R_9 in series with the C_4–R_8 combination. With both switches closed, the grid is shorted to ground and neither stop is heard. With the GAMBA switch open (on) the C_4–R_8 combination is shorted and only R_9 appears between grid and ground. The attenuated pulse tone goes to grid, giving a very stringy quality.

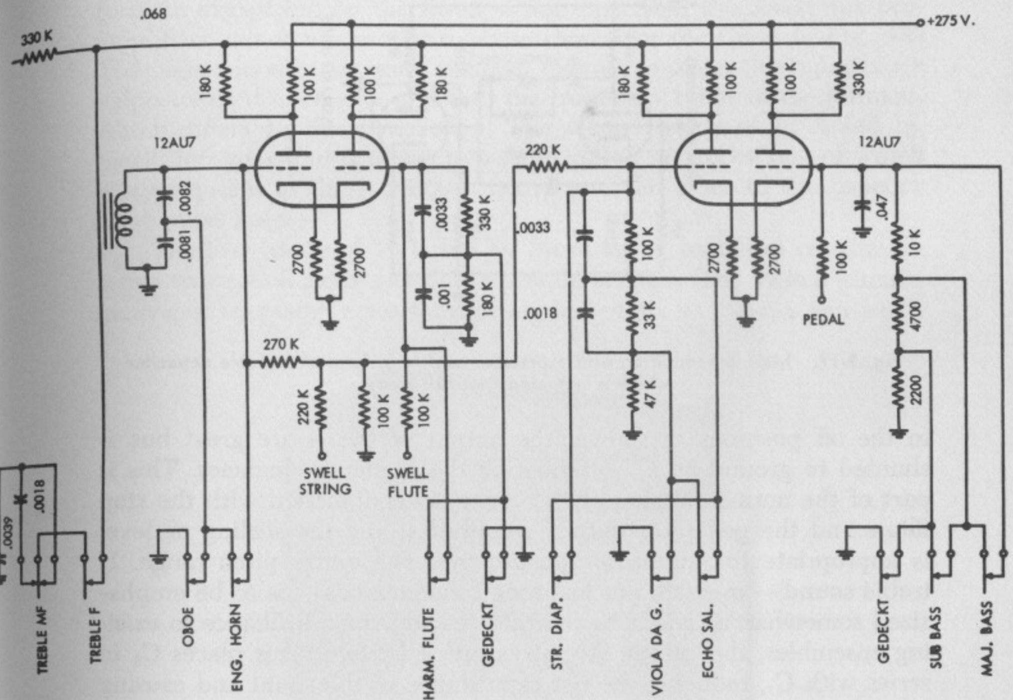


Fig. 8-16. Schematic of Swell and Pedal mixer network.

With only the DULCIANA switch open, R_9 is shorted to ground, and only C_4–R_8 remains in the grid circuit, giving the string tone a definite, though not great rolloff, so that a soft and not strident string tone is heard.

The left half of V_1 also handles two stops, but these are of a flute character. With only the MELODIA switch open, C_5–R_{11} is between grid and ground, rolling off much of the existing harmonic content of the flute tone which is fed to grid through R_{12} and giving a very rounded, smooth tone. With only the GROSS FLUTE switch open, R_{10} is between grid and ground, leaving the harmonic content intact and yielding a flute tone which is of greater complexity and interest.

The last triode of V_2 uses a mixture of flute and string to achieve a full-bodied, penetrating OPEN DIAPASON. Flute tone goes directly to the grid through R_{13}, where it is rounded and given a slight formant by the grid-circuit components. String tone is also introduced through C_6 at a different point. Speaking of the diapason is controlled simply by shorting the grid.

There are two additional tabs connected with the Great division, TREBLE F (*forte*) and TREBLE MF (*mezzo-forte*). With these switches

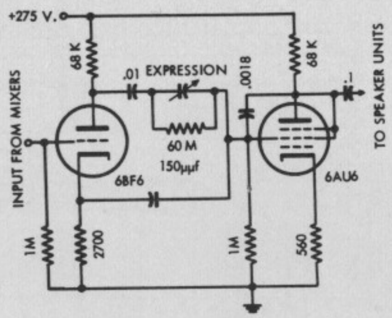


Fig. 8-17. Main preamplifier, with expression control in form of variable capacitor which is actuated by swell pedal.

in the off positions as shown, the output of the entire great bus is shunted to ground by C_7, attenuating the higher frequencies. This is part of the normal voicing of the organ, in conjunction with the stop filters and the generator output amplitudes, and the scaling or level is appropriate for normal organ use over the entire pitch range. If treble sound — over-all, not just tone fundamentals — is to be emphasized somewhat, as might be desirable to add some brilliance to existing ensembles, the TREBLE MF tab is pressed down. This places C_8 in series with C_7, reducing the net capacitance of the shunt and causing highs to rise relative to bass and middle. If an additional treble emphasis is desired, as it might be when playing on a single manual to emphasize melody, the TREBLE F tab is pushed. This completely disconnects the capacitive shunt. If both tabs are pressed, TREBLE F will, of course, take precedence.

The swell and pedal mixers are shown schematically in Fig. 8-16. The first two tabs give the same treble-emphasis results as their

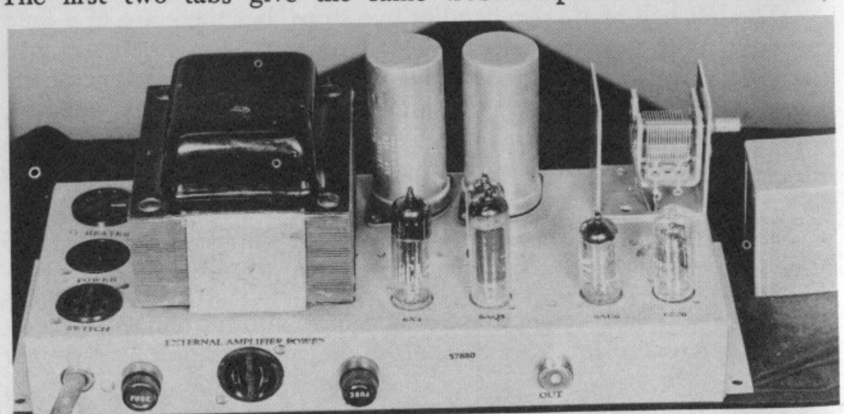


Fig. 8-18. Electronically regulated power supply and the preamplifier occupy the same chassis and provide interconnection between components.

counterparts in the great circuit. The left grid of V_1 has a simple tuned formant circuit fed by the Swell string tone from the Swell bus preamplifiers, which are similar to those shown for the Great in Fig. 8-7. The OBOE and ENGLISH HORN switches operate to select the capacitance value across the inductor and thus the frequency range of the formant, which differs for the two voices. The second triode of V_1 is fed by Swell flute tone and contains two series rolloff networks, one of which is appropriate to the HARMONIC FLUTE and the other to the GEDECKT or stopped flute.

The first triode of V_2 is fed by Swell string tone and contains in series two rolloff networks plus a simple resistor. The switch circuits for STRING DIAPASON, ECHO SALICIONAL, and VIOLA DA GAMBA can easily

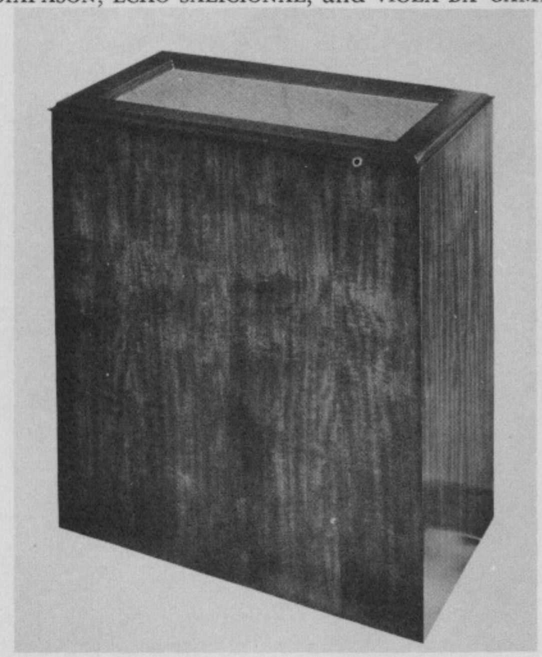


Fig. 8-19. "Tone Cabinet" — loudspeaker system with self-contained power amplifier — suitable for home use.

be traced to show that the first puts all the rolloff in the circuit, the second only some for a stringier tone, and the last none at all, leaving only the 47,000-ohm resistor.

The final triode handles the pedal stops. The characteristics of the three voices are determined almost entirely by the rolloff due to the series 100,000-ohm resistor and the .047-μf capacitor to ground. The three series grid resistors and the switches give three different levels. There is, of course, some change in quality when the total grid resistance is changed because this resistance shunts the .047-μf capacitor.

AMPLIFIERS AND SPEAKERS

Collected tone from the combined mixer outputs is fed to the input of the main preamplifier diagrammed in Fig. 8-17. This circuit includes the swell-shoe or expression control, which is capacitive and has been designed to get around the eventual difficulty with noise which is found with almost every type of pedal-operated potentiom-

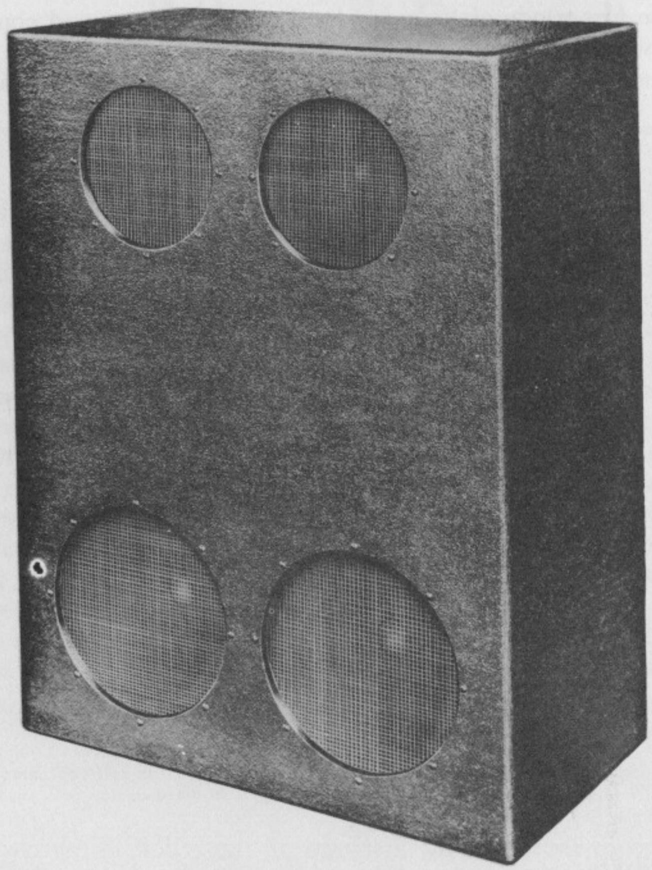


Fig. 8-20. Designed for installation in concealed location, this utility tone cabinet is equipped with a 40-watt amplifier.

eter. The chassis which holds the preamplifier and the electronically regulated power supply for the entire console may be seen on the floor of the console shell in Fig. 8-6 and in a close-up in Fig. 8-18. The expression control is the radio-type variable capacitor seen on the right end of the chassis in the latter figure; the swell-shoe is connected to it mechanically so that operation of the shoe rotates the rotor.

The variable capacitor is part of a capacitive voltage divider. The shunt leg of the capacitive divider is the capacitive input of the 6AU6 stage with capacitive feedback. The 150-$\mu\mu$f fixed capacitor feeds signal of opposite phase from the 6BF6 cathode circuit so that at minimum setting of the variable capacitor a bucking or bridge balance condition is approached. This bucking gives the circuit a greater range of control than the ratio of minimum to maximum capacitance that the variable capacitor would otherwise provide. The 6AU6 feedback capacitor is multiplied in value by the gain of the stage and this lowers the grid-circuit impedance so that a recommended value of grid resistor may be used without loss of 32-cps signal. Output of this stage is fed through cables to the speaker units.

Three loudspeaker units or tone cabinets are available for use with the Artist model, though qualified people can sometimes make installations with nonstandard speakers. The Model 110 unit, shown in Fig. 8-19 contains a 15-inch woofer and a 10-inch "tweeter," with a 20-watt power amplifier. Sound radiates upward. Model 119 has the same specifications but has a utility finish for concealed locations and propagates sound horizontally. The Model 159 unit, shown in Fig. 8-20, also in utility finish for concealed locations, has two 15-inch and two 10-inch speakers, with a 40-watt amplifier. Any number or combination of speaker units can be used, depending on the location, since each contains its own power supply and its signal input is simply bridged across the output line from the console.

Chapter 9

The Lowrey Organo

THE Lowrey Organo, made by the Lowrey Organ Division of Central Commercial Industries, Inc., Chicago, gives the player the facilities of a complete, small, one-manual organ, but has no console of its own. Its unique feature is that it utilizes the keyboard of any piano without impairing the regular operation of the piano. Figure 9-1 shows a complete Organo installation on a standard piano. The cabinet at the right houses the speaker, amplifier, and generators; the attachment on the piano's front is a small control panel; the long, narrow frame across the back of the keyboard contains the mechanism by which pressure on the piano keys brings forth organ tones from the instrument.

The Organo is an electronic instrument which generates its tones in vacuum-tube oscillators. Figure 9-2 is a block diagram showing in a general way what the instrument contains.

The range is five octaves – the standard organ-manual compass – from two octaves below middle C to three octaves above. The tones are generated by frequency-dividing multivibrator chains, twelve in all, each synchronized by a master oscillator and vibrato-controlled by a reactance tube fed by a single vibrato-frequency oscillator for all the chains.

The generators are connected to 60 key switches actuated by the piano keys. The output points of the lower 25 switches are connected in parallel. This line is the lower-register bus which carries all the tones up to middle C. The remaining 35 key-switch outputs are paralleled to form the upper-register bus, which carries all the tones above middle C. Two register switches determine whether one or both of the two registers is to be heard, since the player may wish to hear the piano only in the lower register and the organ tone in the upper register, or *vice versa*. These switches appear on the control panel attached to the front of the piano.

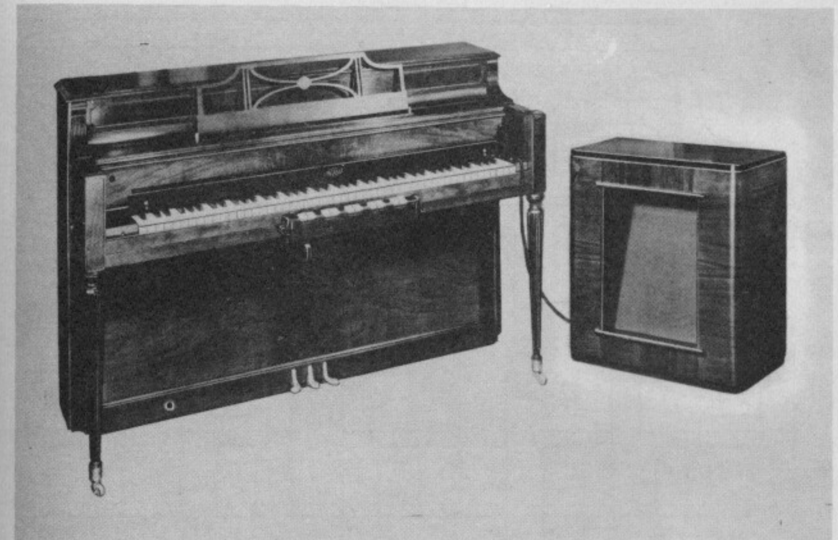


Fig. 9-1. Organo installed on an ordinary piano.

The tones from upper or lower registers, or both, are passed through six *R-C* and *L-C* filters with paralleled inputs and separate outputs. Each imparts a different quality to the tone. These qualities are selected by the stop switches which also appear on the control panel. Three different qualities or stops are available – *principal*, *horn*, and *string* – each loud *(forte* or *f)* or soft *(piano* or *p)*. The solo switches on the panel enable the player to make either register somewhat louder than the other so that a melody may be made to predominate over an accompaniment. The vibrato switches select either a light (small-amplitude) or heavy (large-amplitude) frequency variation at about 6 cycles per second.

The tones selected with the stop switches pass to the amplifier located in the tone cabinet. There is a preset volume control between two of the amplifier stages; and between two other stages is a second

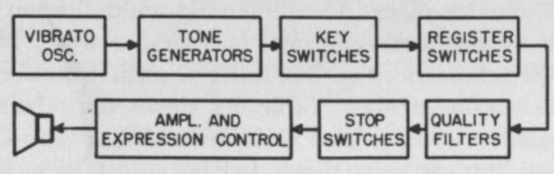


Fig. 9-2. Organo functional block diagram.

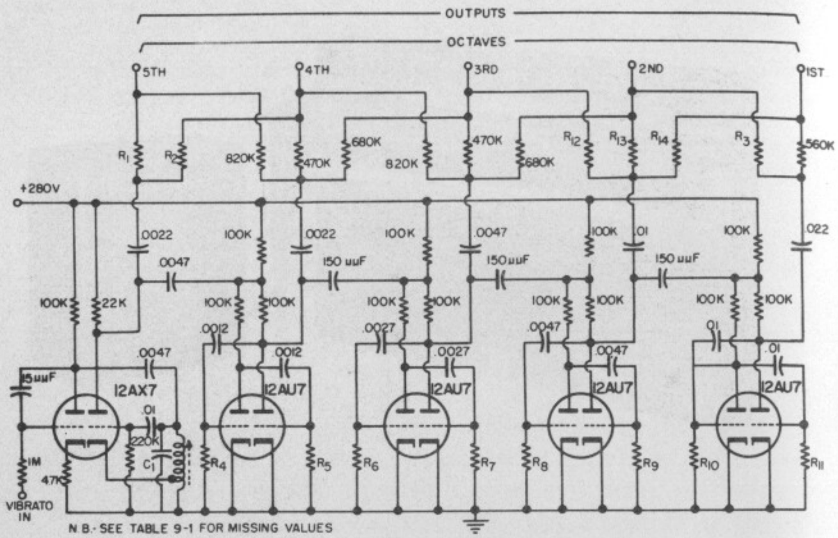


Fig. 9-3. Schematic of a tone-generator chassis.

potentiometer-type volume control, the leads of which are brought to the control panel. This is the expression control, operated by a knee lever.

TONE GENERATORS

The heart of the Organo is the tone-generator system with its 60 tubes and 12 chassis. Each note, A through G sharp, is generated in the third octave above middle C by an inductively tuned master oscillator. A typical generator string is shown in Fig. 9-3, with the master oscillator the second triode of the 12AX7. The oscillator may be tuned by adjusting the slug within the coil form.

Each of the following twin-triode 12AU7's is a multivibrator with the customary cross-connected plates and grids. The first multivibrator is synchronized to the master-oscillator frequency by the capacitive connection between the oscillator plate and the resistive network in the multivibrator plate. The remaining multivibrators are synchronized, each by the preceding one, in a similar way. The unmarked values on the schematic, Fig. 9-3, differ according to the notes generated by the various strings. Table 9-1 shows the missing values.

The master oscillator and each multivibrator furnish an output lead, the five outputs being separated by an octave each. Each output is taken from a plate through a capacitor and resistor. These outputs are not single-frequency tones, however. Each of the lower

TABLE 9-1

Values For Fig. 9-3. All Resistances In Megohms.

NOTE	R1	R2	R3	R4-R11	R12	R13	R14	C1
C	0.22	0.33	8.2	0.82	2.7	0.56	1.0	.039
C#	0.22	0.33	8.2	0.82	2.7	0.56	1.0	.039
D	0.22	0.33	8.2	0.82	2.7	0.56	1.0	.033
D#	0.22	0.33	8.2	0.68	2.7	0.56	1.0	.033
E	0.15	0.22	5.6	0.68	1.8	0.56	0.82	.027
F	0.15	0.22	5.6	0.68	1.8	0.56	0.82	.027
F#	0.15	0.22	5.6	0.47	1.8	0.56	0.82	.027
G	0.15	0.22	5.6	0.47	1.8	0.56	0.82	.027
G#	0.1	0.15	3.9	0.47	1.0	0.47	0.68	.022
A	0.1	0.15	3.9	0.47	1.0	0.47	0.68	.022
A#	0.1	0.15	3.9	0.47	1.0	0.47	0.68	.018
B	0.1	0.15	3.9	0.47	1.0	0.47	0.68	.018

(1 through 4) stages borrows some tone from the stage above it through resistors R_2, R_{14}, and the other similarly placed. In addition, each of the higher four stages (2 through 5) borrows some tone from the one below it through R_3, R_{12}, and so on. Thus each output contains principally the tone which it is supposed nominally to supply, with the addition of some tone an octave higher and an octave lower.

The first triode of the 12AX7 is a reactance tube placed across the master-oscillator tuned circuit. The grid is controlled by a signal from the instrument's 6-cycle vibrato oscillator. Varying the pitch of the master oscillator varies that of all the tones in the string because of the synchronism.

Figure 9-4 is a photograph of a portion of the key switch frame with the cover removed. The frame is placed over the rear of the keyboard and is fastened down with adjustable end brackets. Sixty small plungers project downward and rest on sixty of the piano keys. The shorter plungers in Fig. 9-4 are those which touch the black keys.

Each switch is a single-pole, double-throw unit, the arm of which is a small coil spring which moves up and down with the plunger when the key is pressed. The spring, connected to a generator output, is grounded to the topmost bus bar (see photo) when the key is up, short-circuiting the tone to ground. When the key is pressed, the spring contacts the lower bus bar. The lower bar is divided into two parts, the lower part encompassing the first 25 notes – the lower register – and the upper part of the remaining notes – the upper register. These are collector bars on which all generator outputs switched by the player appear.

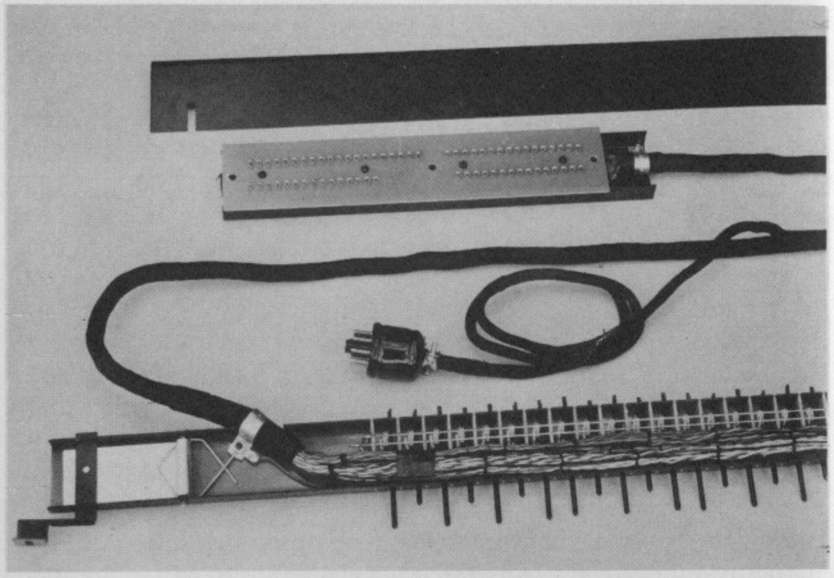


Fig. 9-4. Key-switch frame with cover off.

The vibrato oscillator is shown in Fig. 9-5. It is a simple phase-shift unit located on the power-supply chassis. The left triode of the 6SL7-GT is the oscillator itself, with its grid connected to the second triode which acts as an amplifier. Output is taken from the plate of the amplifier to modulate the twelve reactance tubes in the generator strings.

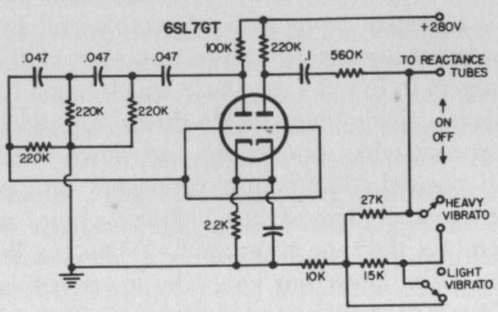


Fig. 9-5. The vibrato oscillator.

The degree of vibrato (its amplitude, not frequency) is controlled by a pair of switches and three resistors located in the control panel (drawn as part of Fig. 9-5 for easier understanding). With both HEAVY VIBRATO and LIGHT VIBRATO switches in the OFF position the amplifier output goes through a voltage divider consisting of the 560,000-ohm resistor as one element and a total of 10,000 ohms in the switching network as the second element, across which the reactance-tube grids are connected. The reduction in output voltage reduces the vibrato effect to negligibility.

If the LIGHT VIBRATO switch is placed in the ON position, the net resistance of the switching network rises to 10,000 plus about 9,600 (27,000 and 15,000 in parallel) ohms or about 20,000 ohms, which raises the network output sufficiently to obtain a moderate vibrato effect. When the HEAVY VIBRATO switch is on, the total in the switching network is 27,000 plus 10,000 or 37,000 ohms, which raises the vibrato effect to a large value.

TONE COLORING

The key-switch upper- and lower-register collector bars are connected to the circuit shown in Fig. 9-6, the components of which are located in the control panel. Each register goes to a switch which either grounds the bus or passes it on to a voltage-divider network and solo switch. If both registers are to be played at the same volume the solo switch is in the normal position, as the lower-register solo switch is in Fig. 9-6. The upper element of the voltage divider is the resistor from the plate of a multivibrator tube to the octave output terminal in Fig. 9-3; the lower element, across which the output is taken, is whatever appears between the register switch and ground in Fig. 9-6. With the solo switch at normal, the lower element is 4,700 ohms. With the switch in solo position, it is 22,000 ohms. Thus the register selected is louder than the other and a solo melody may be played on it, with a softer accompaniment on the other. This is one form of "split keyboard" very often employed on single-manual organs. The split keyboard is common in nonelectric instruments such as the old-fashioned harmonium, where it usually has a separate group of stops — tone colors — as well as different volume level.

The tones emerging from the solo-switch network are fed to two buses, one for each register. The buses are brought together through 500-$\mu\mu$f capacitors to the first stop switch, labeled STRING *p*. The small value of capacitance tends to filter out low frequencies from the complex waveform of each note, producing pulse-type waveforms with fairly sharp "spikes." This simulates a string tone, which has this character because the hairs of the bow have comparatively rough surfaces and pull the string in little jerks.

The buses are connected through a second pair of capacitors with a value of .0018μf to the STRING *f* switch. This gives the tone a bit more body, changing the quality somewhat and in the process allowing more of the fundamental to come through with an effect of greater volume.

All horns are resonant in at least one frequency range, as explained in chapter 4. The horn effect is simulated in the Organo by an *L-C* resonant filter composed of two .0027-μf capacitors and a 24-henry inductor in series. This is the *piano* or soft horn. For the *forte* or loud horn one of the capacitors is increased to .01 μf, with a lowering of resonant frequency.

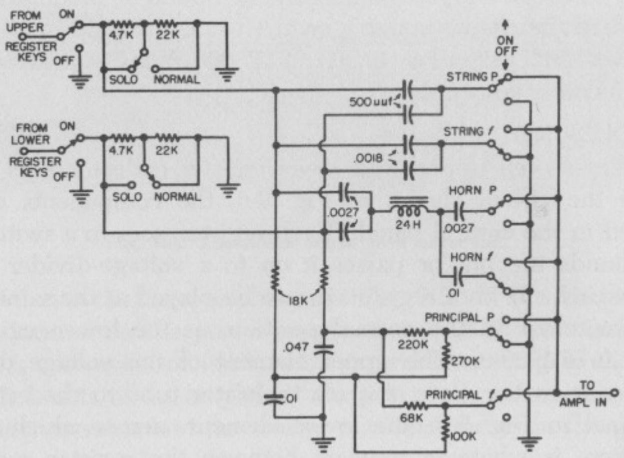


Fig. 9-6. Tone-color filter circuit.

The principal tone is simply the normal output waveform of the generators, with some of the higher harmonics reduced. The principal networks consist of a pair of resistors and a capacitor in a T configuration. The second resistor value is lowered from the value used for *piano* to obtain the *forte* volume.

Each stop switch is single-pole, double-throw and the arm is connected to ground for the OFF position and to an output bus for the ON position. The bus is connected to the amplifier.

AMPLIFIER AND OTHER EQUIPMENT

The amplifier is pictured in Fig. 9-7 and diagrammed in Fig. 9-8. Tubes V_5 and V_6 in the photograph are the power supply rectifier and the vibrato oscillator.

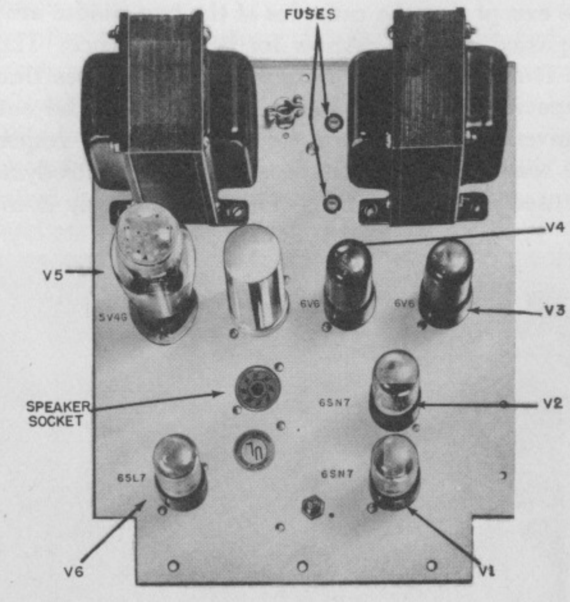


Fig. 9-7. The amplifier.

Referring to Fig. 9-8, the first triode of a 6SN7-GT is grid-excited by the output of the stop-switch bus in Fig. 9-6. Between this and the second triode is a volume control which is preset at installation for the maximum volume desired in the room. The second triode is a cathode-follower, included to obtain a low-impedance line feeding to the expression control. The expression control, a 2,000-ohm potentiometer, is located in the control panel, the inside of which appears in Fig. 9-9. The lever and spring mechanism operated by the knee-lever turns the potentiometer. The low-impedance cathode line and the low resistance of the potentiometer prevent hum pickup or appreciable loss of treble in the line between the amplifier (which is in the tone cabinet) and the control panel on the piano.

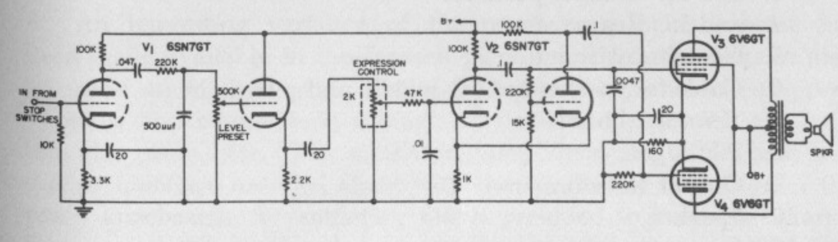


Fig. 9-8. Diagram of the Organo amplifier.

The next stage is a phase inverter of the classical tapped-grid-resistor type except that the cathodes of the two triodes are commoned to an unbypassed cathode resistor for better balance. The two plate outputs are fed to the grids of the 6V6 output tubes through 0.1-μf blocking capacitors. The .0047-μf capacitor across the output of the first phase-inverter tube reduces the high-frequency response slightly to eliminate some noise. The loudspeaker is an electrodynamic whose field coil is used as a filter choke. The power supply is conventional.

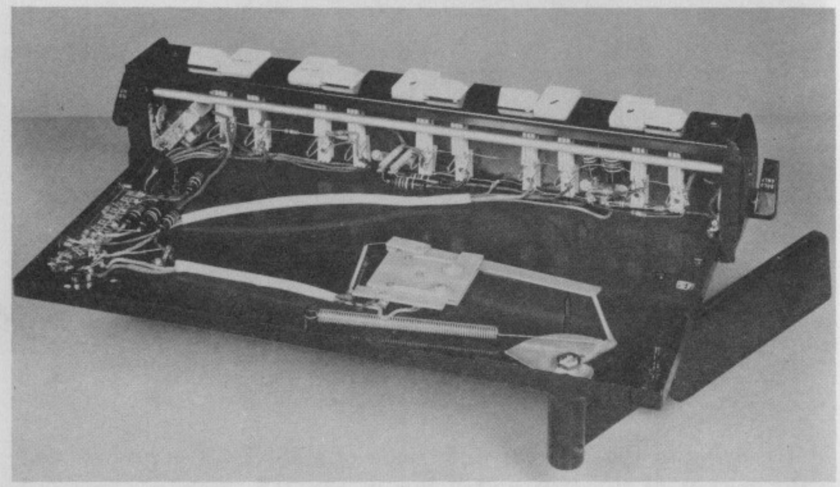


Fig. 9-9. Control panel on piano front includes a knee-action volume control.

The layout of the Organo components is simple and straightforward. The control panel shown in Fig. 9-9 contains all the tone-coloring components as well as the expression control and the switches.

Figure 9-10 is a rear view of the tone cabinet with the covers removed. The numbered components are as follows:

1. Auxiliary power switch (there is a switch and pilot light on the front of the cabinet).
2. and 18. Power and filament transformers.
3. and 14. Cabinet portions.
4. and 12. Rear covers.
5. Component frame.
6. Cable for speaker, power-switch, and pilot-light.
7. Terminal board.
8. Pilot light.
9. Output transformer.
10. Speaker.
11. Generator tubes.

13. Speaker plug.
15. Power cord.
16. Rectifier tube (a pair of 6W4's has been included in the newer models using 12AU7 generator tubes instead of 12AX7 as formerly).
17. Filter capacitor.

All connections to and between the various units are brought to the terminal boards at the center, so that all units are completely accessible and may be removed readily for repairs.

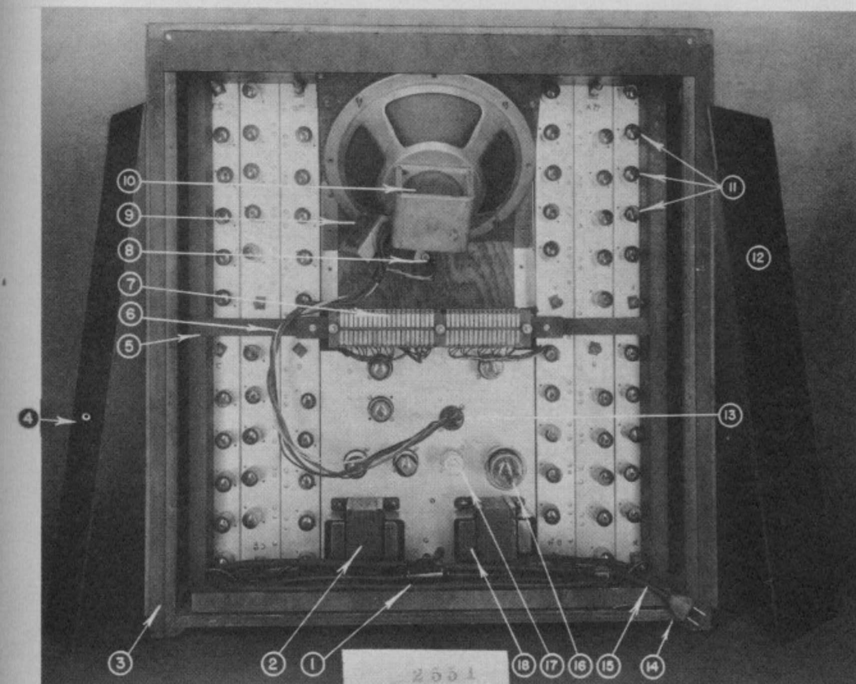


Fig. 9-10. Rear view of the tone cabinet, cover removed.

An interesting variation of the usual components scheme has been made available in the Janssen piano, of which the Organo may be made an integral part. Figure 9-11 shows a piano so equipped. There is no separate tone cabinet; all units of the Organo are built into the piano case. The oscillators, amplifiers, and other heat-producing units are installed along with the speaker on the inside of the piano kneeboard. A ventilating fan is provided to keep the interior cool enough to prevent damage to the piano mechanism. The key

Fig. 9-11. A Janssen piano with built-in Organo.

switch frame is placed inside the piano out of sight toward the rear ends of the keys.

In playing an ordinary piano equipped with an Organo attachment, striking the piano keys with normal pressure will actuate both mechanisms and produce both piano and organ tones simultaneously. However, in experimenting with the instrument, the writer has found that it is possible to play the Organo without appreciable "interference" apparent from the piano strings. It is a matter of regulating touch, since the key switches are adjusted so that tone will be heard when the key is depressed about halfway. Full depression sounds both piano and organ tone, the effect of which can be quite pleasing. However, a damper bar may be obtained and installed in the piano. It has a lever at the end, with which the bar can be rotated on its axis, placing damping material on the strings so that they do not sound. The action then produces only organ tones. Damper bars are not common, however, and users do not seem to have felt a great need for them.

Chapter 10

The Hammond Solovox

THE Hammond Solovox (Hammond Instrument Company, Chicago, Ill.) is and has been for a number of years the best-known monophonic electronic musical instrument made in this country. While it permits the playing of only a single note at a time and has neither the types nor variety of tone colors to appeal to most serious musicians, it has found extensive use in popular-music ensembles and particularly in the home among amateurs. It is ordinarily used with the piano – the player's right hand playing the melody on the Solovox and the left hand playing an accompaniment on the piano – where its sustained tones contrast with the percussive effects of the piano to produce pleasing sounds with a minimum of study, effort, and expense.

Unlike the Hammond organ, the Solovox is completely electronic, with no moving parts other than the keys and controls. (To be accurate, the older models J and K had a vibrating-reed vibrato system, but later models are electronic in even this respect.) This chapter describes the model L.

A typical Solovox installation is shown in Fig. 10-1. The two principal units are the keyboard and a tone cabinet. The tone cabinet (not shown in the illustration) houses the speaker and all vacuum tubes. The keyboard, attached to the front of the piano, holds the switches which are actuated by the keys, and all control circuits.

Figure 10-2 is a block diagram of the Solovox which shows the principal components and their relationships. A master oscillator is tuned through a three-octave range by the keys, and its frequency may be varied cyclically to produce vibrato. The oscillator synchonizes a frequency divider which produces similar tones one octave lower down. This, in turn, synchronizes a second and third frequency divider.

Fig. 10-1. Typical Solovox installation.

One output is taken from the oscillator, which operates in the highest three-octave range of the instrument, and one from each of the frequency dividers. In accordance with their pitch ranges the outputs are labelled SOPRANO, CONTRALTO, TENOR, and BASS. Any one or a combination may be switched into a common output bus which goes to a series of five tone filters. Whatever tone qualities are switched on are passed to a preamplifier, thence to a push-pull control stage. In the control stage the output volume is controlled by a knee lever fastened under the keyboard, and the attack of each note is slowed down by using an extra set of keyboard contacts to remove cutoff bias on the stage at a controlled rate. From here the signal passes to an output stage and loudspeaker.

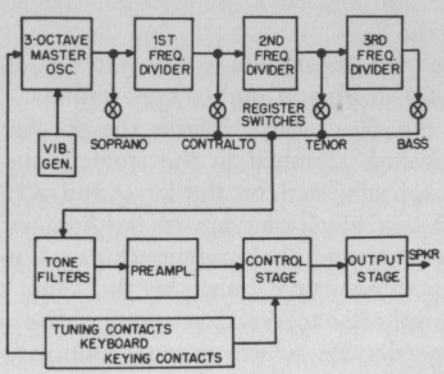


Fig. 10-2. Solovox block diagram.

MASTER OSCILLATOR

The master-oscillator circuit is diagrammed in Fig. 10-3. It is basically very simple and capable of wide-range operation. The output of V_{1a} is *R-C* coupled to the grid of V_{2a}. The plate of V_{2a} is *R-C* coupled back to the grid of V_{1a}. Since the circuit is re-entrant and the feedback phase is positive, the combination oscillates. The frequency of oscillation is controlled by capacitance and resistance placed between the grid of V_{1a} and ground.

Two step-type controls are provided. C_1 is the main tuning capacitor, and C_2, C_3, and C_4 are added to give the exact value required for the fixed capacitance. The next bank of six .002-μf capacitors is so arranged that as the arm labeled ROUGH TUNING is moved

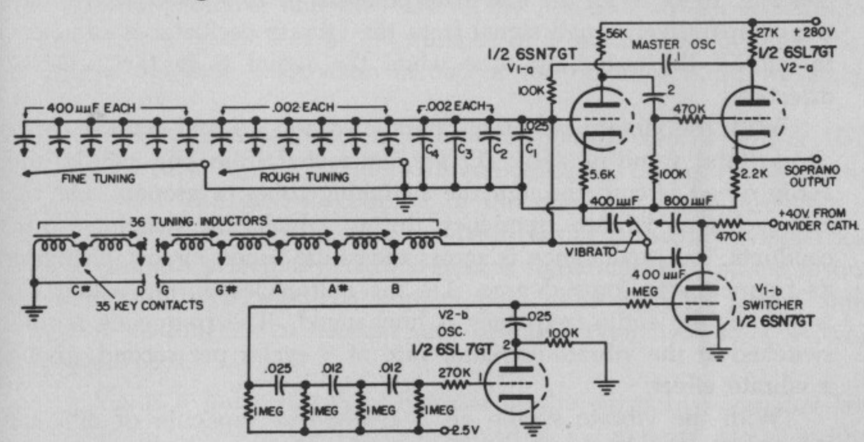


Fig. 10-3. The Solovox master oscillator.

upward, it contacts the capacitors one by one and at maximum position all six are across the main tuning capacitors. Six steps of 400 $\mu\mu$f each are available with a similar switching arrangement for the fine-tuning adjustment.

The inductance part of the tuned circuit is used to vary the frequency to give the pitch required for each key over the three-octave range of the oscillator (523.3 to 3951 cycles). Thirty-six small inductors, each with a movable core for individual adjustment, are wired in series, with a switch contact connected between each adjacent pair. When a key is pressed, the corresponding contact is grounded, decreasing the net inductance between the grid of V_{1a} and ground. When the lowest C key is pressed, no contact operates — the full value of all the series inductors is in place and the lowest frequency sounds. (Actually this frequency is being generated all the time, even when no key is being pressed. The control tubes.

however, block the amplifier until a key is pressed, as described later.) When the topmost key (B of the top octave) is pressed, all inductors but the one nearest the grid are shorted to ground and the remaining inductance is just large enough to tune the circuit to the topmost note. With this arrangement, no spurious notes are heard even when two or more keys are pressed at once by accident; only the uppermost of the notes played is heard.

The vibrato circuit includes the two remaining triodes of the 6SN7-GT and 6SL7-GT used in the oscillator. V_{2b} is the 6-cycle oscillator, a standard phase-shift unit, the output of which is fed to V_{1b}, a switching tube. The plate-supply potential for V_{1b} is only 40 volts, obtained from the drop across the frequency-divider cathodes (see Fig. 10-4). With the low plate potential of 10 volts and zero bias, the comparatively small signal from the vibrato oscillator is sufficient to cut off the switching tube when the signal is in the negative direction.

With the vibrato switch on, paralleled 400-$\mu\mu$f and 800-$\mu\mu$f capacitors (total value of .0012 μf) are connected from one side of the audio tuned circuit, through the switching tube, to ground. On the half-cycles of vibrato frequency during which the switching tube conducts, the capacitance is across the audio tuned circuit, changing its frequency to some degree. On the half-cycles during which V_{1b} is cut off, the audio frequency is unchanged. The frequency is thus switched at the vibrato-oscillator rate of 6 cycles per second, giving a vibrato effect.

With the vibrato switch off, a permanent capacitor of 400 $\mu\mu$f is connected across the tuned circuit. The 400-$\mu\mu$f capacitor between V_{1b} plate and the audio tuned circuit is still in place and the switching tube is still doing its job. The vibrato effect is therefore still present to a very slight degree. This is desirable, for it is enough to destroy the "perfection" of the electronically generated tones, which would otherwise be so perfectly steady as to lack interest.

This is an interesting point in all electronic instruments – perfection is undesirable! A pipe organ, a wind instrument, a violin – all of them have inherent random irregularities of pitch and volume caused by small variations in the wind supply or slight unsteadiness in the player's control. One of the essential factors in art appreciation by the emotions is variation; monotony is inartistic and unpleasant. Thus the natural slight unsteadiness of acoustic instruments is welcome, and to attain a really ideal musical instrument, the electronic engineer should deliberately avoid the perfection which we normally look for in engineering. In one very practical sense this is a major difference between a set of code-practice oscillators tuned to musical

pitches and a good electronic musical instrument. The code oscillators have constant pitch and no variation in tonal quality. The musical instrument must have at least a vibrato and a selection of tone colors.

FREQUENCY DIVIDERS

Frequency dividers are very common in electronic musical instruments. In polyphonic instruments there is usually one set of them for each of the twelve notes of the scale and each divider need work at only one frequency. In the Solovox there is only a single set of three dividers, each of which must work over a three-octave range. To fulfill this requirement they are designed to be not particularly frequency-selective and they are of the non-oscillating multivibrator type. This means that in the absence of a synchronizing signal they do not oscillate but remain in one or the other of their two stable conditions.

The frequency-divider section of the Solovox appears in Fig. 10-4. The first tube V_{3a} is a rectifier which rectifies the output of the master oscillator. The grid is coupled to the oscillator plate through C_1. The waveshape of the master oscillator output is roughly symmetrical and contains principally odd harmonics, sounding like a muted instrument, or a woodwind, or stopped pipe. The rectified output from V_{3a} is no longer symmetrical and contains even harmonics as well as odd ones.

V_{3b} is a pulse rectifier. Because of the capacitive coupling between V_{3a} and V_{3b}, the rectified output of V_{3a} appears as a.c. on the grid of V_{3b}. Due to the average plate currents of the tubes in the dividers the cathode of V_{3b} is 40 volts positive, giving the tube a high negative grid bias. Negative input to the grid therefore has no effect, but positive input increases the plate current and produces negative pulses at the plate.

V_{4a} and V_{4b} are the first frequency divider. In the resting condition one of the tubes is cut off and the other is conducting. When the driver V_{3b} puts out a negative pulse, it passes through C_3 and C_4 to the divider grids. The negative pulse has no effect on the tube which is cut off, for instance, V_{4a}. However, it causes the tube which is drawing current, V_{4b} in this case, to put out a positive pulse at its plate. The positive pulse is transferred to the grid of V_{4a} through the R-C plate network, causing V_{4a} to conduct and put out a negative pulse of its own. This negative pulse is transferred to the grid of V_{4b}, adding to the negative input signal. In a very short time V_{4b} is cut off and V_{4a} is conducting, the reverse of the original state. At this point the circuit is again stable. However, the next negative

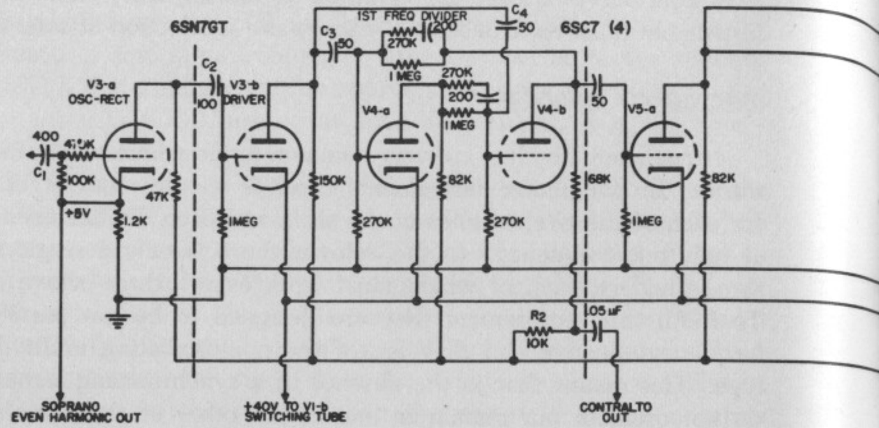


Fig. 10-4. Frequency dividers of the Solovox.

pulse to come from the plate of V_{3b} starts things all over again and in the same manner V_{4a} now cuts off and V_{4b} conducts.

In this way, it takes two input pulses from V_{3b} to make the V_4 circuit execute a complete cycle of change, returning to its original condition. Since the frequency of pulses from V_{3b} is the oscillator frequency, the contralto output taken from the plate circuit of the divider at the junction of R_1 and R_2 is one-half the oscillator frequency and has a rectangular waveshape.

Each of the two following dividers works in the same way and includes a driver-rectifier triode (V_{5a} and V_{5b}) and two multivibrator triodes (V_{6a} and V_{6b}, V_{7a} and V_{7b}). In each case the input frequency is divided in half, so that there are four outputs an octave apart from the generating section of the instrument – one from the oscillator and three from the dividers.

TONE COLOR SELECTION

Two circuits are shown in Fig. 10-5. The first is the register-control section of the Solovox and the second the tone controls.

The register controls do two jobs. The first and most obvious is the selection of which ranges shall be sounded. The entire instrument covers six octaves. When the soprano switch is closed the three highest octaves are fed to the output bus. When the contralto switch is closed, each note keyed sounds one octave lower, which adds one lower octave to the instrument, and so on down to the bass switch.

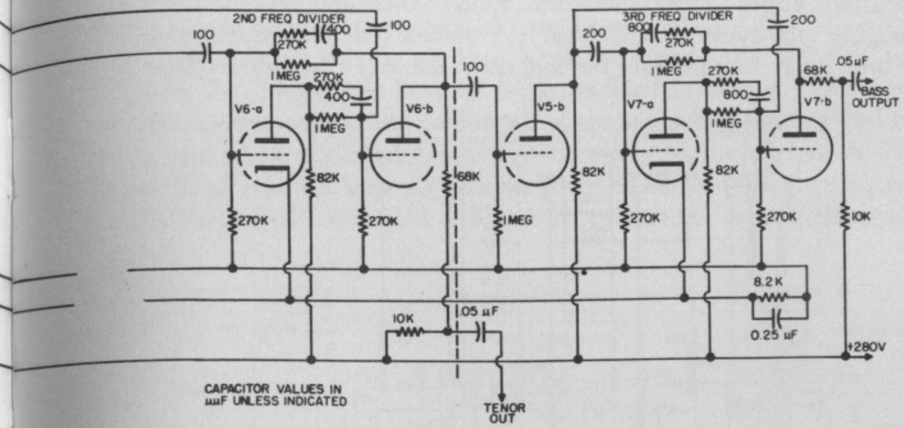


Fig. 10-4. Frequency dividers of the Solovox.

The outputs of the frequency dividers are symmetrical in waveform. This is a "muted" tone carrying only odd harmonics. To add even harmonics to the tones of the bass register, a certain amount of tenor output can be added, since the fundamental of the tenor is the second harmonic (one octave above) of the fundamental of the bass. The amplitude of the added tenor must be kept down so that it does not sound to the ear like an additional octave repetition. If, for example, the original bass tone has ⅓ as much third harmonic as fundamental, 1/9 as much fifth harmonic as fundamental, and so on, then adding a tenor fundamental about ½ as loud as the bass fundamental will give a second harmonic to the bass which will simply fill out the harmonic content of the bass in the correct proportion. The effect is to get rid of the muted tone quality.

With the mute switch of Fig. 5 in the off position the bass receives some additional tone for this purpose from tenor; tenor receives some from contralto; and contralto receives some from soprano. Since there is nothing higher than soprano the special oscillator rectifier of Fig. 10-4 supplies even-harmonic additions for the soprano tone. With the mute switch on, these additions are removed and the muted quality is sounded.

There are five tone-control switches in Fig. 10-5. All the filters are in series across the register-control output bus. The tone switches are normally closed; when a tone quality is selected the switch is opened. With all but the DEEP TONE switches closed, the tone develops across a highly capacitive load and the high-frequency harmonics are very much reduced. With the FULL TONE switch open, frequency response is almost flat, but with some attenuation of treble to round off the sharp edges of the waveform. FIRST VOICE and SECOND VOICE are roughly 400-cycle and 800-cycle resonant circuits which give brassy and reedy tones, while the BRILLIANT filter is a high-brass filter

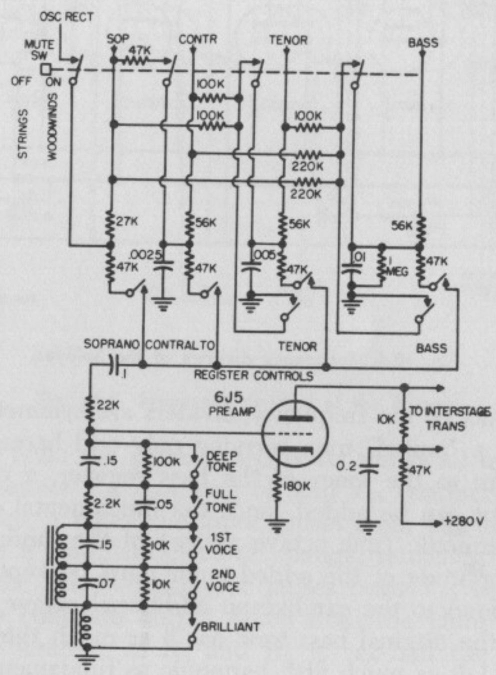


Fig. 10-5. Register and tone controls.

with a very bright tone. The filters are in the grid circuit of the 6J5 preamplifier, the plate of which couples to the control stage through a transformer.

CONTROL AND OUTPUT STAGES

The control stage, diagrammed in Fig. 10-6, consists of a pair of push-pull, remote-cutoff 6SK7's. The two cathodes are connected to the center of a voltage divider consisting of R_1 as one leg and R_2, R_3, R_4, and R_5 in parallel as the second leg. Normal voltage at the cathode tap is 145 because of the connection of plus 290 volts to the input of the voltage divider. The center of the voltage divider is also connected through R_6 and R_7 to a bus running the length of the keyboard. One of the contacts under each key is permanently grounded. When any key is pressed the bus is grounded, placing R_6 and R_7

across the lower (cathode) leg of the divider and reducing cathode voltage to about 45. The rate of drop of the voltage is controlled by the 6-μf capacitor, which causes a time delay.

A second voltage divider is placed across the 290-volt supply, with R_8 as its upper leg and R_9 as its fixed lower leg. The center of this divider goes to the center-tap of the input transformer and thence to the grids. Shunted across the lower leg of the divider is the network of resistors and the selector switch shown, which varies volume by varying the d.c. grid voltage. At maximum volume the grids are

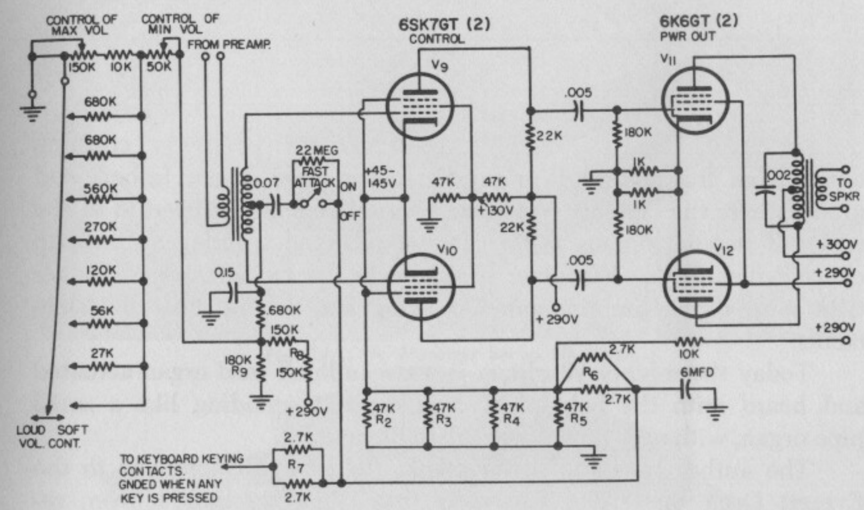


Fig. 10-6. Control stage delays attack and decay.

about 30 volts above ground. Since the cathode, when a key is down, is about 45 volts positive, the net grid bias is about 15 volts and the tubes operate normally. As the volume control is closed the lower leg of the grid divider is shunted more and more and the grids become less positive. They therefore become more and more negative with respect to cathode. When no key is pressed the cathodes are about 145 volts positive, enough to cut off the tubes no matter what the position of the volume control.

The fall of grid-cathode voltage when a key is pressed is normally slowed up additionally by the .07-μf capacitor between transformer center-tap the keying circuit. For faster attack the switch is opened and a 22-megohm click-suppressing resistor is in series with the capacitor.

The output stage is conventional, using a pair of 6K6 tubes.

Chapter 11

The Wurlitzer Organ

"AND then came a crashing chord from the mighty Wurlitzer!" In the days of our fathers and grandfathers the "mighty Wurlitzer" was as much of a byword in the realm of popular organs as the Hammond organ is today in the soap opera radio studio. Wurlitzer has been in the music business longer than most of us can remember, making and selling musical instruments.

Today the mighty Wurlitzer is a streamlined reed organ actuated and heard with the help of electronics, and sounding like a small pipe organ, with nary a reed wheeze to be heard.

The author has often been asked, "What ever happened to the Everett Orgatron?" The answer is that Wurlitzer took it over, refined it, and built it into a line of several large electronic organ models. The one pictured in Fig. 11-1, for instance, is a Wurlitzer Series 50, a complete two-manual organ with 32-note radiating pedal clavier, AGO dimensions, and 22 stops. More and less elaborate models are made, as well.

The newest Wurlitzer is possibly the most interesting of the lot, for several reasons. It is the Model 44 organ, shown in Fig. 11-2. As the photo shows, it is a spinet, but has two short manuals in what is coming to be something of a style among modern spinet electronics, each with 44 notes, from F to C. It also has a toe-pedal clavier of 13 notes, C to C. The power amplifier and speaker are built into the console, which is a complete organ with nothing coming out but the a.c. cable and the music.

What particularly distinguishes the Model 44, however, is the fact that all the reeds are blown all the time, at low air pressure. The keying is done entirely electronically, with consequent elimination of the characteristic slow speaking of reeds which has bothered

Fig. 11-1. A Wurlitzer Series 50 organ.

Fig. 11-2. The Model 44 Wurlitzer spinet organ described in this chapter.

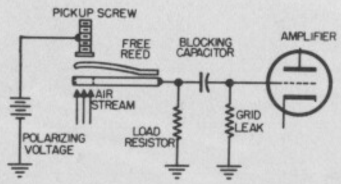


Fig. 11-3. Wurlitzer tone generator is a brass air-blown reed.

some people. The system tends to reduce mechanical ciphers, too, because dirt has little chance to accumulate on a reed that is waving in the breeze all the time. Electronic keying also makes the keying much simpler, doing away with the direct action and its Rube Goldberg lever system or electrical keying with its solenoids by the ton. Another advantage is that the attacks may be made as sharp or as delayed as the designer wishes.

PRINCIPLES OF OPERATION

The Wurlitzer uses the same basic principles as most other electrostatic transducer devices, as illustrated in Fig. 11-3. The basic component is the reed of brass. A stream of air strikes the reed through a slot in the reed cell. If the stream comes upward, the first breath pushes the outer end of the reed upward. Since the reed is a high-Q

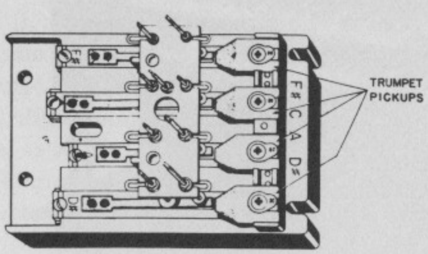


Fig. 11-4. A group of reeds with tone-pickup screws.

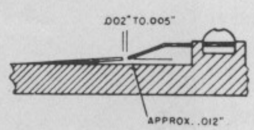


Fig. 11-5. A trumpet pickup.

mechanical resonator, the burst of pressure tends to make it overshoot and go past the point to which the wind pressure would push a nonresonant body. Then the energy stored in the reed because of its elastic properties pushes it down, and again it overshoots the initial resting position, after which it springs back up again. Because of its resonance, the angular velocity of the movement is a function of the frequency of resonance; and the reed always finds some air pres-

sure at the proper point in its cycle to give it back the energy it expended in the last cycle, thus keeps moving at its resonant frequency.

The electrostatic transducer system is necessary to convert the reed motion into audio voltage since the reeds themselves are enclosed in a soundproof compartment. The reed and the pickup screw are the two "plates" of a capacitor whose capacitance varies with the position of the reed. With each variation in capacitance there is a rush of electrons from one plate to the other. The electrons pass through the load resistor and create an audio-frequency voltage drop through it, which is transferred to the grid of a tube. The Model 44 has a bank of 73 reeds which provide all the tones and tone colorings for the organ.

Two types of tone are available directly from the reed pickups. A flute-type tone (not a sine wave, but only moderately rich in harmonics) is picked up by the tone screws directly above the reeds. Most of the reeds have at least two pickup screws, for reasons which will appear. A horn-type tone is picked up by special so-called trumpet pickups, which are bent pieces of sheet metal. Figure 11-4 shows a group of four reeds with two tone screws each (except for the C, which has three) and the trumpet pickups. As Fig. 11-5 makes clearer,

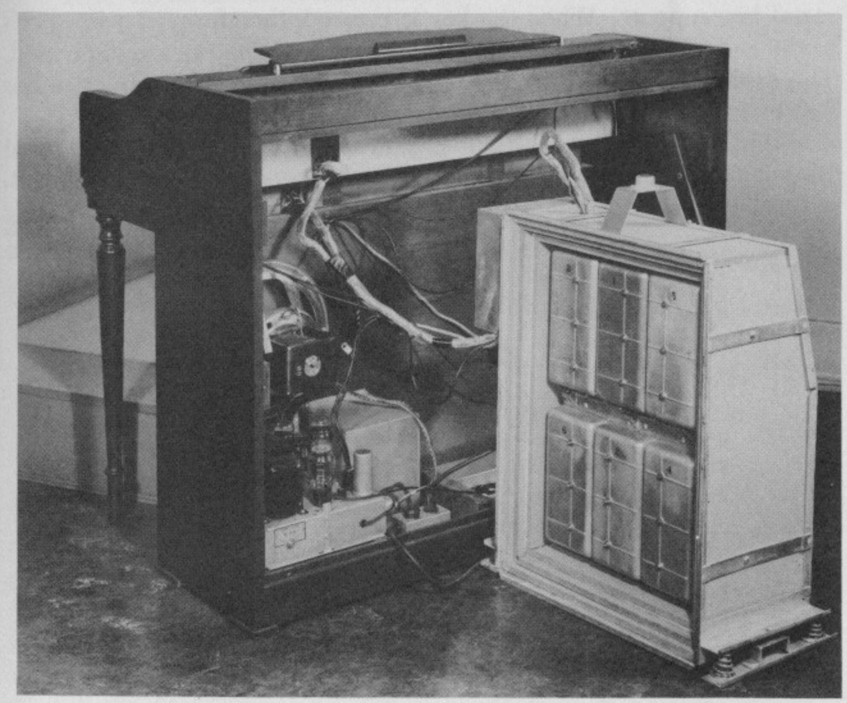


Fig. 11-6. The complete reed unit.

the latter are located a very small distance away from the end of the reed and, as can be imagined, give a rather sharply peaked waveform.

The entire tone unit, containing reeds, blower motor, and windchest, is shown in Fig. 11-6, as it looks when removed from the organ case with the rear of the tone unit taken off for access. Each of the six pans can be removed separately to give access to the group of reeds and pickups under it. Figure 11-7 shows the tone unit with its soundproofing cover in place. This makes it impossible to hear the reeds acoustically. Figure 11-8 shows one of the pans removed to reveal the reeds and pickup screws. Note in Fig. 11-7 that the compressor motor is outside the tone unit but attached to it.

THE KEYING SYSTEM

The basic keying system for the Model 44 is shown in the drawing of Fig. 11-9, which illustrates keying for one pickup of one reed. Reference to Fig. 11-10, a photo of the bass end of the upper manual keying and stop mechanism, will help clarification.

The keys are mounted on a strip of rigid material at the rear of which is attached a short strip of spring metal, the rear of the latter being screwed down to a transverse rail which is a part of the formed metal key support. When the key is pressed, therefore, all parts back to the end of the spring holding the assembly to the rail are depressed. The usual bearing and levelling provisions are toward the front of the key to keep it in line and equalize its action with the others.

When the key is pressed it pushes down the activator, a vertical strip of insulating material with a hole for each of the 10 key con-

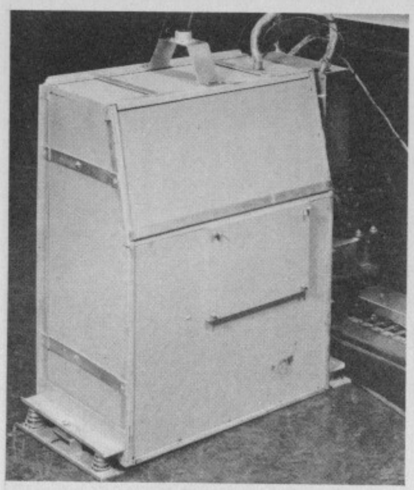


Fig. 11-7. Tone unit with soundproofing in place.

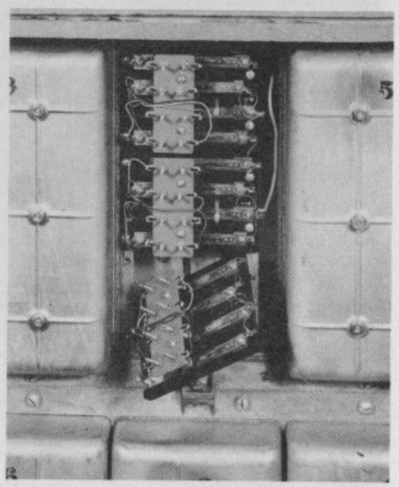


Fig. 11-8. Cover removed from a pan to show reeds and screws.

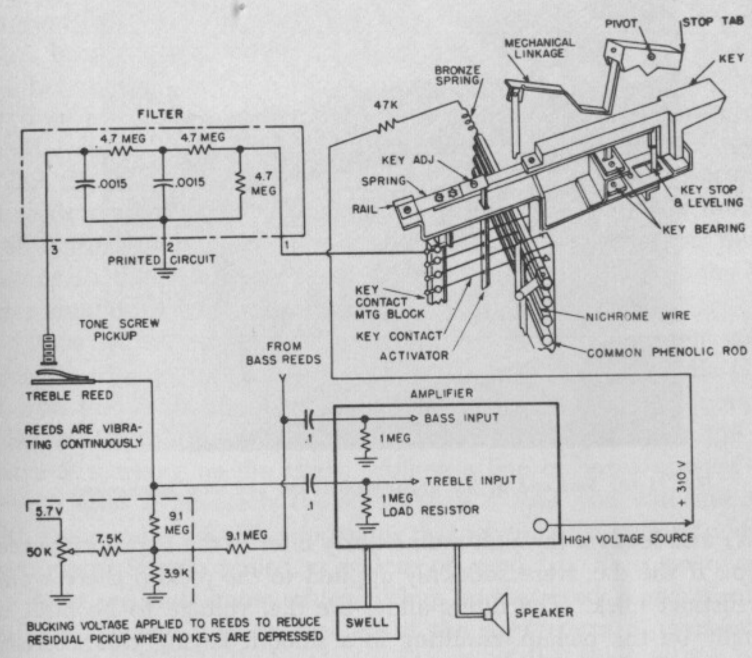


Fig. 11-9. Diagram of the keying system.

tacts (9 for stops and the tenth for gating the amplifier, as we shall see later). Each key contact is a spring wire fastened at the rear to the key contact mounting block free at the other end. Since each contact passes through a small hole in the activator, when the key is pressed and activator lowered, the contacts are pushed downward.

Beneath each contact is a rod of phenolic material with a heavy nichrome wire running its length. See Fig. 11-10 for the manner in which the nichrome wire is mounted on the rod. Each rod is pivoted axially (pivots not shown). Each is controlled by a stop tablet, through the mechanical linkage system shown. With the tab in the OFF position (face most nearly horizontal) the corresponding phenolic rod is so rotated that the key contact above it strikes the phenolic instead of the nichrome. In the ON position, the rod is rotated so that the nichrome wire is uppermost and is contacted by the key contact.

A supply of plus 310 volts is fed from the amplifier power supply to the nichrome wire on each of the rods through a 47,000-ohm resistor and a bronze spring which serves as a flexible element in view of the rod rotation. When the key is played each contact picks up this positive voltage from its stop rod (assuming the particular stop is

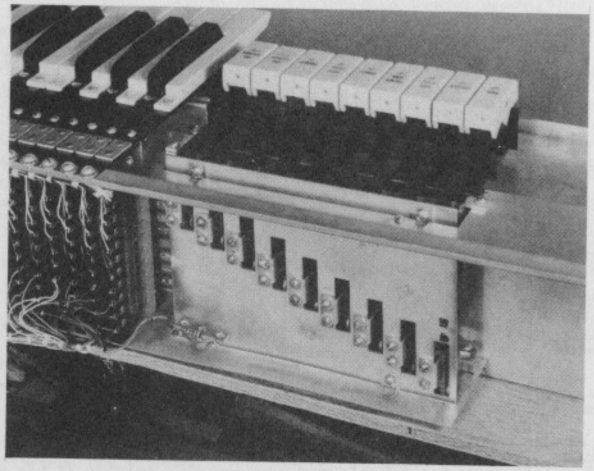


Fig. 11-10. Bass end of the upper-manual key and stop mechanism.

pulled) and feeds it through a time-delay filter to the appropriate reed pickup. If the d.c. were suddenly applied to the pickup there would be a distinct click. The filters allow the d.c. voltage to be built up gradually on the pickup, resulting in a smooth keying characteristic. The filters are printed circuits. There are 165 filters, each going to a pickup screw or strip. Seventy-three are for flute-tone pickup screws; 36 are for accompaniment pickup screws, which are similar to the flute pickups but are not so close to the reeds and are therefore softer; 43 are for trumpet pickup strips; and 13 are for pedal pickups, which are similar to the flute-tone pickups for the manuals.

All the reeds for the lowest octave of notes are connected together and through 9.1-megohm and 3,900-ohm resistors to ground. The series combination is the load across which audio is developed. From the top of the load resistor the signal goes through a 0.1-μf capacitor to the bass input of the amplifier. The remainder of the reeds are connected in common to another 9.1-megohm resistor which goes through the same 3,900-ohm resistor to ground to make up the treble load. The signal passes through another 0.1-μf capacitor to the amplifier treble input. A 50,000-ohm potentiometer with its arm connected to the top of the 3,900-ohm resistors provides 0-5.7 volts which polarizes the reeds slightly to prevent any signal when no keys are pressed.

TONE COLORING

The available variety of tone colors for both manuals is achieved through mixing stops, since actually there are only two varieties of

color – flute and trumpet. This scheme can be shown readably in diagram form only on a page several times the size of this one, so it has been tabulated and condensed into the chart of Fig. 11-11.

Let us first take the upper manual, which is equivalent to the swell on a usual organ. The keys are shown in pictorial form, numbered from F^{18} to C^{61}. C^{25} corresponds to middle C, 261.7 cycles. Notice the reed chart above the upper-manual chart. This shows the 73 reeds and their notes, beginning with C^{1} at 65.41 cycles to C^{73} at 4186 cycles. These notes are equivalent on the piano or on the frequency chart of Chapter 2 to C^{16} to C^{88}, but we shall use the Wurlitzer numbering to avoid confusion.

Now let us see what happens when we play the upper manual with the SOFT FLUTE 8′ stop pulled. Start by pressing Middle C, key 25, and find it on the manual picture in Fig. 11-11. Now note that a horizontal column of figures extends to the right from the SOFT FLUTE 8′ marking on the chart. Follow a line of boxes upward from Key 25 until it intersects the SOFT FLUTE 8′ line; you will find a box with the number 25. This indicates that when C^{25} is pressed with the SOFT FLUTE 8′ stop pulled, a pickup on reed No. 25 is energized.

Next we must know *which* pickup on that reed is energized. To find out we note that to the left of the SOFT FLUTE 8′ marking is a solid square box symbol. At the bottom of the chart is the legend which makes it clear that all pickups for this stop are the soft flute pickups – tone screw immediately over the reed, with greater distance (and therefore less effect) than the other similar screw on the reed which is for normal flute tone.

Thus, from the chart of Fig. 11-11 we know that under these circumstances we will get only a soft flute tone from the organ, at the pitch of middle C.

Now, leaving the SOFT FLUTE 8′ tab in the ON position, we also pull the FRENCH HORN 8′ tab. Again following the boxes up from key 25 to the FRENCH HORN 8′ line, we find the number 25, indicating the same reed. But at left of the FRENCH HORN 8′ title we see the black triangle, which, after reference to the legend at bottom, shows that a normal flute pickup screw has been energized. Now both screws on reed 25 are energized.

Now suppose we want to add some interest to the tone quality. Let us add to the former two stops the one called TONE COLORING $2\frac{2}{3}'$. Again following the boxes upward to the TONE COLORING $2\frac{2}{3}'$ intersection, we find the number 44 and referring to the reed chart we see that we have energized a normal flute pickup (triangle to left of TONE COLORING $2\frac{2}{3}'$ designation) on the reed supplying note G one and one-half octaves above C^{25}. This is approximately the third harmonic of middle C.

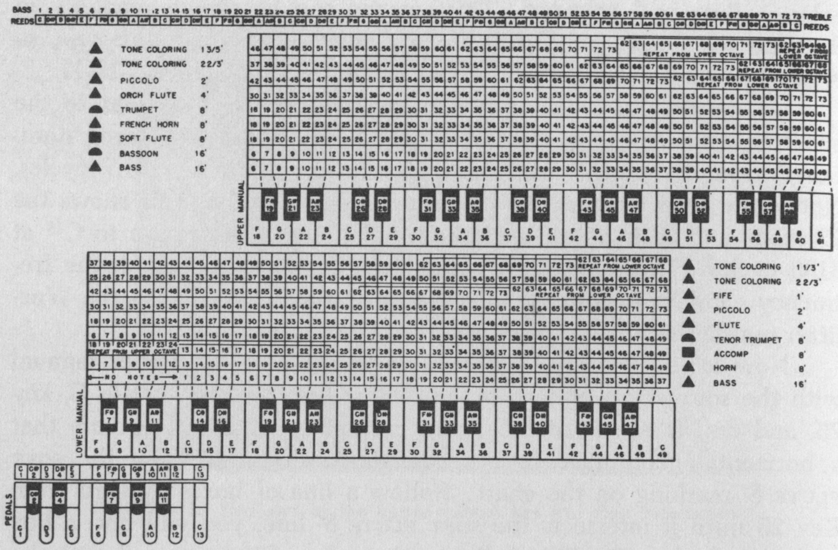


Fig. 11-11. Chart shows which tone screws are used for which keys and registers.

Let us add a fourth stop, TONE COLORING $1\frac{3}{5}'$. By the same process we find that we have added the normal flute pickup on reed 53, the E about two and one-quarter octaves above the middle C, approximately the fifth harmonic of middle C.

Thus we can see that pulling TONE COLORING $2\frac{2}{3}'$ adds the third harmonic to any notes played at 8′ pitch and pulling TONE COLORING $1\frac{3}{5}'$ adds the fifth harmonic. This is carried on as far as possible until the TONE COLORING stops run out of reeds, after which octaves are repeated, giving sub-third and sub-fifths. This scheme is simply a harmonic synthesis system somewhat similar in principle to the more complex Hammond system.

The TONE COLORING stops are not the only adjuncts of the harmonic synthesis system, taking care of only the fifth and third harmonics. The second harmonic, an octave above the 8′ note, is handled by the ORCHESTRAL FLUTE 4′ stop. When we play middle C with that stop pulled, we energize the flute pickup on reed 37, which is C an octave above middle C. The fourth harmonic is supplied by the PICCOLO 2′ stop, which energizes reed 49, two octaves above middle C. Thus the system gives us control over the fundamental, second, third, fourth, and fifth harmonics. Note that there is also a BASS 16′ stop which gives us in addition the sub-fundamental, one octave below the key struck. Naturally, these stops can be manipulated in many different ways to give different total tone colors.

The upper manual also has two trumpet-tone stops which have no relation to the harmonic synthesis system, though they work in the same way. The first is the TRUMPET 8′ stop. Assuming we have struck middle C, key 25, with the TRUMPET 8′ stop pulled, we find that we have energized a pickup on reed 25. From the black semicircle opposite the stop name and the legend below we see that it is the trumpet pickup – the strip of metal close to the end of the reed – which is energized. The second trumpet stop is the BASSOON 16′, which energizes the trumpet pickup on the reed one octave below the key struck. Though the trumpet stops are not related to the harmonic synthesis scheme they can, of course, be added to any combination of flute stops to produce still more combination.

The lower manual operates in the same way, except that there is no fifth harmonic, but a sixth instead, plus an eighth. The lineup is as follows:

Subfundamental – BASS 16′
Fundamental – HORN 8′
2nd Harmonic – FLUTE 4′
3rd Harmonic – TONE COLORING $2\frac{2}{3}'$
4th Harmonic – PICCOLO 2′
5th Harmonic – none
6th Harmonic – TONE COLORING $1\frac{1}{3}'$
8th Harmonic–FIFE 1′

In addition, the lower manual has a TENOR TRUMPET 8′ which uses the trumpet pickups and an ACCOMPANIMENT 8′ which uses the soft-flute pickups.

The keys of the lower manual energize many of the same pickups as those of the upper manual, and do not add anything. That is, if key 25 on the upper manual is pushed with the TRUMPET 8′ stop pulled, pushing key 25 on the lower manual with the TENOR TRUMPET 8′ stop pulled will simply energize the same pickup, but will produce no additional sound.

The pedals have no selection of stops, with a single tone quality obtained by making each one energize a separate pedal pickup screw on the lowest thirteen reeds. An additional time-constant network is used at each pedal contact to make the pedal tones speak and decay more slowly than those of the manuals.

Figure 11-12 shows the connection board on the reed chamber, to which all the keying leads go. This board on its inner side contains all the 165 printed-circuit time-constant filters. It should be noted that each keyed pickup has 160 volts d.c. on it (dropped from the original value of 310 volts by incidental action of filters. The first eight notes of the trumpet pickups have only 50 volts, brought about

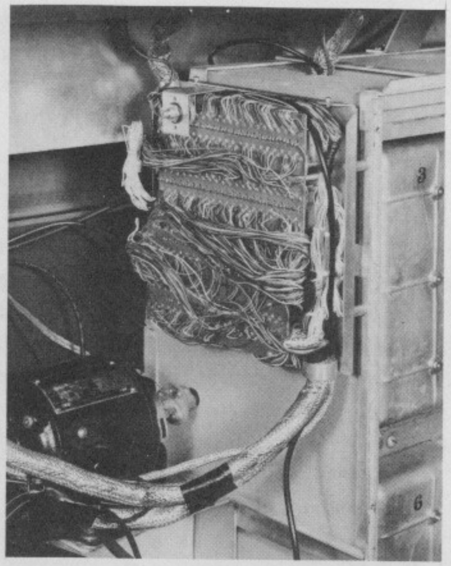


Fig. 11-12. Connection board with printed-circuit filters mounted on rear.

by special dropping resistors at the bass-end entrances of the corresponding stop rods and splitting of the nichrome wires. Figure 11-13 shows the rear of the console with the upper-manual key action picked up and brought into sight. Note the 12-inch speaker secured to the board beneath the manuals in front.

THE WURLITZER VIBRATO

The amplifier system of the organ is not particularly notable, but the vibrato circuit may well be called exciting by many people who have sought a practical way to introduce genuine vibrato – frequency shift – into systems where the original tone source must remain at a constant frequency. We may dismiss the amplifier after noting that one of the stages is used for gating; it normally has cutoff bias which is removed through a time-constant circuit by a series of paralleled contacts under the keys whenever any key is pressed. This eliminate noise of any kind in the absence of a signal.

The heart of the vibrato circuit (aside from the low-frequency oscillator) is shown in Fig. 11-14. The treble input from the reed pickups containing frequencies between 138.6 and 4186 cycles is fed to the grid of a 6SQ7 preamplifier in a standard cathode-biased circuit. (The bass signal is fed through a fairly similar stage directly through the main amplifier system without vibrato.) From the plate circuit of the 6SQ7 it goes through C_1 to the grid of V_2, half of a 6SN7-GT.

V_2 is a phase splitter of the "long-tailed" type, with one signal taken from the cathode circuit, across R_7, and the other from the

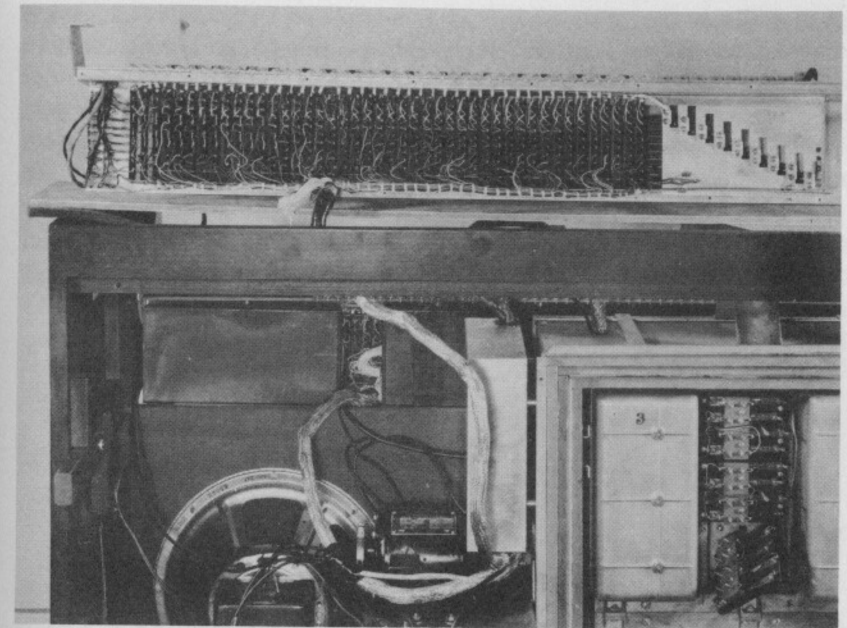


Fig. 11-13. Console rear. The upper-manual action has been raised to make it visible.

plate, the two signals 180 degrees apart in phase. At the plate the signal is divided into two parts, one part through C_3-R_8 and the other through C_4-R_9. The cathode signal is also divided in two paths, one through R_{10}-C_5 and the other through R_{12}-C_7. The plate and cathode signals in each leg are then combined, one set through C_6-R_{11}, and the other through C_8-R_{13}.

The entire purpose of the six legs enumerated above is to act as a phase-shift network, producing two signal outputs which have a constant phase difference of about 90 degrees. These two outputs appear at the points marked X. Their phase relationship to the original V_2 input signal changes, of course, with change in frequency. But they maintain a difference *between themselves* of about 90 degrees between about 500 and 15,000 cycles. It is appropriate to use the language of, for instance, phase modulation transmitters and call them quadrature voltages, for the vibrato of the Wurlitzer 44 is actually a phase-modulation system!

The two quadrature voltages are fed to the grids of V_3 through blocking capacitors C_9 and C_{10}. The two signals are mixed at the plate of V_3 and the mixed output is again a single signal taken from C_{11}.

Vibrato-frequency voltage at either 5.7 or 6.7 cycles is obtained from a low-frequency oscillator and phase inverter; it appears in push pull on the grids of V_3, as indicated in Fig. 11-14. It causes the two triodes to conduct singly. When the phase of the low-frequency signal makes the left grid positive and the right negative, the left triode conducts. When the phase is reversed, the left triode conducts. This causes a continuous change in the phase of the audio signal coming out of V_3 over approximately a 90-degree range.

When the left triode conducts, one signal of given phase goes through. When the right triode conducts the other quadrature signal comes through and there has been approximately a 90-degree phase shift. And the shift is smooth, for the signal emerging from the mixer is the vectorial sum of the two voltages at the plates of the two triodes.

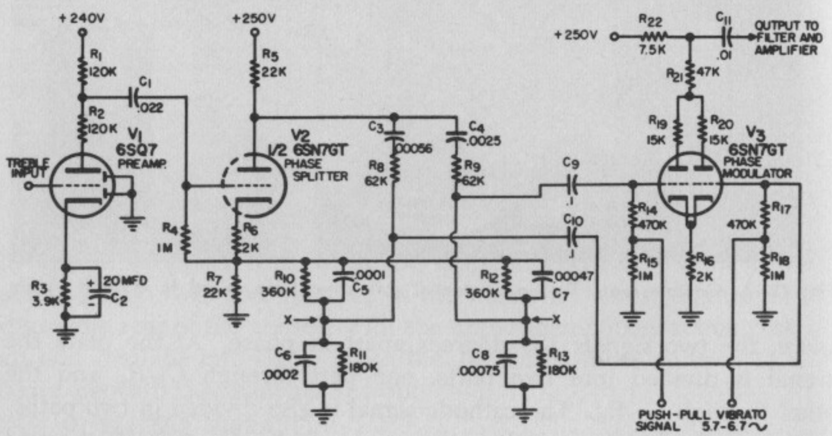


Fig. 11-14. The phase-shifter which produces vibrato.

If both are conducting equally, as is the case at the 0-, 180-, and 360-degree times of the low-frequency push-pull signal, the mixer output is 45 degrees away from either extreme. Thus the phase at any instant is dependent on the relative contributions of the two triodes, which is a function of their relative low-frequency grid signals at that instant. The latter vary in a sine manner, giving a phase swing that is smooth and natural.

Needless to say, phase modulation is equivalent to frequency modulation; it is, so to speak, a reciprocating electronic Doppler effect, as if the classic train blowing its whistle were to come toward and retreat from the listener again and again, with consequent apparent up-and-down change in the pitch of the whistle.

The output of the mixer is connected through C_{11} to a filtering stage which, by using frequency-selective negative feedback, cuts off sharply below 130 cycles so that none of the vibrato-frequency signal can affect the amplifier and speaker.

The oscillator is shown in Fig. 11-15; it is one-half of a 6SL7-GT operating as a standard phase-shift oscillator. The primary control is the two-circuit, three-point VIBRATO SPEED switch. In the FAST position R_2 is selected as the second resistor of the phase-shift network, determining the frequency at about 6.7 cycles. The second section of the switch connects the grid circuit of the other half of the 6SN7-GT, which is a phase splitter, to the arm of the VIBRATO DEPTH switch. The latter determines how much oscillator signal is sent to the phase splitter by tapping at one of three points on a voltage divider carrying oscillator output signal from the plate.

When the VIBRATO SPEED control is at SLOW, R_1 is selected for the phase-shift network; this makes the oscillator frequency about 5.7

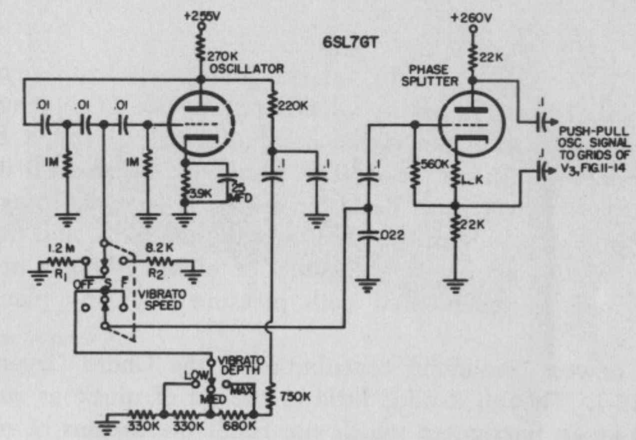


Fig. 11-15. This oscillator and phase-splitter provides switching voltage for vibrato circuit.

cycles. When it is at OFF, the output section of the switch disconnects the phase-splitter grid from the oscillator output.

Since the rate of phase swing and apparent music signal frequency swing depends on the oscillator frequency, the speed switch determines the vibrato rate in the music. And because the oscillator output determines how much total phase shift will occur the DEPTH switch determines how deep or wide the vibrato will be.

This vibrato circuit, it may be remarked, effectively does the same job as the Hammond vibrato scanner described in Chapter 5, but it does the job electrically, without moving parts, in a manner which can only be called elegant. Such a vibrato, with its ease of construction and compactness, does an almost impossible job which has puzzled many people who wished to add automatic vibrato to guitar amplifiers, amplified string instruments, and the like, only to be forced to settle for amplitude tremolo.

Chapter 12

The Hammond Chord Organ

THE Hammond Instrument Company has an interesting history which not only includes being among the first to popularize the nonacoustic musical instrument but also emphasizes the unconventional. The Hammond organ with its drawbars and the Solovox with its original idea of a melody instrument for use with the piano are the most outstanding items; and the Novachord, with its complex combinations of effects, though no longer manufactured, is remembered with pleasure by many players and listeners.

The newest Hammond contribution is the Chord Organ shown in Fig. 12-1. Though it adds little to the art of music as such, it is designed as an instrument which can fulfill the dreams of many innately musical people who have not had the opportunity to learn to play normal instruments. The Chord Organ is primarily for one-finger artists, and it gives them, with small practice, the ability to play full musical selections, complete with harmonies. To do this it resorts to more complexity than most electronic instruments (which are usually *compound* – having many similar circuits) but it is easy to understand and is so ingeniously designed that the Rube Goldberg aspect disappears after thorough examination.

WHAT THE CHORD ORGAN DOES

Examination of Fig. 12-1 shows that the organ has a 37-note key manual, a board at the left with 96 buttons, a row of control tablets above the manual and buttons, and a pair of pedals.

There are four divisions. The solo division operates in the same manner as a Solovox; it is suitable for one-note-at-a-time melody playing, using the keyboard. BASS, TENOR, and SOPRANO tablets control the

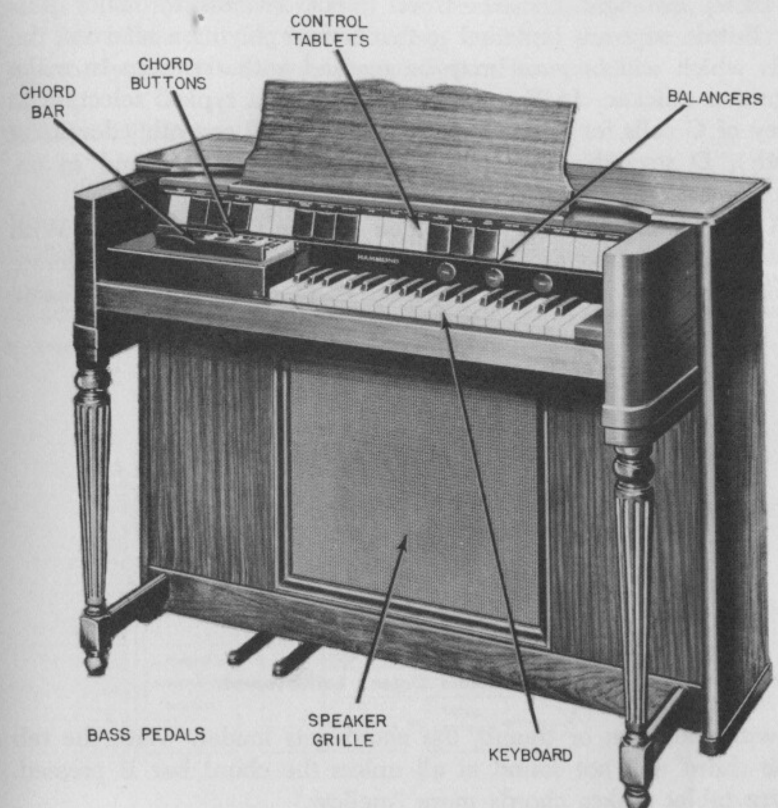


Fig. 12-1. The Hammond Chord Organ.

registers, somewhat as the 4′, 8′, and 16′ registers are selected in a normal organ. FAST ATTACK and ACCENT tablets give the solo tones a fast attack or a percussive quality. A WOODWINDS tablet gives a symmetrical waveform emphasizing odd harmonics. And DEEP TONE, FULL TONE, FIRST VOICE, SECOND VOICE, and BRILLIANT tablets give various tone colors.

The organ division, played on the same keyboard at the same time, gives polyphonic music – several notes simultaneously. String and flute tone tablets are provided to call forth either quality or both together. Thus, when the manual alone is played with several notes simultaneously, the organ division is heard on all notes and the solo division on the top note only.

The chord division is the main distinguishing feature. Fig. 12-2 is a drawing of the button board. When a single button is pressed, a chord sounds. There are 96 buttons; for each of the twelve musical keys there are eight chords – from minor seventh to major plus sixth. Button caps are provided so that before playing a selection the chords which will be used may be marked with the caps to make recognition quicker. In Fig. 12-2, for instance, a typical selection in the key of C calls for F major (subdominant), G seventh (dominant seventh), D seventh (dominant seventh of dominant), and so on. The caps are movable, of course.

A SUSTAIN-CANCEL tab is provided for the chord division. With the switch on, pushing a chord button brings in the chord at moderate volume; when the chord bar (see Fig. 12-1) is pressed at the same

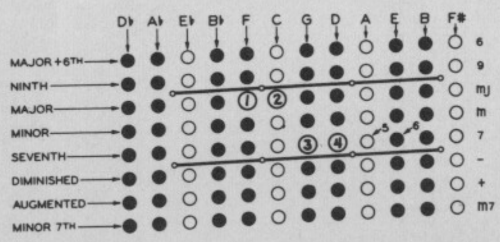


Fig. 12-2. The Chord Organ's button board.

time with the palm or thumb, the chord gets louder. With the tab off the chord will not sound at all unless the chord bar is pressed. A MUTE tablet makes chords more "mellow."

The pedal division consists of the two pedals and a FAST DECAY tab. When the left pedal is pressed, a low-octave tone similar to the bass note of the chord being played is heard; when the right pedal is played, the pedal tone is a fifth higher; the two give variety. The FAST ATTACK tab makes pedal tone disappear almost at once; without it, "the melody lingers on."

HOW THE ORGAN WORKS

The block diagram of Fig. 12-3 gives an idea of what is in the organ behind the panelling, though the diagram is very much simplified. There are three separate generating systems, all using vacuum-tube oscillators. The solo and organ generators are controlled by the keyboard, after which the selected tones go to the tab controls, thence to a pair of volume controls called balancers, the amplifier, expression control, and built-in speaker.

The chord generators are controlled by the chord button, the chord bar, and the pedals, as well as by the tabs. Chord-button

tones have no balancer as they are fixed in relative level; pedal tones can be balanced. In the following we shall describe the divisions separately, which will clear the cobwebs of complexity.

SOLO DIVISION

The solo division of the Chord Organ is really a complete Solovox. It is shown schematically in Fig. 12-4. V_1-C_2 is the oscillator, which is tuned over the range from F-349.2 cycles to F-2794 cycles. The tuning is done by the 37 tuning inductors, which are connected in series between the grid of V_1 and ground. When a key is pressed, a corresponding key contact connected between two of the series

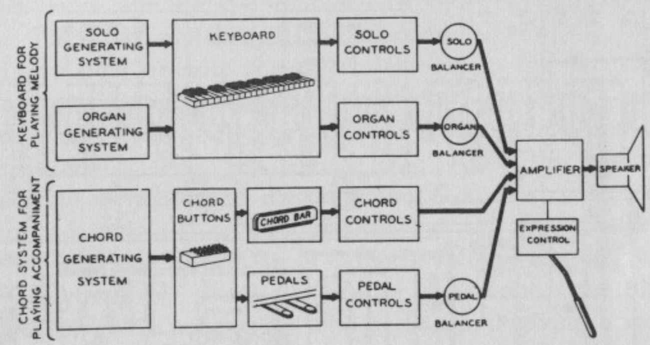


Fig. 12-3. Block diagram of the Chord Organ.

coils shorts the junction to the solo tuning bar, which is grounded. This reduces the net inductance between V_1 grid and ground, raising the oscillator frequency to that of the key. The lowest F has no tuning contact since with all the inductors working the oscillator tunes to the low F. C_{84} is the main tuning capacitor, while the two groups C_1 to C_5 and C_7 to C_{12} are for coarse and fine tuning respectively. They tune the entire range, of course, not the individual notes. The latter may be tuned by sliding the cores in and out of the inductors.

The oscillator rectifier V_{3a} creates the sharp positive pulse necessary for the driver, the second triode of V_3. The latter drives an aperiodic flip-flop circuit which reverses its condition once per pulse – once per oscillator cycle. Output taken from one side of the flip-flop is an almost square wave of frequency half that of the oscillator, or one octave below. The output of the first flip-flop frequency divider is taken from pin 1 of V_4 through R_{20}.

A second driver and flip-flop divider is driven by output from the first, so that for each oscillator tone, there are three octavely related frequencies made available.

For each frequency there are two tones, woodwind tone and a complex one. The highest-tone woodwind output is the cathode of oscillator V_1, a sine wave. The next two are square waves from the plates of one triode of each divider. The complex tones are taken from the plates of the rectifiers and driver stages. Waveforms are shown in Fig. 12-4.

The six outputs pass through modifying networks of resistors and capacitors to take some of the "edge" off the tones and make them

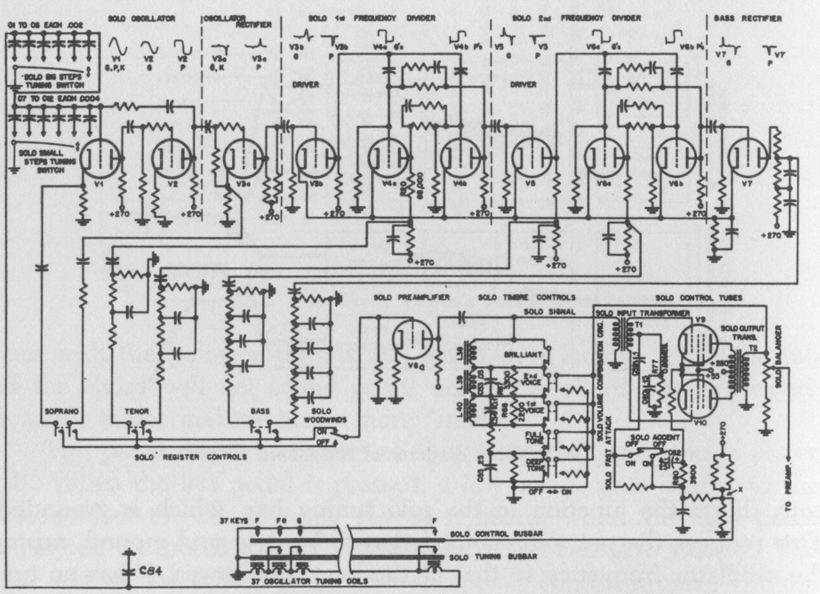


Fig. 12-4. Partial schematic of the solo division.

more pleasing. The BASS, TENOR, and SOPRANO switches – double-pole, double-throw – switch the three registers as desired to the WOODWIND switch. With the latter on, the grid of solo preamplifier V_{8a} is connected to the woodwind modifying networks; when the tab switch is off, the more complex tone goes to the preamplifier grid.

The preamplifier plate circuit feeds a tone-color network containing five sections in series between the plate line and ground. Each section is normally shorted by a tab switch when the switch is in the nominal off position – contacts closed on one side. When, for example, the 1ST VOICE tab is placed in the on position, the parallel combination of L_{40}, C_{54}, and R_{68} is placed across the signal line. This gives the tone a peak near 750 cycles, imparting to it a horn-like quality. The 2ND VOICE section peaks at around 1,000 cycles. DEEP TONE places a capacitor across the line to cut highs and make the tone more "mellow," while FULL TONE has only a resistor and gives flat response. BRILLIANT shunts the line with an inductor, reducing bass to give a rather piercing quality.

The solo control stage V_9-V_{10} exists to allow control of the tonal attack. Normally the cathodes are at about plus 65 volts, obtained by voltage division from the 270-volt point shown in the diagram. This cuts off the stage. When any key is pressed a solo control line connected to point X is shorted to ground by the solo control busbar under the keys and the key contact. This shorts the bias voltage to ground. With the switches in the positions shown, C_{58} makes the attack fairly slow because a sudden decrease in the cathode voltage causes a negative surge through the capacitor, charges C_{60} negatively, and moves the grid in the negative direction, which remains until the charge on C_{60} leaks off through R_{77}. When the fast attack tab is operated its switch opens, disconnecting C_{58}. With the SOLO ACCENT switch on not only is C_{58} disconnected but C_{62} is connected across R_{80}. For the sudden decrease in cathode voltage caused by pressing a key, C_{62} effectively shorts the resistor and reduces the bias for an instant, causing the note to be loud at first and giving a rather percussive effect.

Output from the solo division is controlled by a balancer potentiometer, the arm of which goes to the main organ preamplifier in common with outputs of the other divisions.

ORGAN DIVISION

A simplified circuit of the organ division is shown in Fig. 12-5. The generator system for the 37 tones consists of sixteen L-C oscillators, each of which can be tuned to either of two frequencies, except for the lower four and upper one, which can be tuned to three frequencies.

The tuning is done automatically when a key is pressed. Normally the lowest oscillator, for instance, the first triode of V_{12}, is tuned to the frequency of low G by C_{87} across the tuning inductor. When the F# key is pressed a contact connected to the lower end of C_{90} strikes the upper busbar, connecting C_{89}-R_{117}-C_{90} across half the coil and lowering the frequency. When the F key is pressed a contact connects only C_{89} across the lower half of the coil, lowering the frequency to F.

The organ oscillators are normally not operating because they have no plate voltage. Whenever a key is pressed, a contact in the lower row strikes the lower busbar. This carries the B-supply of 280 volts to the plate of the corresponding oscillator through a simple *R-C* network which softens the attack somewhat.

Each organ oscillator has two outputs, one from the upper end of the tuned circuit giving a sine-wave or flute tone, and the other from the lower end giving a waveform like that shown, known as the string tone. All similar outputs of all oscillators are paralleled and brought through tab switches and tone modifying networks to the organ division balancer, the arm of which goes to the common pre-amplifier grid.

It should be noted that four busbars run under all the keys. Two are shown in Fig. 12-5 and two in Fig. 12-4. The idea of using one os-

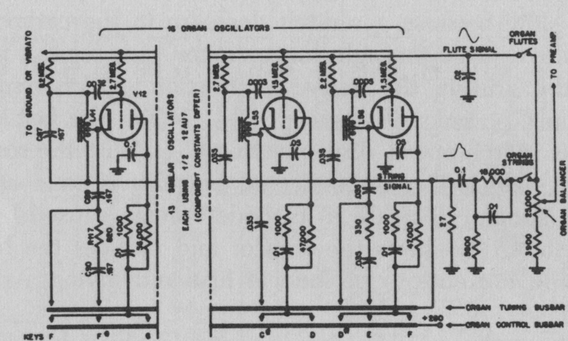


Fig. 12-5. Partial circuit of the organ division.

cillator for two or three notes is that not too much music calls for simultaneous playing of two adjacent notes, especially the fairly simple music which a typical Chord Organ player would probably use. The frequencies covered by the organ oscillators are 174.6 to 1397, the F just below middle C to that three octaves higher.

CHORD AND PEDAL DIVISIONS

Figure 12-6 shows the pedal and chord divisions, as well as the amplifier section, which is simple.

There are six chord oscillators, each tunable to two frequencies, making a full octave of twelve tones available. No chord has more than four notes and all use tones between F-174.6 and E-329.6 cycles. The chord oscillators also supply the tones which pass through frequency dividers and give the pedal notes.

Each oscillator has four contacts, one associated with each of four busbars. When a contact is touched to the upper busbar the oscillator is moved down one half tone by connecting a capacitor from the tap on the tuning inductor to the grounded bar. When a contact is touched to the second bar the oscillator plate output is connected to that bar. This second busbar carries the chord output signals.

The lower two busbars carry pedal output signals taken from the oscillators in the same way.

The chord actuating system is a mechanical assembly which cannot well be shown here. Each of the 96 buttons selects the right three or four notes for the chord and the correct two notes for the pedals. The notes are predetermined by the positions of small pro-

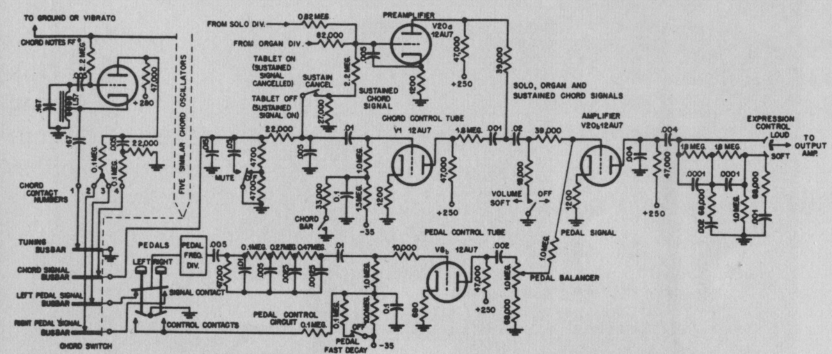


Fig. 12-6. Pedal and chord divisions and amplifier.

jections on 96 pivoted levers underneath the buttons. The projections press actuators to operate the required contact springs.

Let us take an example, say the button which creates the C-major chord consisting of C-E-G. When the button is pressed one lever projection actuates the contacts connecting the B-C oscillator to the chord signal busbar. (This oscillator need not be tuned since it is normally running at the frequency of C.) A second projection connects the D#-E oscillator output to the chord signal busbar. A third tunes the G-G# oscillator to G by closing the tuning contact for that oscillator. A fourth closes the contacts carrying G-G# oscillator output to the chord signal bar. This completes the formation of the chord.

In addition a fifth lever projection connects a B-C oscillator output contact to the left pedal signal busbar and a sixth connects the G-G# oscillator output to the right pedal signal bar.

The chord output signals from the chord signal busbar go to the chord control tube, shunted on the way by the mute switch which

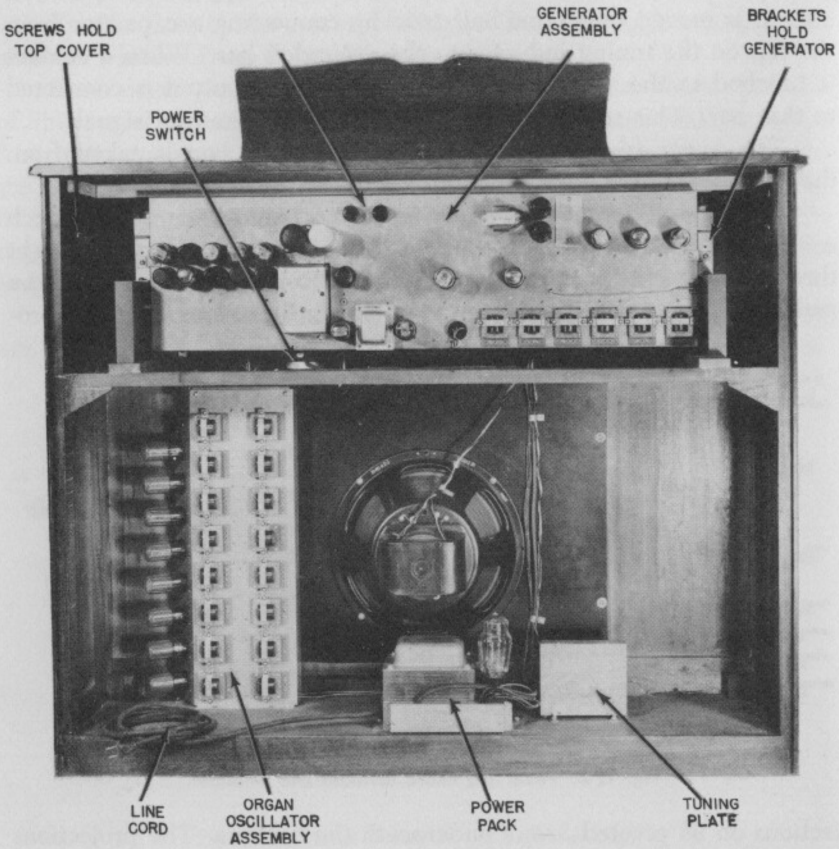


Fig. 12-7. Rear view shows how parts are mounted.

places a capacitor across the line to produce more "mellow" quality when desired. The tube, V_1, normally has 35 volts of negative bias on the grid from a fixed source, cutting off plate current. When the chord bar is pressed the bias disappears, allowing the chords to come through.

Chord signals also go through the SUSTAIN CANCEL switch to the input of the preamplifier, pin 7 of V_{20a}, to the same point reached by the outputs of the solo and organ divisions. The preamplifier output goes to the second half, V_{20a} bypassing the chord control tube. Thus when the SUSTAIN CANCEL switch is closed a reduced-level chord signal comes through even though the chord bar may not be pressed.

The outputs of the two pedal signal busbars go to the two pedals, which are mechanically interlocked. Considering only the signal con-

tacts of the pedals, for the moment, output from whichever pedal is pressed goes to the input of a two-stage frequency divider exactly similar to those used in the solo division. In this way pedal tones two octaves below the chord tones are produced. The output of the frequency dividers goes through an *R-C* tone modifying network to the grid of the pedal control tube, pin 2 of V_{8b}. The grid is normally at 35 volts negative. When the pedal is pushed, a control contact on it removes the tube bias, allowing the tone to come through. The mechanical interlocking of the pedal signal contacts is such that when the pedal is released the last signal contact made is maintained. This keeps the tone going while the bias on the pedal control tube slowly returns through the time-constant network and the tone dies away. The pedal fast attack switch modifies the time constant to make for faster attack and decay.

The output of the pedal control tube joins all the other signals at the grid of amplifier V_{20b}, pin 2, after passing through a pedal balancer potentiometer. The VOLUME SOFT switch in this grid circuit simply shunts the line to ground through the 18,000-ohm resistor, reducing the volume of the entire instrument.

The output of the amplifier triode goes to the expression control.

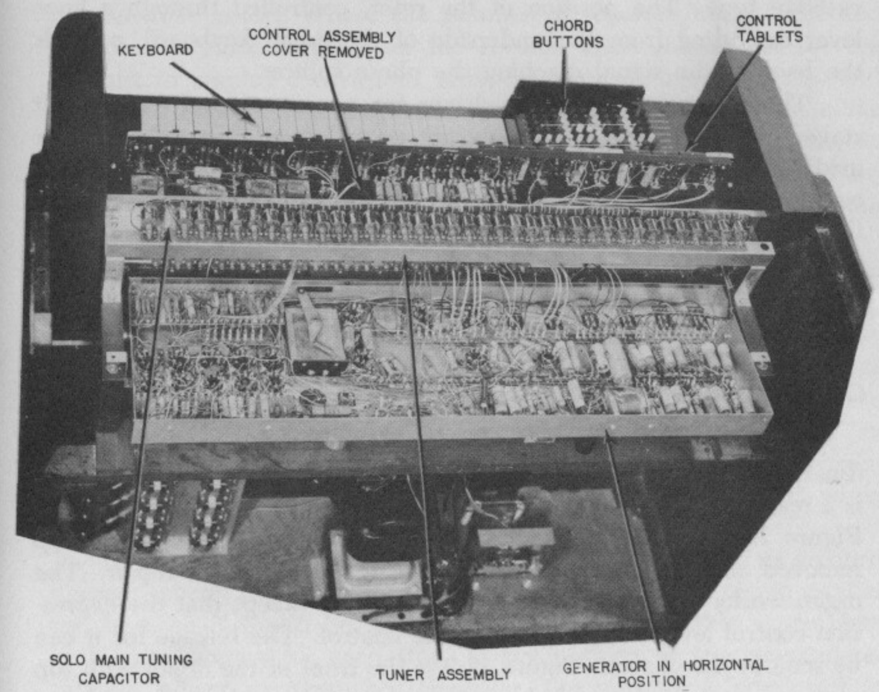


Fig. 12-8. Photo from rear, with top removed and upper chassis swung down.

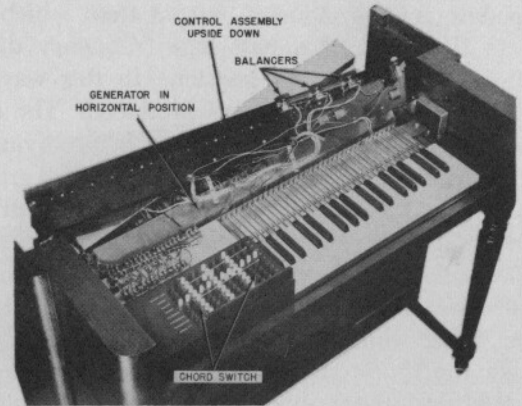


Fig. 12-9. Front of the organ. Top is off and the switch board is upside down.

This control is a special variable air capacitor with two stator plates. One stator is connected directly to the amplifier plate through a blocking capacitor .004-μf. The other stator plate is connected to the amplifier plate through a tone-compensated attenuation network. The rotor is connected to the No. 2 grid of a phase splitter of the common-cathode type. The position of the rotor, controlled through a knee lever extending from the underside of the organ keyboard, controls the level of the signal reaching the phase splitter.

The phase splitter is the driver for a conventional 6V6 output stage. Radio-phonograph inputs are provided at the phase splitter grid. The amplifier has a negative-feedback connection from output transformer to phase-splitter grid, the frequency characteristic of which can be adjusted somewhat by a variable capacitor. The vibrato of the Chord Organ is similar to that in the Solovox – a phase-shift oscillator and switching tube connected to the grids of the oscillators through a switch which grounds the grids when vibrato is not desired.

CONSTRUCTION

Despite the apparent complexity of the Chord Organ it is extraordinarily compact. Figure 12-1 shows the main controls. Figure 12-7 is a rear view showing how the electronisms are mounted in the case. Figure 12-8 is taken from above the rear of the organ with the top removed and the upper chassis swung down for test or repair. The organ works with the chassis in this position, except that the expression control lever will not actuate the control. The linkage for it can be seen in the chassis. Figure 12-9 is the front of the organ with top removed, and the board holding the tab switches taken off and lying upside down.

Chapter 13

The Allen Organ

The Allen electronic organ, made by the Allen Organ Co., Inc., Macungie, Pa., is simple in principle, yet it is among the most interesting, both because Allen organs are among the largest electronic instruments manufactured today and they are among the closest analogies to the pipe organ in certain respects.

A pipe organ consists of several ranks of pipes, each with a distinctive tone color. Where the number of separate ranks is limited by space or cost, controls on the console make it possible for the organist to employ the various ranks on more than one console and to employ each rank in more than one register. The organ is unlike other instruments (except possibly the piano, harpsichord, etc.) in that each note of each tone quality is created by a separate "instrument" or pipe which can be voiced to have an appropriate volume level and harmonic content.

The Allen organ at its best and most elaborate does exactly this. It can have a separate vacuum-tube oscillator for each tone quality, so that the oscillators are simply substitutes for pipes, one by one. In such an instrument, playing a single note with, say, three stops pulled will cause three separate oscillators to sound. Not only does the hearer receive the desired sound separately of voiced "pipes" in such a case; the fact that the oscillators are not "locked" or synchronized gives a chorus or choir effect which cannot be obtained when all notes are in locked phase relationship.

In practice, few Allen organs employ this possibility to its greatest degree; several qualities are ordinarily obtained from a single oscillator, because it would be extremely expensive to provide a complete rank of oscillators for each stop, to say nothing of the tremendous primary power requirement. However, the practice is employed in greater or less degree throughout the Allen line. This is possible

Fig. 13-1. One of the largest Allen consoles.

because there is no such thing as a standard model, though the lower-priced organs are standardized in some degree and a home entertainment model is now being marketed with no alterations possible. Organs are sold by Allen on an individual design basis in the same manner as pipe organs, where the builder visits the location, discusses requirements with the organist and others, and then all come to an agreement on the specification. The number of stops, manuals, couplers, and so on thus varies from organ to organ.

Figure 13-1 shows the console of one of the larger Allen organs, costing, as is often the case, several times the price of any other American electronic organ because of its greater resources and complexity. This model is played like and sounds very similar to a unified organ of fair size. Fig. 13-2 shows in contrast one of the smallest Allen models, a two-manual organ with a small number of stops and couplers, a single swell shoe, a two-octave pedal clavier, and no combination "pistons." An organ such as this latter may have a single rank of oscillators for each manual, each oscillator having two to four tone-

color filters, and some unification couplers to add flexibility by making the oscillators available at 16-, 4-, and possibly 2-foot pitches in addition to the 8-foot unison. Such an instrument, except for its lack of chorus effect on a single manual, is equivalent to a pipe organ with a number of ranks equal to the number of filters associated with each note plus unification and duplexing.

TONE GENERATION

Figure 13-3 is a schematic diagram of an oscillator-filter circuit found in typical Allen organs. There is one such circuit for each note of each manual and for each pedal. One triode of a 6SN7 is used for each note, so that a single tube takes care of two notes.

The oscillator itself is a Hartley circuit with plate grounded through the power-supply filter capacitor (not shown). Output is taken from the junction of the grid-bias network C_5-R_5 and the tuned circuit L_1-C_1-C_2-C_3. Two of these capacitors are within the chassis, while the third is connected across a pair of thumbnut terminals

Fig. 13-2. A small Allen model.

on the front panel of the chassis. These capacitors can be seen in Fig. 13-4, which is a rear view of an Allen model with self-contained generators. The oscillator is tunable over a range of about a semitone with R_6, but if it gets out of tune by more than a semitone, the outside capacitor must be replaced to bring frequency back within the tunable range. A tuning kit which contains a variety of capacitors for this purpose is made available to service personnel.

The narrow tuning range is due to the desire to avoid variable capacitors, since they contribute to instability on account of temperature and moisture changes. The range of adjustment made possible by R_6 is kept low to avoid reducing the Q of the tuned circuit too much. The oscillator is inherently extremely stable, however, and in

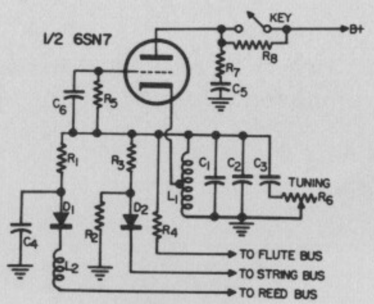


Fig. 13-3. A typical Allen oscillator-filter circuit.

the normal location yearly tuning usually suffices. The tuning inductor L_1 is a universal-wound air-core coil, an unusual practice in audio and one which makes for very high Q.

Referring again to Fig. 13-3, the oscillators are normally nonoperative, though a small plate voltage is maintained, current passing through large-resistance R_8, to increase emissive life. When a key is pressed B-plus voltage is applied to the plate. R_7 and C_5 make up a time-constant network which cause the plate voltage to build up in a finite time interval rather than instantly, so that the oscillator output increases gradually from zero, giving a soft tone envelope to imitate the gradual speech of pipes. The decay envelope is also gradual because of the time taken by C_5 to discharge.

The oscillator output is a sine wave before modification by the filter circuits. The sine wave, attenuated as desired by R_4, is connected to a flute bus, to which the outputs of all other oscillators in the same rank are also connected.

The sine-wave tone is also passed through R_3 and copper-oxide

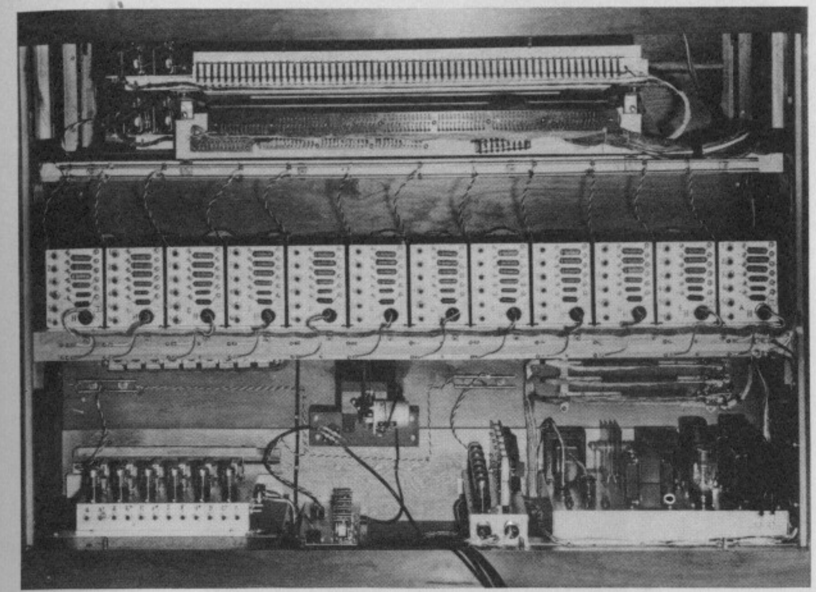


Fig. 13-4. Rear view of the self-contained organ.

diode D_2. The diode rectifies the sine wave, producing a half-sine-wave, which is rich in harmonics. This simulates string tone and it is passed to a string bus, which, like the flute bus, is common to all oscillators in the rank.

A third attenuating resistor R_1 and distorting diode D_1 is used to produce a reed tone. The reed tone is also rich in harmonics but it has a pronounced formant in a particular part of the spectrum in which harmonics are prominent. This formant is produced by C_4 and L_2, one each of which is provided for every note.

The reader may now see the idea behind the Allen scheme of tone production and begin to relate it to those described in Chapters 6 and 8 for the Baldwin organ and the Connsonata. In the Baldwin, all generated tones are rich in harmonics and all are passed through a single filter. In the Connsonata each generated tone is available both pure and with harmonics and, again, a single filter produces each tone color. In both of these cases the single filter attenuates or emphasizes fundamentals as well as harmonics. If, for example, it is a string-type filter which emphasizes upper harmonics, it also attenuates lower notes so that notes at the lower registers are not so strong as a whole as those above. This is what actually happens in an orchestral instrument, but in a pipe organ each pipe can be voiced

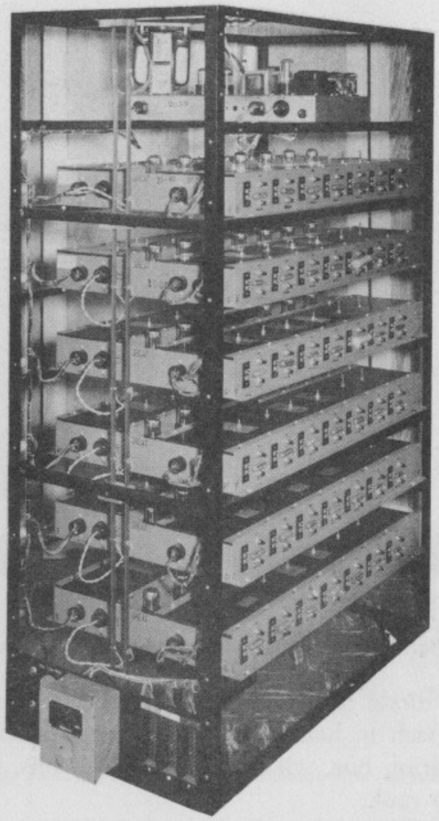


Fig. 13-5. An Allen generator rack.

so that its total loudness is appropriate and is properly related to the rest of the scale, while its harmonic structure is still controlled to give the desired tone color.

In the Allen a similar procedure is made possible by the inclusion of individual filters for each note. Using a string tone again as an example, if the note concerned is a low one and the necessary prominent harmonic structure makes the total loudness wrong for smooth scaling, R_3 (Fig. 13-3) is simply selected for the correct total output of that note in that tone color. This is even more important with strong-formant qualities such as reed, where the tuned filter may cause notes with fundamentals far removed from the formant frequency to be weak. R_1 can simply be adjusted for the correct level, and it will vary from note to note. While the procedure employed in the other instruments results in a very acceptable compromise,

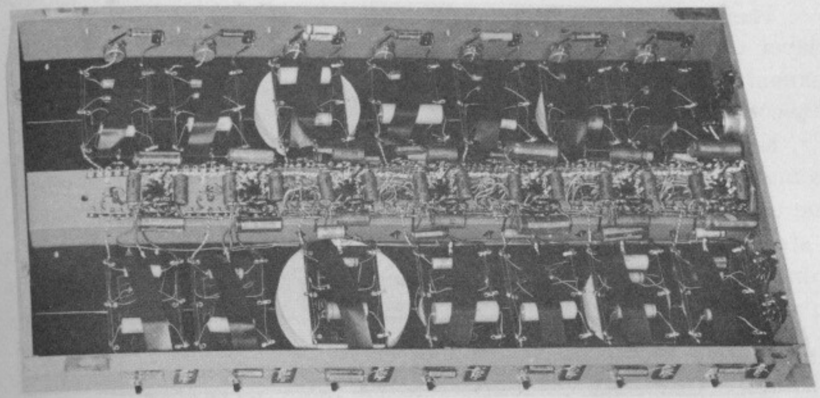


Fig. 13-6. Underside of a generator chassis.

therefore, and is, in engineerese, more "elegant," the Allen scheme is more closely analogous to pipe-organ construction. It is consequently, of course, a great deal more expensive and space-consuming. And to provide the wide range of tone colors obtained in the Baldwin, for instance, by the Allen method would be so expensive and bulky as to be almost impractical.

While some Allen models do have the electronism contained within the console, most employ separate racks connected to the console by cables. Such a rack is shown in Fig. 13-5, containing six generator chassis, each generating seven octaves of two notes, for a total of 85 notes, including the extra top C. A power supply and tone changer (see below) is at the top of the rack. This particular rack contains the generator for one rank for the great manual. An organ with minimum specification would have another similar rack for the swell and pedal generators. More elaborate organs have several such racks for each manual containing an equal number of ranks of generators. Different ranks will have, of course, different output filters and busses, though all filters are built on the same lines—diode plus tuned circuit, simple resistance, or high- and/or low-pass filters. The "stopped" qualities, incidentally, are produced by employing two diodes in parallel and in polarity opposition. These qualities, required for clarinet, stopped diapason, stopped or doppel flute, and other woodwinds, are characterized by a symmetrical waveshape containing almost exclusively odd harmonics. This is the same effect produced in the Baldwin by the outphaser circuit. Figure 13-6 shows the underside of one of the generator chassis used in the racks.

THE REGISTRATION SYSTEM

The registration system of the Allen — the method by which the player can select the desired tone colors, pitch registers, and intermanual couplings — is found in the tone-changer chassis (plural) and key-contact couplers.

Keep in mind that each rank of generators supplies output from as many signal busses as there are types of filters. The rank of which one oscillator is illustrated in Fig. 13-3, for instance, has three busses — string, flute, and reed — and therefore three outputs, each taken from one bus. The string bus is connected on the tone-changer chassis to a preamplifier such as that diagrammed in Fig. 13-7. The reed bus is connected to another similar preamplifier, and the flute bus is connected to a third, similar except that the input is direct instead of

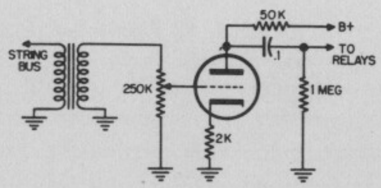


Fig. 13-7. The string preamplifier.

transformer-coupled. The transformer primaries have a very low impedance, perhaps 5 ohms, while the grid resistor used with the flute preamplifier is 250,000 ohms.

The input to each preamplifier is controlled by a potentiometer so that a certain amount of voicing is possible to adjust to the characteristics of the auditorium. The outputs of the preamplifiers go to relay contacts on the tone-changer chassis. These relays are controlled from the console with the stop tablets. The tabs close relays in various combinations to channel selected mixtures of the basic tone qualities to the power amplifier. Resistors are used between preamplifiers and certain relay contacts so that mixtures can contain the components at various levels. A tone-changer chassis appears in Fig. 13-8.

A typical small Allen organ may therefore have, with the three basic ranks of generators and with three tone qualities available from each, the approximate equivalent of a pipe organ with nine ranks of pipes. This is, of course, a very small organ, but the maker feels that the equivalent of a few ranks, with the ranks voiced in the same way as organ pipes and the organ provided with sufficient coupling facilities is preferable to the much larger array of tonal

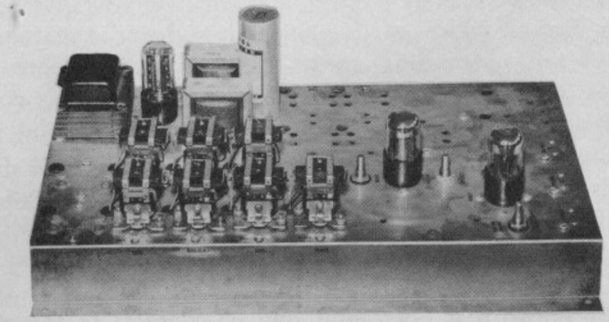


Fig. 13-8. A tone-changer chassis.

qualities obtainable by systems like the Baldwin and Connsonata which do not allow such voicing. There is no doubt (and the writer has confirmed this personally) that the Allen organ does give a very fine scale. Whether this advantage is preferable to the greater range of individual colors obtainable in the other instruments is a matter of personal taste.

An additional factor to be considered is the pitch registration inherent in the simpler Allen instruments (and also in the Connsonata). Figure 13-9 is a photograph of one end of a playing manual and the organ-type coupler arrangement. In speaking of this, let us, for clarity, decide that the horizontal direction is the player's right-left, while vertical is from front to rear of the instrument.

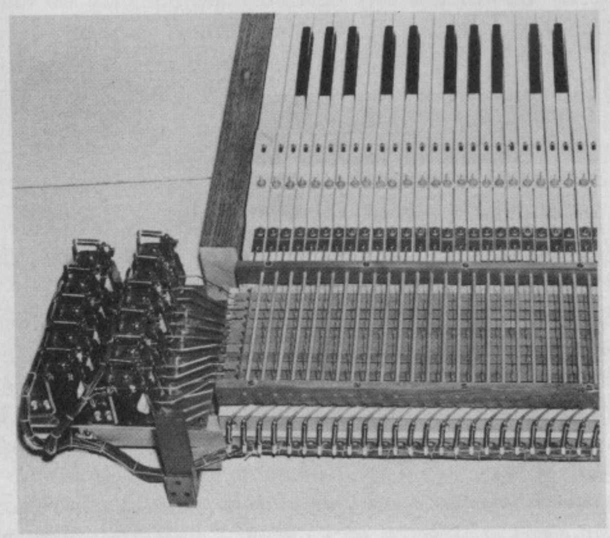


Fig. 13-9. End of a manual showing coupler solenoids.

In the coupler system there is a vertical rod for each key; this rod carries B-plus. Associated with each rod and key is a vertical series of eleven upstanding spring-wire contacts. There may be more or less than eleven in any particular organ. Each contact in, say, the first horizontal row is wired to the plate of the oscillator corresponding to the pitch denoted by the associated key. All contacts in this first horizontal row pass through small holes in a horizontal strip of nonconducting material, the right end of which is connected to the actuator of a solenoid relay. When the relay is energized, the entire nonconducting strip moves somewhat to the right, bending the entire first horizontal row of contacts to the right and bringing each one close to (but not touching) the vertical bar which carries B-plus. When a key is pressed, a lever rotates this bar (which is not round in cross-section but more or less flat) so that it moves toward the contacts. Because the contacts in the first horizontal row have been pulled close to the bars, pressing any key will make the bar associated with that key contact one of the first-row spring contacts. Thus pressing any key will energize the oscillator of pitch corresponding to that key and the 8-foot register will sound.

The second horizontal row of contacts may be wired to the plates of the oscillators an octave higher than the associated keys. When the second solenoid is actuated, these second-row contacts are pulled close to the bars, and pressing a key will sound the oscillator an octave higher. The third horizontal row of contacts may be connected to oscillators an octave lower than the keys for 16-foot pitch, and so on. By this means any desired coupling may be had, so that the tone qualities selected by the stop tablets may be played in the 16-, $5\frac{1}{3}$-, 4-, $2\frac{2}{3}$-, 2-, $1\frac{3}{5}$-, $1\frac{1}{3}$-, and 1-foot registers. Depending on the number of contact wires and solenoid-actuated strips, intermanual coupling is also possible, so that pressing a key on one manual may energize the oscillators of another manual in any of several pitch registers.

This process of unification and duplexing is probably essential to any organ with few ranks of pipes or, in this case, oscillators. Organ authorities differ to some extent as to whether it is a desirable feature. As a practical matter, note that it is not possible, for example, to set up a combination on one manual consisting of a string stop at 8-foot pitch plus a very soft flute at 4-foot. In an instrument of the Allen type, all voices will sound at all pitches which are coupled in, so that pulling a string and flute will result in both voices at both 8- and 4-foot pitches.

The limitations discussed above and the necessity for extensive use of unification for flexibility exist to a much lesser degree, of course, in the more elaborate versions of the Allen organ. The com-

pany sometimes makes models selling at tens of thousands of dollars which contain respectable numbers of ranks of oscillators, different ranks giving different tone colors and pitch registers. Extra ranks are also sometimes added, tuned slightly sharp to give the beating celeste effect. If enough ranks are used (and enough separated speakers) both quality and musical integrity closely resemble their pipe-organ counterparts. In theory there is little to limit the closeness of resemblance to the finest organs, provided enough generators are used and correctly voiced. Of course, certain pipe-organ qualities are almost unattainable: the partly percussive attack of a few stops and the random unsteadiness of pitch which contributes so much to the vibrancy of organ tone and which caused a famous authority to say, "No good pipe organ is ever quite in tune."

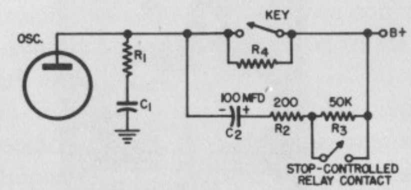


Fig. 13-10. The harp circuit.

One other voice is of interest, the harp effect. This is a purely electronic circuit capable of imitating the percussive tones included in some organs—celesta, harp, bells, etc. The circuitry is shown in Fig. 13-10. The oscillator is one of the regular ones used for other tones. An additional time constant circuit is connected between plate and B-plus, including C_2, R_2, and R_3. A relay contact (one per key) across R_3 is normally open, so that the additional circuit is not effective because of the large value of R_3. When the relay is closed (harp effect in operation), and the playing key is closed, plate voltage is applied to the oscillator tube with no more delay than usual because the key almost directly shorts C_2 and discharges it very quickly. This is the percussive attack. When the key is opened, a very long time is required for C_2 to charge up to the voltage equal to the difference between the supply voltage on its positive plate and the low plate voltage at its negative end. During the discharge period, therefore, it allows tube plate voltage to die away very gradually, so that the very slow decay characteristic of harp-type tones is produced. The harp circuit is practical because of the extreme frequency stability of the oscillators over a wide range of plate voltage. The circuits which differentiate among the various types of percussive tone are simply diode-R-C filters similar to those used to obtain other tone qualities at the oscillator outputs.

GYROPHONIC PROJECTOR

An interesting method is used in all Allen organs to obtain vibrato and also some semblance of chorus effect over and above what is caused by the use of more than one generator string. This is the Gyrophonic Projector pictured in Fig. 13-11.

In the unit shown, two bass and two treble speakers are mounted on a circular piece of wood. The entire assembly is mounted on a shaft and caused to rotate by belt drive from a d.c. motor energized

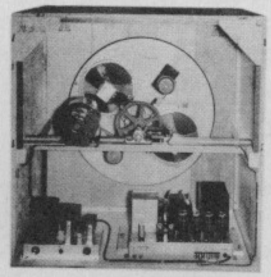


Fig. 13-11. Inside the Gyrophonic Projector.

by a dry-rectifier supply which can be seen in the lower left corner of the enclosure. The power amplifier is at right. The amplifier and speakers are connected by a pair of slip rings, each having two brushes.

When the speaker assembly rotates at vibrato speed, the hearer receives the effect of phase changes because of the doppler effect caused by changes in distance between his ears and the source. The vibrato is very pleasing, though whether it is superior to the usual type caused by frequency shift of the oscillators is open to question.

A more profound and important effect of the Gyrophonic system is that when vibrato is not desired the speakers are rotated at a very slow rate. This gives a constant slow phase-change effect which, while not really identifiable, gives a wandering, slightly varying effect to the tone which is very similar to what is obtained with more ranks of separate tone generators in a pipe organ and also gives a slight indefiniteness of pitch such as might be caused by the random variations of a pipe-organ air supply. The slowly moving speakers tend to mitigate the "electronic" sound which otherwise results from perfect, motionless sound-source tuning, and are a distinct contribution to organ-like tone. Tablets on the console enable the organist to vary the speed of rotation for fast and slow vibrato or the slow nonvibrato.

Chapter 14

The Stroboconn

WHILE this book is concerned with electronic musical instruments, the Stroboconn (C. G. Conn, Ltd., Elkhart, Ind.) is so closely allied with electronic music and so useful with electronic organs as well as more conventional acoustic instruments that it should be of great interest to all who work with both electronics and music.

The Stroboconn is in essence an extremely accurate frequency meter. It was specifically designed by the manufacturer of the Connsonata (see Chapter 8) as an aid in tuning musical instruments. It is also highly effective in the laboratory and in industry for measuring vibrations, oscillations, and other periodic phenomena within its frequency range.

The instrument is in effect a superstroboscope. Twelve discs, each a replica of the one shown in Fig. 14-1, revolve at different

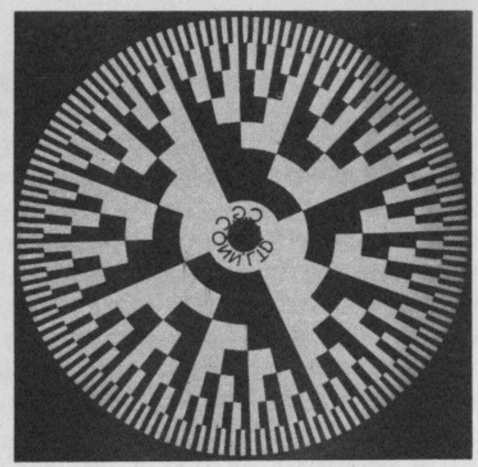


Fig. 14-1. One of the 12 discs used in the Stroboconn.

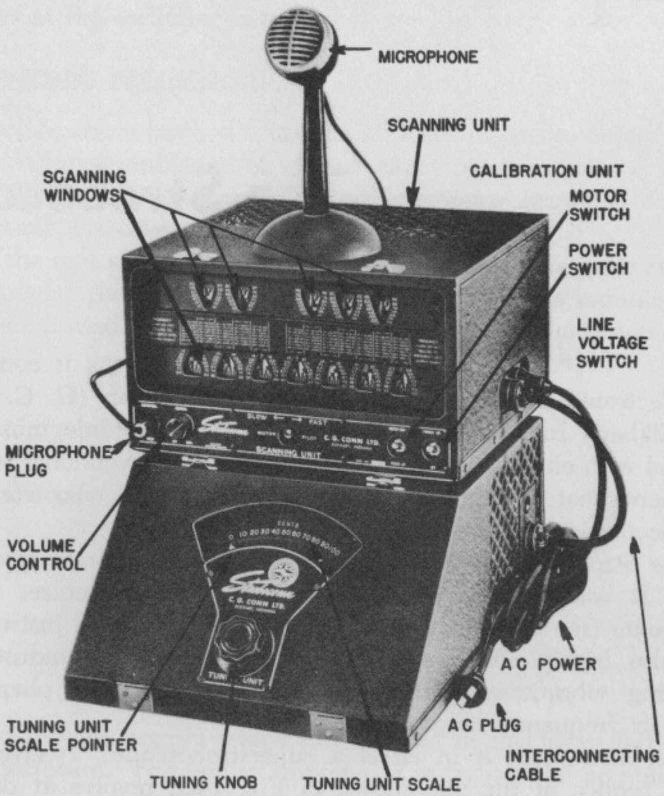


Fig. 14-2. The complete Stroboconn.

speeds. Each disc corresponds to a note of the tempered musical scale and revolves faster than the preceding lower one by a ratio of almost exactly the 12th root of 2. A special large neon lamp is excited by a microphone-fed amplifier. When the microphone is placed near a source of sound – a piano tone, for example, or an organ note – the neon lamp flashes at the frequency of the sound, illuminating the discs stroboscopically. If the speed of one of the discs is such that the number of black segments passing a given point per second is the same as the audio frequency being measured, the pattern appears to stand still. Calibrations on the Stroboconn then indicate the frequency. All twelve discs can be seen through small windows, as Fig. 14-2 shows.

Figure 14-3 is a photograph of one of the windows, exposing a segment of the disc behind it in motion. As Fig. 14-1 indicates, each disc has seven concentric rings of patterns, each ring having

twice the number of black portions as the adjacent inner one. Thus each disc takes care of the same note in seven succeeding octaves. In Fig. 14-3 the narrow part of the window exposes the innermost ring on the disc. The first three rings are invisible because the frequency being measured corresponds to that of the fourth ring, in which the stopped pattern can be seen. This tone apparently has some harmonic structure, for the fifth and sixth rings are also stopped, though they are illuminated less well.

ELECTRONIC CIRCUITRY

The most essential requirement for accuracy in an instrument of this kind is that the motor driving the discs run at constant

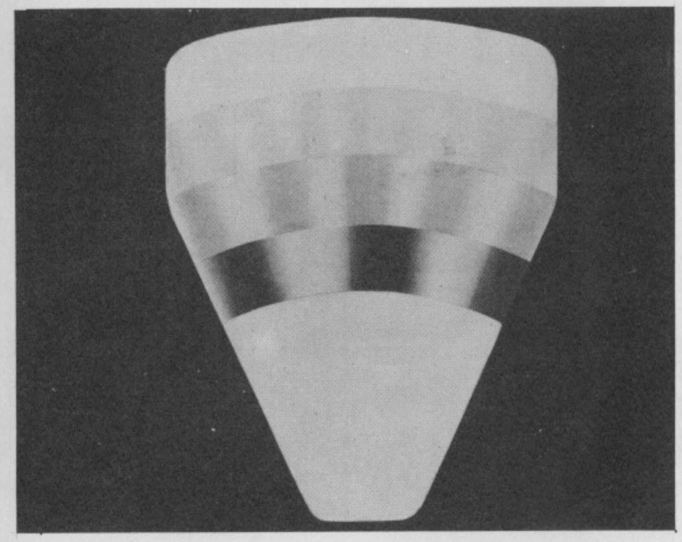


Fig. 14-3. Closeup of one of the windows.

speed. In the Stroboconn this regulation is provided by driving the motor with a tuning-fork-excited amplifier, diagrammed in Fig. 14-4. The tuning fork is made of a special alloy, called Connivar, which has an extremely low temperature coefficient. The metal is so stable that the accuracy of the fork varies only a maximum of .002 percent per degree Centigrade. For musical purposes a pitch accuracy of .05 percent is considered very high and few people can perceive pitch variations so small. In continuous operation the entire instrument, including gears, fork, and all other variants, is accurate to within .05 percent.

The fork is driven by a regenerative circuit, in which respect it resembles electronic oscillators. The pickup coil (Fig. 14-4) drives the first grid of a 6SC7. The plate output is fed to the grid of the second section, part of whose plate load is the fork drive coils. Thus the output of the fork is fed back in the correct phase to furnish driving signal, the driving power being supplied by the B-voltage source, a standard rectifier-filter combination.

The fork-controlled signal is then fed through a 6SN7-GT phase inverter to a push-pull-parallel 6V6-GT output stage. The secondary of the output transformer drives the synchronous disc motor.

The normal frequency of the tuning-fork oscillator is 55 cycles, but the frequency can be varied at will. Figure 14-5 shows the fork and the sliding weights whose position can be changed to

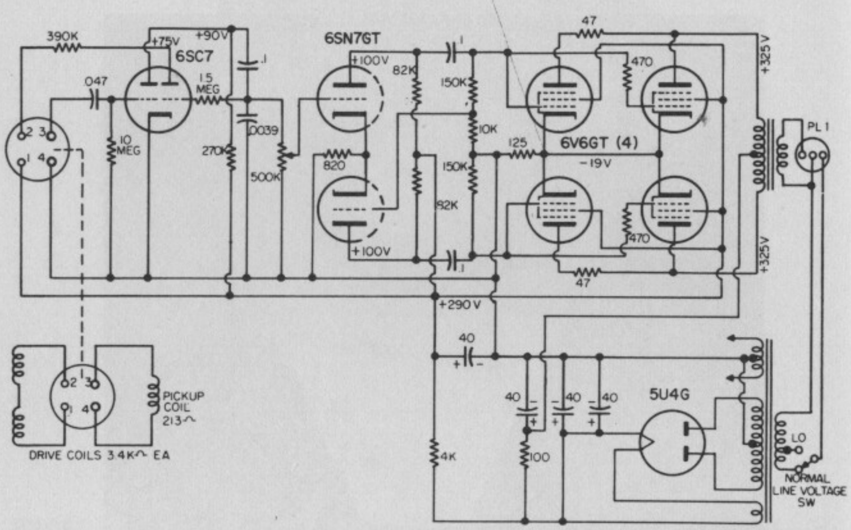


Fig. 14-4. Tuning-fork amplifier drives the motor.

vary resonant frequency. The varying mechanism is operated by a knob which appears at the bottom of the lower unit in Fig. 14-2.

The upper of the two units in which the Stroboconn is mounted contains the motor, discs, and flashing amplifier, diagrammed in Fig. 14-6. The amplifier is entirely conventional: a voltage amplifier, phase inverter, and push-pull output stage. A special, large, U-shaped neon lamp is fed by the secondary of the output transformer. The lamp is large enough to provide light behind all the discs, which are translucent.

The power connections in Figs. 14-4 and 14-6 are rather confusing. The scheme is shown better in Fig. 14-7, an extract from the other two diagrams. Two power switches are supplied. One, called the

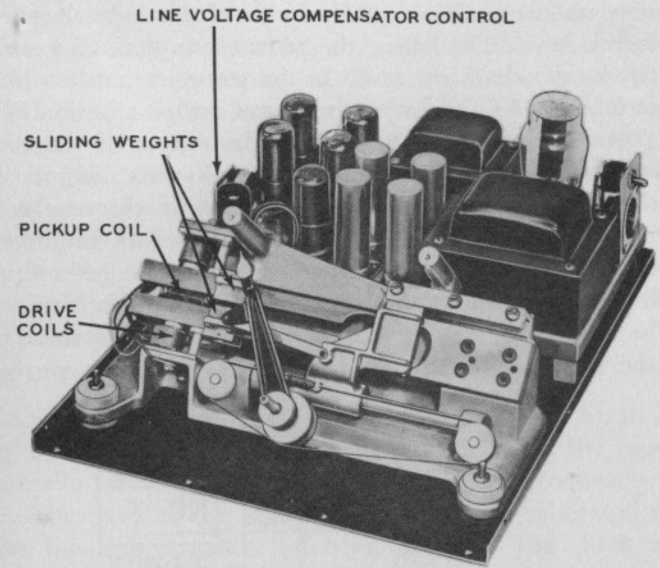


Fig. 14-5. The tuning fork.

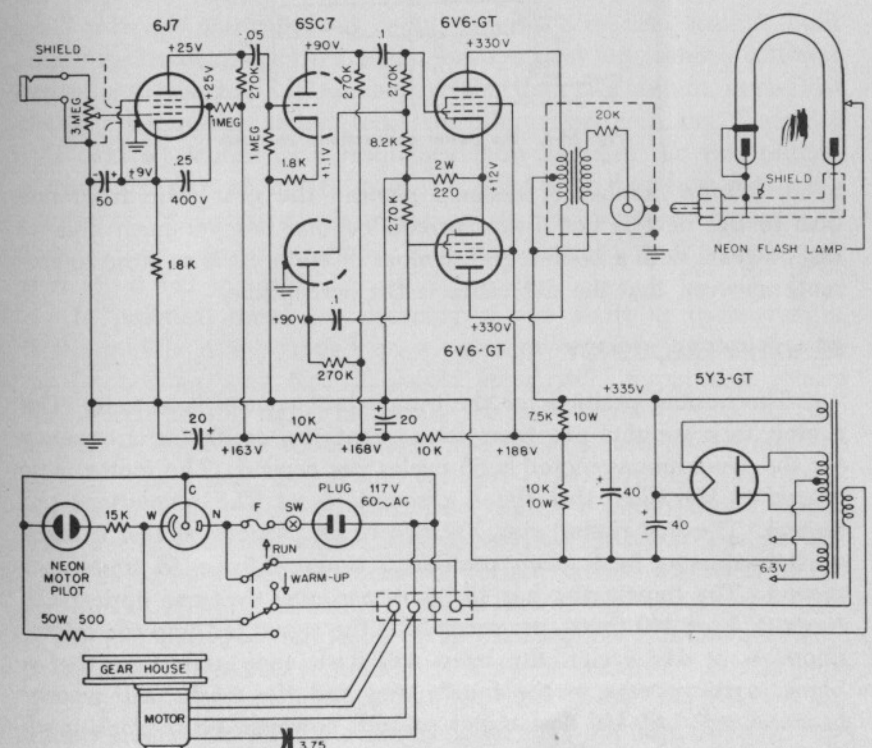


Fig. 14-6. Motor and flashing amplifier.

POWER SWITCH, is in series with one side of the 117-volt a.c. line and a protective fuse. The other, the MOTOR RUN-WARM UP switch, is a double-pole, double-throw unit. In the warm-up position in which it appears in Fig. 14-7 (and with the power switch closed) 117-volt, 60-cycle power is fed to the motor and the output of the tuning-fork amplifier is dummy-loaded by a 500-ohm, 50-watt resistor. After the motor and fork have warmed, the switch is thrown to the RUN position in which the motor is connected to the fork amplifier and the flashing amplifier power transformer is energized. After the flashing amplifier has warmed up, the instrument is ready for use.

The discs are run by the motor through a system of gears. Since the 12th root of 2 is not a rational number – it can be carried

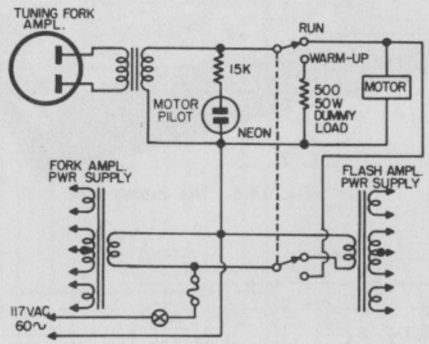


Fig. 14-7. The power connections, redrawn.

to an infinite number of decimal places – the gear ratio from one disc to the next is not quite correct; no one has yet been able to design gears with a nonintegral number of teeth. It is so close to correct, however, that the difference is not perceptible.

READING FREQUENCY

The normal position for the tuning unit control is at zero. The tuning fork weights are then set so that the oscillation frequency for the synchronous motor is 55 cycles per second. The motor is so geared to the discs that the A disc rotates at 27.5 revolutions per second. The first (inner) ring has two black portions so that a black portion appears in a given position 2 times 27.5, or 55 times, per second. The fourth ring has 16 black portions; thus one appears 16 times 27.5, or 440 times, per second. If the signal fed into the microphone is at 440 cycles, the lamp will flash once each time that a black mark appears in the fourth ring and the marks will appear to stand still. At 440 flashes per second, however, each black mark of the first ring is illuminated eight times each time it appears,

so it does not stand still. As a result the fundamental frequency of whatever tone is being measured is indicated by the innermost stopped pattern. Octave harmonics of the tone are indicated by the stopping of additional rings toward the outer edge of the disc (or the wide portion of the window).

Each ring of each disc is numbered around the windows in accordance with the ordinal number of the note. The calibration strip between the two sets of windows indicates the exact frequency of each note, based on an A of 440 cycles. That calibration is not necessary, of course, for routine instrument tuning, but is helpful for other purposes.

INTERPOLATION

Probably the feature most responsible for the Stroboconn's versatility in both music and nonmusic applications is the ingenious provision made for reading with great accuracy any frequency within its range. The basis of this is the knob-controlled, calibrated sliding weight on the tuning fork. Referring again to Fig. 14-2, as the tuning unit knob is rotated the pointer on the scale above it reads the frequency change in terms of hundredths of a semi-tone. These small intervals are known as *cents*. Since the musical scale is built on a logarithmic, not an arithmetic, basis, a cent represents a constant *percentage* change in frequency rather than a constant numerical change. A semi-tone or half-tone (the interval between, say, C and C# or E and F) represents a frequency ratio between the two pitches of the 12th root of 2 (see page 8). One cent therefore represents a ratio of $\sqrt[12]{2}/100$ which works out to an increment of approximately .059%. As an example, a 1-cent increase in the frequency of A-440 is 440 $\times$ 1.00059, or 440.25 cycles.

In practical terms the meaning of the above is quite simple. If the middle A is sounded on a piano or organ being tuned with the Stroboconn, ring No. 57 should stand still. Suppose it rotates very slowly to the right, indicating the tone is sharp; then to tune it to exact pitch the tuner adjusts the musical instrument until the pattern stands still. Or, to find out just how sharp it is, he rotates the tuning unit control upward from zero until the pattern stands still. If the pointer is then at 1 cent, he knows it is 0.25 cycle sharp.

Tuners rarely need to know how sharp or flat a tone is except for the "stretching" techniques to be described. Of greater value is the fact that any frequency of an oscillator, or vibration, or anything else in laboratory or industry may be measured in this way. A book of tables is furnished with the Stroboconn. With its aid the operator can discover the exact frequency (to five significant figures) of anything that can be either fed to the flashing amplifier or picked up by the microphone. He merely manipulates the tuning knob until one of the patterns is stationary, then refers to the tables for the frequency.

PIANO TUNING

The old-time piano tuner was usually a craftsman of great skill and years of experience. Many of the best of them started their careers in piano factories, working up by stages to the exacting job of adjusting new pianos before they were shipped. Expert tuning has always required an intimate knowledge of piano construction, a first-class ear for pitch, and much practical experience.

One of the reasons that tuning has never been a purely mechanical job is that an expert tuner does not actually tune the piano notes to theoretically correct pitch, except in the middle octaves. The overtones of a struck string are not true harmonics of the fundamental but are instead slightly sharper – higher in pitch – than the true harmonics would be. If the entire piano is tuned to exact pitch, the upper overtones of the middle octaves are sharper than the corresponding fundamentals of the upper octaves. Too, the upper harmonics of the lower octaves are sharper than the corresponding fundamentals of the middle octaves. The result is, as any musician who has ever heard a piano so tuned will instantly say, that it does not "sound right."

To make it sound more natural, the tuner "stretches." He tunes the upper octaves very slightly sharp and the bass ones flat. Most tuners, aware of the fallacy of tuning an entire instrument to true pitch, balk at the idea of using an electromechanical device.

However, the variable, calibrated pitch adjustments of the Stroboconn overcome the difficulty. After a little experiment, a tuner determines just how sharp or flat each note of these "stretched" octaves should be tuned to suit his and his clients' tastes and he notes the pitch change in cents. In future tunings, he needs merely to shift the tuning knob according to his notes and tune for a stopped pattern. The time saved and the accuracy gained are well worth while. The accuracy is especially valuable when two pianos, to be used in duo-piano playing, are tuned, for then it is essential that they correspond. For tuning the lower octaves it is not even necessary to experiment and make notes. The tuner simply adjusts the piano so that with the pointer on zero cents, the higher overtones of the bass notes, which appear in rings outside that of the fundamental, give stopped patterns. In this way, he is actually tuning the overtones of the notes to true pitch, not the fundamentals, which will automatically be slightly (and desirably) flat.

THE STROBOTUNER

Conn has recently placed on the market a new and less elaborate version of the Stroboconn, known as the Strobotuner. This unit, pictured in Fig. 14-8, has a single rotating disc run by a synchronous motor which is driven by an oscillator. The oscillator, which is calibrated before each use by comparison with the a.c. power line, may be switched to 12 frequencies corresponding to the 12 notes of the chromatic scale. In each position the disc is made to run at a speed

Fig. 14-8 The Conn Strobotuner

which is an integral submultiple of the note frequencies. Seven bands of patterns on the disc are illuminated by a neon lamp excited by the sound which is to be tuned. The instrument thus covers seven octaves, with a claimed accuracy of plus or minus 1/100 of a semi-tone. A calibrated control allows fine variation of the oscillator frequency and thus the motor speed to achieve variations of plus or minus 20 cents from the note frequencies based on A-440.

The Strobotuner weighs only 14 pounds and costs only about a third the price of the Stroboconn. It is thus ideal for general and especially portable use, where the more elaborate features of the older instrument are not necessary.

Chapter 15

Other Electronic Instruments

ALMOST all of the American-made electronic musical instruments on the commercial market have been described in the preceding chapters. There are a few other instruments, mostly in Europe, which have little or no sale in the United States. These additional instruments will be described briefly in this chapter.

THE THEREMIN

Perhaps the earliest electronic musical instrument to gain fame, if not general acceptance, was the invention of Dr. Leon Thérémin and was called by his name. It was manufactured commercially for a time by RCA Victor but is no longer anything but a curiosity.

The Thérémin is basically a simple beat-frequency audio oscillator containing two superaudio-frequency *L-C* oscillators. One is fixed in frequency and the other is varied. The outputs of the two oscillators are mixed in a detector and filtered. The resultant detector output is the difference frequency and is in the audio range.

Any variable audio oscillator can be used to produce music by simply manipulating the tuning knob with enough skill. It is an awkward trick, however, and the singular distinction of Dr. Thérémin's method was his method of controlling the variable-frequency oscillator. It is indicated in the circuit of Fig. 15-1. This diagram is that of a simple home-built version of the Thérémin, taken from a 1935 issue of *Radio-Craft* magazine.

Fixed and variable oscillators are shown. The variable oscillator has a metal rod connected to its plate circuit and this rod projects upward out of the chassis or case of the instrument. When the player's hand is brought near it, hand capacitance changes the tuning of the oscillator, the amount of change depending on the proximity of the

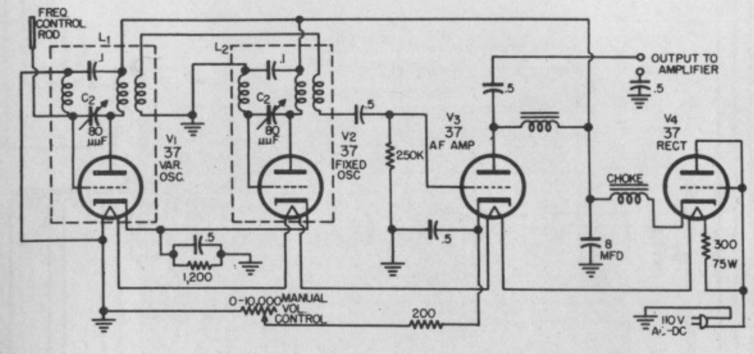


Fig. 15-1. One version of the Theremin.

hand. The oscillator trimmers are initially adjusted so that the two oscillators are on the same frequency – at zero beat – and there is no audio output. As the hand is brought near the rod the variable oscillator frequency decreases, so that an audible beat is heard. The highest audio tone is produced when the hand is closest to the rod.

It requires a good deal of practice to achieve any kind of musical results. There is no form of "keying," so the jumps between pitches are always glides unless the volume control is operated to shut off the sound between notes. The vibrato is produced by a rhythmic fluttering of the hand. The tone of the instrument is an eerie sort of wail, both because of the gliding tone and the purity of the waveform. No form of the instrument of which the author knows had any method of varying the approximately sine shape of the detected beat note, though it is not hard to see how it might have been passed through distorting and filtering circuits without much trouble.

The circuit of Fig. 15-1 shows an ordinary manual volume control, which some of the inventor's earlier instruments had. He found this awkward, however, for doing any sort of keying and later developed a volume control which, like the pitch-determining system, depended on the proximity of the hand to a rod. The RCA version of the Thérémin, diagrammed in Fig. 15-2, shows this modification.

V_1 is the variable oscillator, controlled by the pitch-control rod coupled to the tuned circuit, and V_3 is the fixed-frequency oscillator. V_2 is a detector, the plate of which passes audio through a transformer to V_4, the first audio amplifier, thence to V_6, the output tube. A PLAY-OFF switch shorts the grid of V_6 to ground to silence it.

Volume control is made possible by V_5 and V_7. Plate current for V_4 passes through V_5, which is connected as a rectifier. The fila-

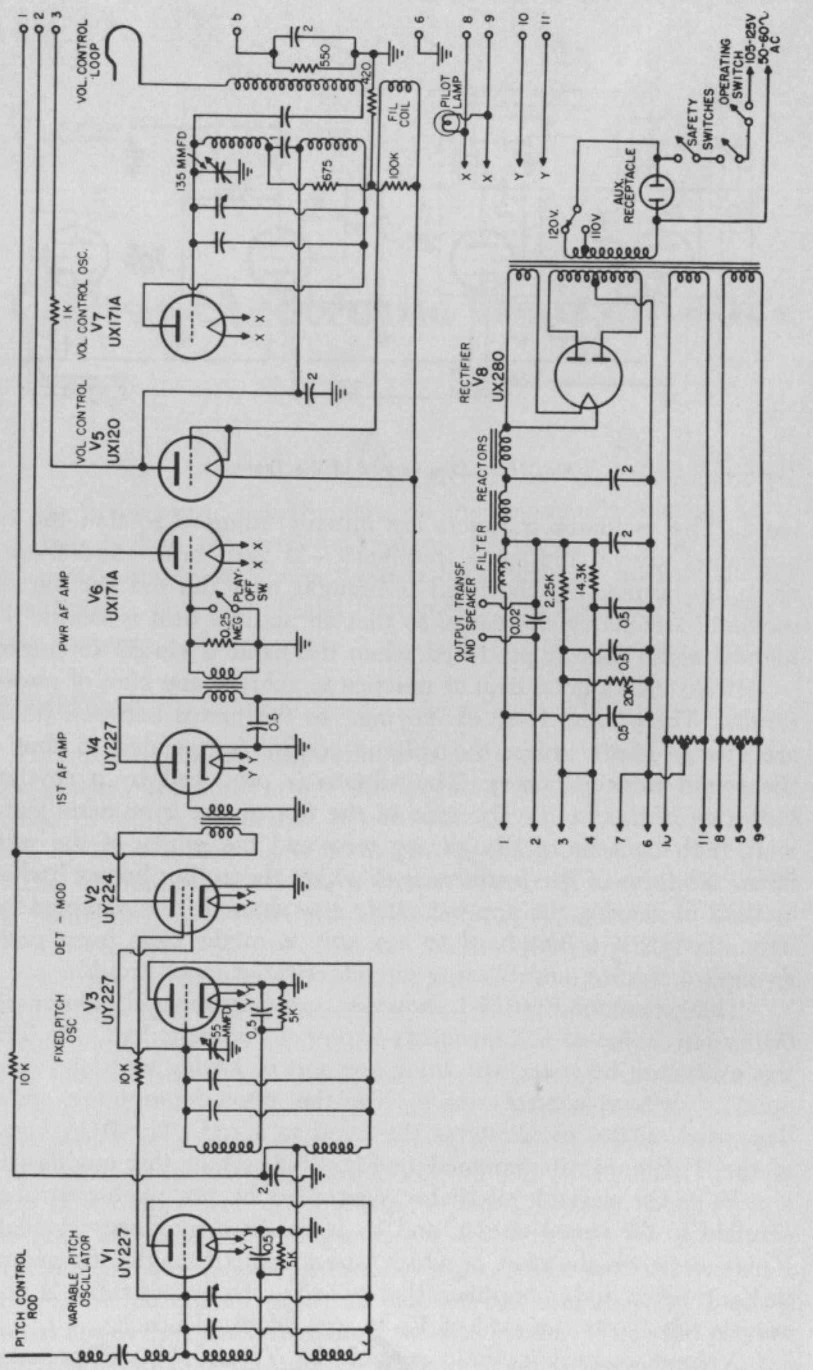


Fig. 15-2. A Theremin made in former years by RCA.

Fig. 15-3. Console of the Compton Type 347.

ment of V_5 is not heated by the power supply of the instrument, but by power taken from oscillator V_7. Under normal conditions the oscillator resonant coil (so marked on the diagram) is not resonant at the oscillator frequency. It does not, therefore, transfer any appreciable power to the filament coil coupled magnetically to it. To raise the volume the operator brings his hand near the volume control loop. The hand capacitance brings the oscillator resonant coil nearer resonance at the oscillator frequency and some energy is coupled to the filament coil. This heats the filament of V_5, allowing some plate current to pass through V_4 so that V_4 can amplify. When the hand is nearest the loop, the coil is nearest to resonance, maximum filament heat is obtained, and maximum V_4 plate current allows maximum volume.

With this arrangement, a practiced player can produce some interesting effects not only by varying volume in the usual way, but by giving varying attack and decay characteristics to the keying of tones. It takes a particular adroitness, however, and this, coupled with the rather limited musical interest of the tone, has prevented any really widespread use of the Thérémin.

THE COMPTON ELECTRONE

The most important European instrument today is the English Compton Electrone, made by the John Compton Organ Co., Ltd., London, and developed almost entirely by L. E. A. Bourn, the company's Chief of Research. The several models of the Compton include some of the most elaborate specifications of any electronic instrument

Fig. 15-4. A special Electrone made for the BBC.

ever made. The three-manual model made for the Festival of Britain, for example, had 93 stops, eight couplers, and three separate tremulants. The more standard models are like the type 347, the console of which appears in Fig. 15-3. Figure 15-4 shows a model used by the BBC; the Festival organ is similar to this, but with three more rows of stops.

The Compton is an electrostatic organ, using twelve rotating wheels to generate the tones. Each of twelve 5-inch-diameter stator discs is made of Bakelite, with one surface metallized. In the metallized surface, grooves are engraved with the desired waveforms and frequencies. Figure 15-5 shows a sector of one simplified disc which generates four tones. Area W_1 generates the first tone. The lines show the engraving, which cuts through the metal surface down to the bakelite, thus insulating the area from adjoining ones. Area W_1 is bounded by an outer insulating circle and an inner insulating pattern representing tone waveform. It is thus very similar in form to a variable-area film soundtrack.

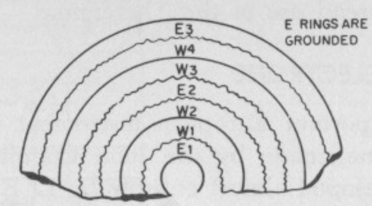


Fig. 15-5. A sector of one generator disc.

Area E_1 simply results from the engraving and is grounded. Area W_2 is a second tone-generating area, with, theoretically, a different waveform but the same frequency. Area E_2 is a "waste" resultant and is also grounded. Areas W_3 and W_4 are additional waveform areas (supposedly, though not accurately shown in the drawing, of the same frequency), while area E_3 is another grounded "waste" area. On an actual disc there may be as many as thirty different tone areas. Since there is only one set of generators in the usual instrument, each disc must contain as many frequencies as there are octaves in the organ, and they contain repetitions of these frequencies in different waveforms.

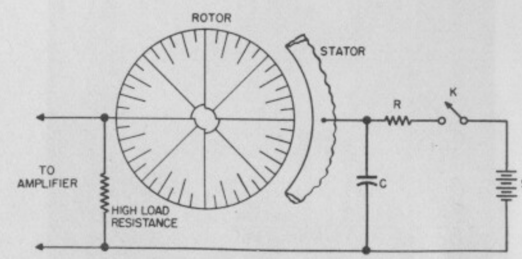


Fig. 15-6. Pattern of a rotor.

The rotors consist of metal or metallized Bakelite discs on the surface of which are radial ribs in a pattern like that shown on the schematicized rotor of Fig. 15-6. The ribs are so arranged, along with the patterns on the stators, that when the rotor rotates very close to the face of the stator the radial ribs opposite any pattern are just one wavelength apart. As is apparent from the drawing, the lowest frequencies appear toward the centers of the discs. The drawing of the rotor in Fig. 15-6 allows for only three octaves (there are three different lengths of radial ribs) but in the actual rotors there are as many different rib lengths as octaves.

The ribs and the tone patterns constitute the two plates of capacitors. A source of voltage and a load resistance connected as shown in Fig. 15-6 produces a tone when the rotor is rotating because the variable generator area is a variable capacitor plate. The principle is exactly the same as the pickups for the Wurlitzer organ described in Chapter 11.

All the ribs of the rotor are connected in common to the top of the load resistor and the amplifier input. A pin through each tone area on the stator connects the area to a key through an R-C time-delay network. Thus no tone will sound until its key is pressed, con-

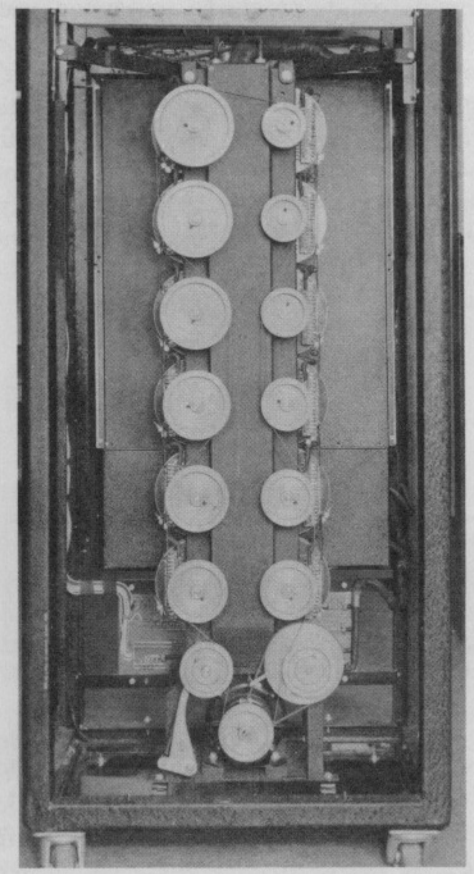


Fig. 15-7. One side of a Compton generator rack.

necting the high voltage to the area. The *R-C* network causes a gradual application of voltage to avoid clicks and to give a good musical attack. On each stator disc the grounded *E* areas isolate patterns of different frequency to prevent interference and act as differential capacitances to keep the capacitive loading of the rotor constant.

The generator and parts of the electronism are contained in a metal rack. One side of the rack is shown in Fig. 15-7 with the cover off to reveal the generator drive assembly. Figure 15-8 shows one of the rotors and its pulley. The twelve rotors mounted in the rack are belt-driven, with speeds fixed by the pulley diameters. The rack with covers on looks like Fig. 15-9.

The Compton is a synthesis instrument but does not operate by combining pure sine waves as does the Hammond. The five basic

tone waveforms (obtained by engraving the desired waveforms in the stator discs) are complex. One is almost pure sine with some odd harmonics. The second is similar but an octave higher. The third is a relatively short progression of odd harmonics beginning at the third and tapering out at about the 13th. The fourth is a waveform composed of even harmonics from the second to the 32nd. And the fifth contains odd harmonics up to the 31st. These five basic tone qualities for each note are mixed in varying proportions to give a large number of different stop qualities.

The mixing circuits include relays, resistors, and regulating strips, which are shown in Fig. 15-10 on a swing-out subpanel in the main

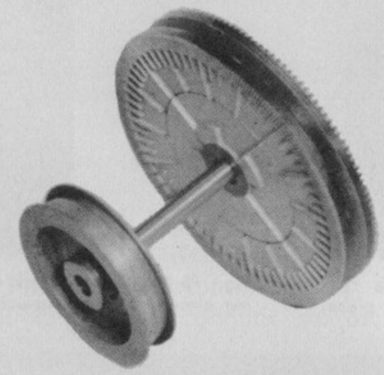


Fig. 15-8. A rotor and pulley.

rack. Figure 15-11 shows the rear of the console of Fig. 15-3, with the terminal board for all the stop and key controls at right and the combination action (piston) setter at left.

THE CONSTANT MARTIN ORGAN

Figure 15-12 shows the console and tone cabinet of the Martinette, a one-manual, pedalless model of an electronic organ type developed by a Frenchman, Constant Martin, and manufactured in England. More elaborate models have also been developed on the same principle.

The organ contains a separate oscillator for each note rather than locked divider chains, the inventor having considered the resultant ensemble effect of great importance. The tone coloring method is of the formant type. Thus the scheme bears traces of both the Baldwin and Connsonata described in earlier chapters.

Each of the oscillators is similar to that diagrammed in Fig. 15-13. It is roughly tuned by capacitor *C* and fine-tuned by moving laminations in inductor *L*. It is keyed by breaking and making the B-plus

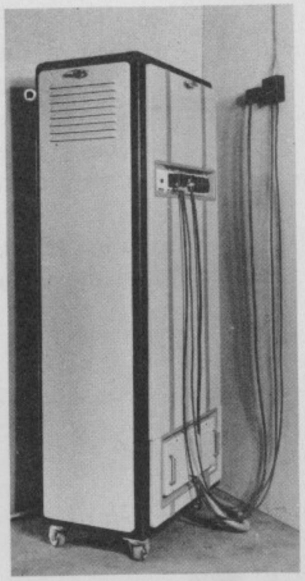


Fig. 15-9. The rack with covers in place.

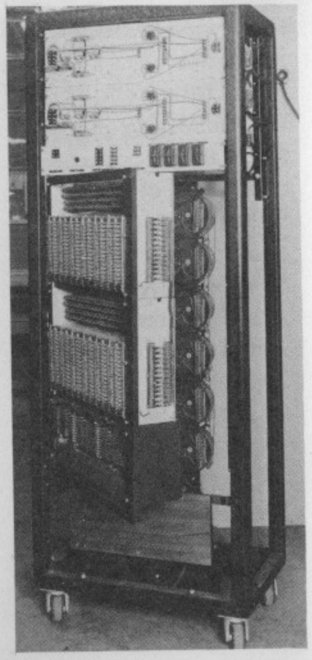


Fig. 15-10. Components of the mixing circuits on a swing-out rack panel.

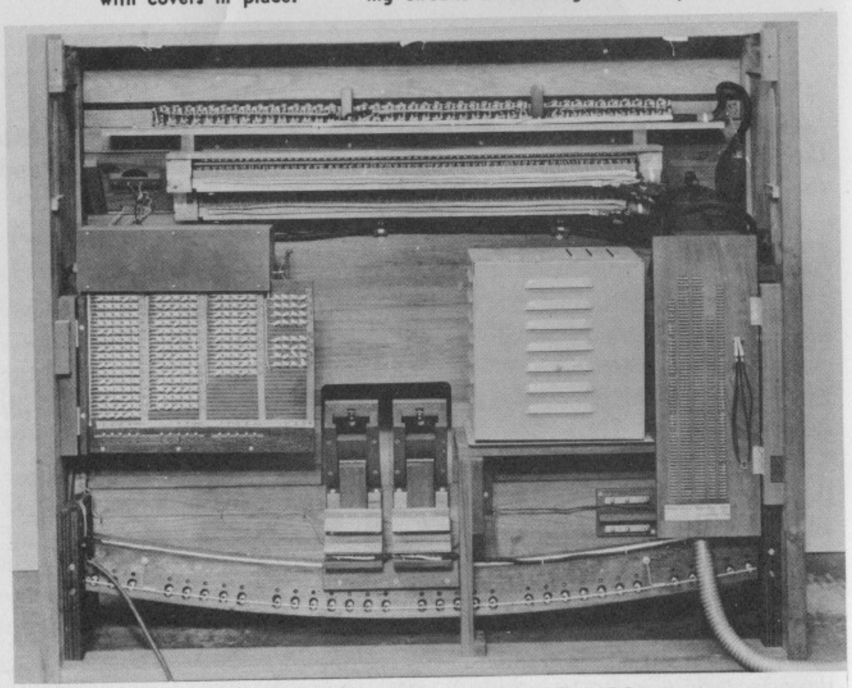


Fig. 15-11. Rear of the console of Fig. 15-3.

Fig. 15-12. The Martinette.

line to the plate circuit and output is taken from the secondary of a plate transformer. The waveform is very complex, the feedback being much more than sufficient for oscillation.

A single oscillator chassis contains six oscillators, two being required for an octave of notes. All the transformer secondaries on one chassis are connected in series. There is a "selector" circuit for each two generator chassis – each octave, that is – and it is diagrammed in Fig. 15-14. The twelve notes for the octave are mixed by interconnections of the transformer secondaries, with each chassis connected to one pair of primary windings. One set of interconnected secondaries has low-inductance windings; the tones going through it are lacking in low frequencies and go to the grid of V_1. The output of V_1 is called the reed output. The other windings have high inductance and take tone to V_2, which gives a flue or flute output, so-called, though it is, of course, replete with harmonics.

There is one of these duo-triode selector circuits for each octave

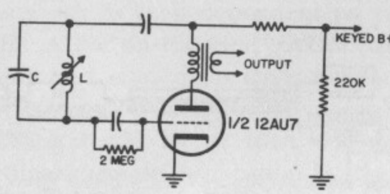


Fig. 15-13. Oscillator used by Constant Martin.

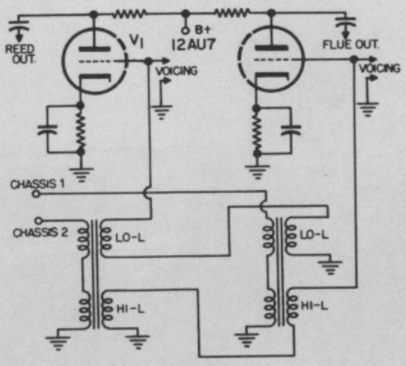


Fig. 15-14. Diagram of the Martin "selector" circuit.

of tones. All the outputs are commoned to an amplifier, expression pedal, and dual speaker system.

The various stop qualities are achieved by connecting capacitors and resistors between the selector circuit grids and ground. The stop tabs connect these so-called voicing components in every selector, so that each tab must operate as many contacts as there are octaves in the organ.

LES ONDES MARTENOT

Les Ondes Martenot (literally The Martenot Waves) is the invention of Maurice Martenot, a Frenchman. It is manufactured under M. Martenot's aegis in Neuilly-sur-Seine, probably in extremely limited quantities. The writer's examination of the only Martenot to be found in New York indicated that construction was done by hand, apparently without benefit of any machine-made parts. The owner of the instrument was a pupil of M. Martenot, and this furthers the impression that very few instruments exist.

The block diagram of Fig. 15-15 indicates the general functioning of the Martenot. It is a beat-frequency instrument with the usual fixed- and variable-frequency oscillators. The variable oscillator is

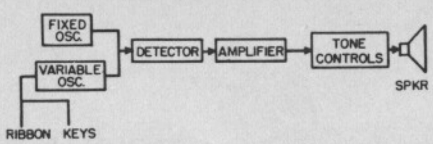


Fig. 15-15. Block diagram of the Ondes Martenot.

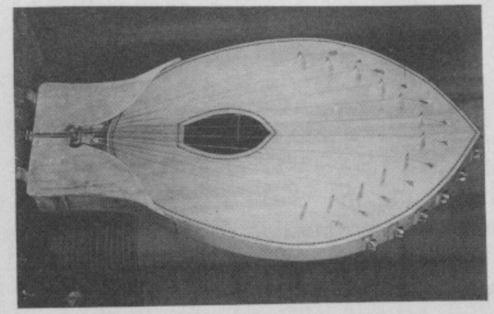


Fig. 15-16. The special Martenot sound transducer which diffuses sound and also has sympathetic resonance effects.

controlled in two ways. The first is by a seven-octave keyboard which switches in various capacitances. The second is by a long ribbon which runs the length of the keyboard and over pulleys at the ends to operate some sort of variable capacitance. To use the ribbon the player places his index finger in a sort of thimble and moves it left and right to vary pitch and introduce vibrato.

The two oscillators are mixed in a detector which yields the audio beat note, probably deliberately distorted to give some sort of complex waveform. It is amplified, then fed through several simple selectable filter circuits, thence to a loudspeaker. Some of the instruments use ordinary loudspeakers. Others have the *diffuseur-resonateur* shown in Fig. 15-16, which consists of a transducer of some kind which imparts vibrations to a sounding board with stretched strings. The strings are tuned to various pitches and when one or more is excited by a particular output frequency of the instrument it vibrates and adds a weird sort of lingering emphasis.

The instrument does not stay in tune for any length of time and must constantly be retuned by adjusting a trimmer until octavely related keys sound correctly. This is less of a handicap when the ribbon is used since this in effect gives stepless control of frequency and the music is then dependent on the player's ear, just as in the Theremin. A bar on the tone control panel must be pressed to obtain any output and is used for keying.

The Martenot is an interesting novelty and has been characterized in more glowing terms by at least one well known modern composer. For ordinary use or duplication by the average technician or musician it is felt by the writer to be impractical, both technically and musically.

THE KINSMAN ORGAN

In 1957, Kinsman Manufacturing Co., Inc., of Laconia, N. H. put into production a new electronic organ which is a development product of a group including former members of the Minshall organization and the writer. Designed to give results fully comparable with those of the most expensive spinet organs, the Kinsman spinet is sold at a much lower price because of economies achieved in the tone-generator circuits. While the organ has many outstanding features, the tone generators are the most interesting of the organ's technical points.

For many years organ designers have been attracted by the possibilities of using neon lamps as tone generators. They are much cheaper than tubes and they yield the sawtooth waveform which Baldwin and

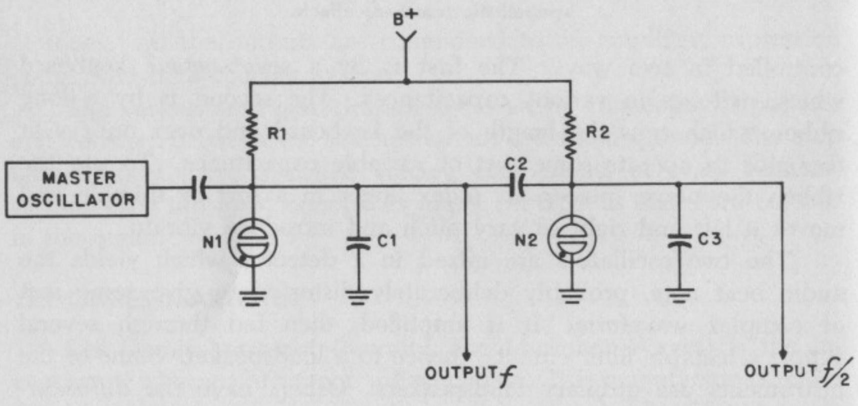


Fig. 15-17. Simplified schematic of typical neon oscillator circuit synchronized by a master oscillator.

Schober have proved to be ideal for formant tone coloring. While several neon circuits have been used by the writer and others in experimental organs, all have suffered from the important defect that the standard neon relaxation oscillator cannot be synchronized reliably with another tone source so as to be usable as part of a frequency dividing system.

Figure 15-17 shows a typical pair of synchronized neon oscillators, with one of several available synchronizing methods, all of which suffer from the same faults. N_1 is the higher-frequency oscillator operating at frequency f, with the standard resistor and neon lamp and the timing capacitor across the lamp. If not coupled to any other oscillators or a load, it will operate at a free-running frequency determined by values of R_1 and C_1, the characteristic striking and extinction voltage of the lamp, and the voltage of the B+ supply. We assume at the moment that it is synchronized to the master oscillator and we are not concerned with the method of sync. The

second neon oscillator is similar to the first, except that the timing components are chosen so that its free running frequency will be about half that of the first oscillator. Its desired final frequency is $f/2$ and to obtain this, synchronizing voltage is injected from the first oscillator into the second through C_2. If C_2 is large enough by comparison with C_3 so that enough sync voltage is fed to the top of N_2, the second oscillator will synchronize and produce frequency $f/2$. However, if C_2 is large enough to accomplish this result, then a large component of f will be heard in the output of $f/2$ and a large component of $f/2$ will be heard in the output of f. Musically this is very unsatisfactory, especially the latter. In addition, because the two oscillators are so closely coupled, each and its components will affect the "tuning" of the other and it will be difficult to arrive at a set of components which will maintain stable division. The divider diagrammed in Fig. 20-5 tends to avoid the interaction but insufficient sync is present for reliable commercial operation over a period of time. The same is true of the method used in the writer's Electron-organ, described in *Radio & TV News* and an early edition of this book, in which a turn of wire around the lamp was used for sync coupling.

What may be termed the first and – so far – only good, reliable, commercial method of synchronizing neon lamps is used in the Kinsman Organ. In combination with well researched methods of pre-aging neon lamps to stabilize their characteristics, this sync method makes possible for the first time tone generators which are really inexpensive yet wholly suitable for commercial distribution to consumers.

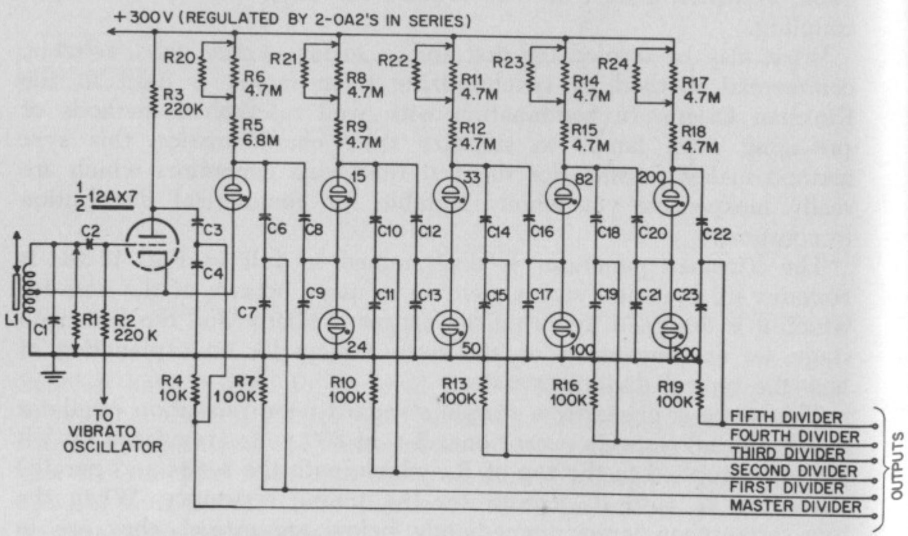
The Kinsman generator is diagrammed in full in Fig. 15-18. It contains six generator stages yielding as many octaves of the note for which it is designed. Ignoring the master oscillator and the first neon stage, let us concentrate on the second stage for an explanation of how the typical divider operates.

This stage is primarily a simple standard neon relaxation oscillator familiar in all respects except one. B+ at 300 volts (regulated by VR tubes) is applied to the top of R_8, which, with the series and parallel values of R_{21} and R_9, constitutes the timing resistance. When the two series neon lamps immediately below are extinct, they are in effect an open circuit. Current passes through the resistance and begins to charge the timing capacitance, which is the net series value of C_{10} and C_{11}. Since at the outset a good deal of current is passing through the resistance, the voltage between the top of the upper neon and ground is relatively low, not enough to cause the lamps to strike. As the capacitance charges, the current through the resistance decreases in an exponential manner, and the voltage across the neons rises in the same way. At a certain time, this voltage is sufficient to ignite the lamps, upon which the lamps become relatively low resistances and the capacitance is quickly discharged through them.

The lamps then go out and again become an open circuit after which the same action begins again. Since the charge across the capacitance builds exponentially and then dissipates very quickly once each cycle, the voltage across the capacitance is in the form of a sawtooth. C_{11} has ten times the capacitance of C_{10}, meaning that it has only 1/10 the reactance. Output for the organ is taken from across C_{11} so that what happens to the output circuit – loading and signal – will not affect the frequency-dividing stage. This is the same capacitive output system as is used by Baldwin and Schober.

As described to this point, the neon oscillator is entirely standard, except that two lamps are used in series rather than a single lamp. Notice now the next neon stage, R_{11} etc., which is identical to that just described, except that the C values are doubled to obtain a free-running frequency approximately an octave lower. Let us see how this next stage is synchronized by the earlier one.

In addition to the timing and output capacitors C_{10} and C_{11}, there are connected across the first oscillator C_{12} and C_{13} in series. The values of these two capacitors are very much smaller than C_{10} and C_{11},



	C1	C2	C3	C4	C6	C7	C10	C11	C14	C15	C18	C19	C22	C23
C	.02	.002	.025	.1	.0008	.008	.0016	.016	.0032	.032	.0064	.064	.0125	.125
C#	"	"	"	"	"	"	"	"	"	"	"	"	"	"
D	.016	"	"	"	"	"	"	"	"	"	"	"	"	"
D#	"	"	"	"	"	"	"	"	"	"	"	"	"	"
E	.0125	.0016	.02	.08	.00064	.0064	.00125	.0125	.0025	.025	.005	.05	.01	.1
F	"	"	"	"	"	"	"	"	"	"	"	"	"	"
F#	.01	"	"	"	"	"	"	"	"	"	"	"	"	"
G	"	"	"	"	"	"	"	"	"	"	"	"	"	"
G#	.008	.00125	.016	.064	.0005	.005	.001	.01	.002	.02	.004	.04	.008	.08
A	"	"	"	"	"	"	"	"	"	"	"	"	"	"
A#	.0064	"	"	"	"	"	"	"	"	"	"	"	"	"
B	"	"	"	"	"	"	"	"	"	"	"	"	"	"

Fig. 15-18. Complete schematic of Kinsman tone generator.

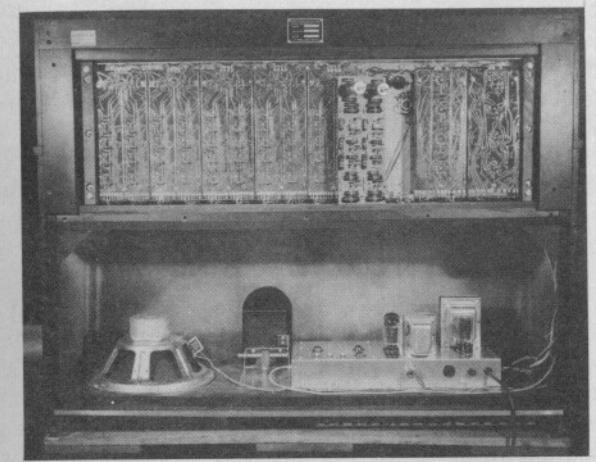


Fig. 15-19. Rear view of the Kinsman organ with the back removed.

so much so that C_{12} and C_{13} have no appreciable effect on the timing of the generator stage. They act as a capacitive voltage divider, and the proportion of the sawtooth voltage generated by the first stage described which appears at the junction of C_{12} and C_{13} is solely a function of the relative reactances of C_{12} and C_{13}. In this case, a little over one-third of the output of the stage appears at this junction and is applied to the junction of the two neons in the second stage. Since the junction of the neons is a point of very high impedance when the neons are not conducting (which is most of the time, since they conduct only a very brief discharge pulse once per cycle), there is a minimum of loading on the capacitive voltage divider. In this way, the full voltage determined by the capacitive divider reaches this point despite the small actual values of C_{12} and C_{13}.

The timing components of the second stage are deliberately chosen so that the voltage at the top of the neons will not quite reach the striking voltage of the lamps at the instant the first pair of neons fires. When the first pair of lamps does fire, a large negative pulse is created across them. A little over a third of this is transmitted through the C_{12}-C_{13} divider to the bottom of the upper lamp of the second stage, Since the voltage at the top of this lamp is nearly positive enough to cause striking, the fact that the bottom is pulled negative creates sufficient over-all voltage across the lamp to fire it. And when the upper lamp fires, it becomes a low resistance and applies voltage to the lower lamp, which then fires a brief instant later. In effect, therefore, the striking of the first set of lamps causes the second pair to strike and synchronism is achieved. Because of the timing components of the second stage, this happens only once for every two firings of the first lamp pair, since on every other firing of the first set, the voltage at top of the second pair is so low that even the negative pulse is not enough to cause firing.

All of the conditions for reliable and musically satisfactory operation are present here. The applied sync voltage can easily be made large enough to guarantee reliable synchronization. Yet signal from the first stage cannot appear in the output of the second because except for the brief instant of firing the first oscillator is isolated from the second by the open-circuited, extinct upper lamp of the second set. Signal from the second stage cannot appear audibly in the first; to be applied it would have to pass through C_{12} to the capacitance of the first stage. But C_{12} and the capacitances C_{10} and C_{11} are a voltage divider, and C_{12} as the series arm has such a large reactance compared with C_{10} and C_{11} that the output lead at the junction of C_{10} and C_{11} receives fed-back signal from the second stage reduced by a factor of at least 300 times or about 50 db.

Two more factors enter into the practical design of this generator. Since production neon lamps (type NE-2 is used) are not uniform, either the R or C of each stage must be aligned at the factory to set the free-running frequency of the oscillator within a range which will respond adequately to the sync voltage. The two resistors R_8-R_9, R_{11}-R_{12}, etc., of each stage are standard, but at final alignment a padder resistor R_{21}, R_{22}, etc., is selected for each oscillator. The second factor is change in lamp characteristics with age. The striking voltage of the typical neon lamp varies rather erratically for the first part of its life on its way to a final high value. It then decreases very gradually for the remainder of the life, which is well upward of 10,000 hours. To overcome the initial erraticism and reach slightly beyond the start of the predictable point, the lamps in the Kinsman Organ are aged artificially before use. After several thousand hours of operation, the characteristic is sufficiently altered so that a change in the padding resistor may be necessary. This is taken care of by a unique servicing system made possible by the organ's peculiar construction.

The entire organ, with the exception of the cabinet, is built as a series of "modules," individual sections, almost all of which are etched-circuit panels, each of which is separately removable. Most of these are shown in Fig. 15-19, which shows the rear of the organ with the back removed. The framework near the top holds all twelve generators, two to a panel, the bus amplifier panel, and the panel containing the pedal generator and the percussion circuit. Any of these may be removed by taking out screws and pulling off connectors. The voicing panel is shown in Fig. 15-20, at the left of the upper manual. This is a single etched circuit which holds all the formant filters for the stops and also mounts all the stop tablets and the vibrato control. It comes out as a unit when the metal cover is taken off. Figure 15-19 shows the combination amplifier-power supply

on the floor of the console. This, with the swell shoe, which is an integral part of the amplifier assembly, comes out after removal of a few nuts. The manuals and the key switches which are mounted to them are also easily removable.

Servicing of the Kinsman does not require a technically trained man. Each dealer is furnished several sets of spare sections. When an organ has a malfunction, the dealer's salesman or other nontechnical representative determines which section is at fault by following simple instructions in the service manual, and then simply replaces that section as a unit. The owner thus gets an immediate and inexpensive repair job. The faulty section is sent back to the factory with a small standard service charge and a good unit of the same type is returned to the dealer. The faulty section is repaired at the factory. No actual repairs are made in the field, which relieves the dealer of the necessity of maintaining a service technician and guarantees the owner the benefits of factory service.

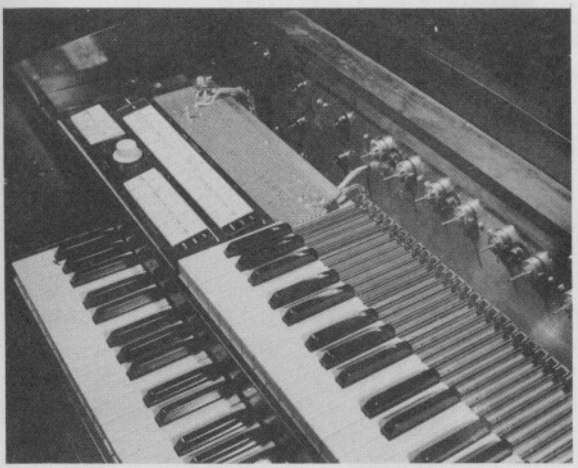


Fig. 15-20. View of the voicing panel which is located at the left of the upper manual.

Several other features of the Kinsman are of interest but cannot be described in detail. There are 19 separate stops on the organ, and duplexing is used, as in the theatre organ, so that many of the same stops may be used on either or both manuals without the use of couplers which would transfer all stops simultaneously between the manuals. There is a soft-normal tab for each manual so that the relative levels of the manuals may be adjusted during playing for the desired balance. The vibrato control, instead of being a tablet, is a knob which allows continuous control of vibrato depth from zero to maximum and allows selection of the precise vibrato desired for each selection. The pedal generator, a flip-flop system synchronized by the main generators, provides four pedal stops at both 8′ and 16′ pitches without the added cost of a seventh stage on each tone generator. Electronic percussion is a part of each standard organ. Extraordinarily good tonal production is obtained by a large and heavy bass speaker plus two treble speakers and deliberate design of the lower portion of the console as a speaker enclosure of the ported type. The manuals have 44 keys each and there are 13 pedals, bringing the specification of the Kinsman to equivalence with the most expensive spinets on the market.

THE THOMAS ORGAN

The Thomas Organ, made by Thomas Organ Co., Sepulveda, Calif., is remarkable chiefly for two factors. First, its sale price has been comparatively low, less than any electronic organ available before it. Second, it is the first single-manual electronic organ to have achieved really wide distribution. The two factors are probably related.

Technically, the point of principal interest is the tone-generation section, the trick in which is the use of a single oscillator to cover three adjacent tones. Pressing a key not only applies keying voltage

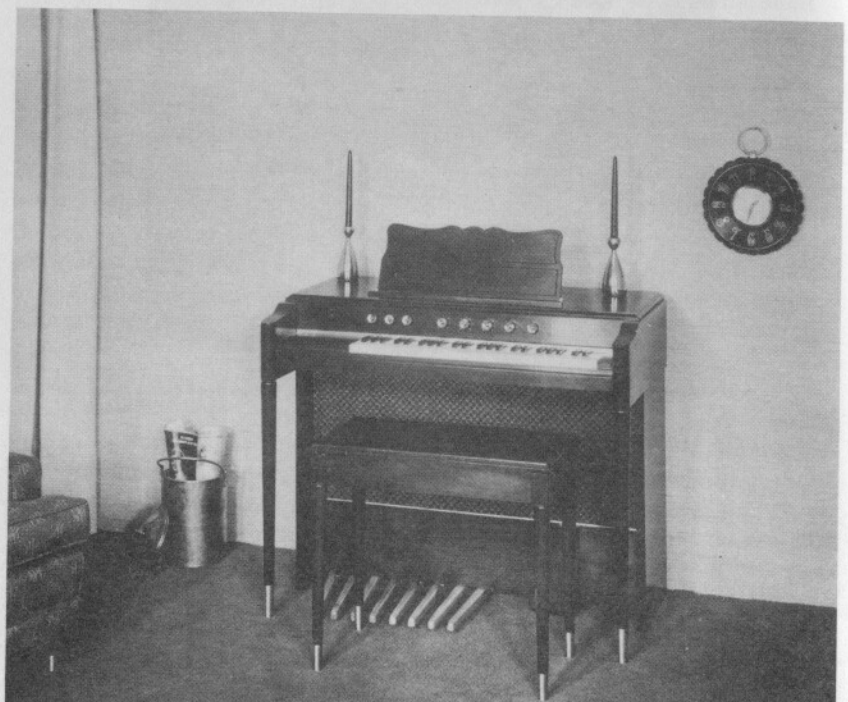


Fig. 15-21. The console of the Thomas Model G-1 organ.

to the oscillator but also tunes the oscillator. The console of the Model G-1 organ appears in the photo of Fig. 15-21.

Figure 15-22 is a schematic diagram of one typical oscillator. S_1, S_2, and S_3 are the switches operated by the keys of the three notes covered by the oscillator. The oscillator is a Hartley of special design which, like those used in the Conn, produces tones of two waveshapes; one is a sine wave and is called flute tone and the other has a complex shape and is called the complex tone.

Plate and cathode voltages are continuously applied to the oscillator at 150- and 23-volt levels. The latter is sufficient to bias the tube off with the switches in the positions shown (key up). When the highest of the three notes is played, the arm of S_3 contacts the upper stator

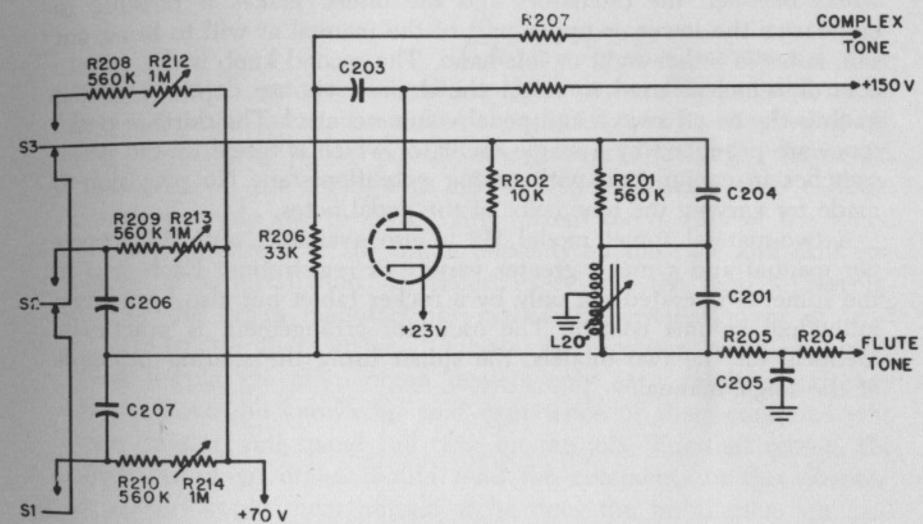


Fig. 15-22. Schematic of one typical oscillator section of the Thomas organ.

contact and the lower one is opened. This connects the 70-volt keying bus through R_{212} and R_{208} to the tube grid (through R_{202}) and the tube oscillates at a frequency determined primarily by the main tuned circuit L_{201}-C_{201}-C_{204}.

When the middle note is played keying voltage is applied to the grid through R_{213} and R_{209} and the normally closed contacts of S_3. In addition, C_{206} is shunted across the tuned circuit. When the lowest note is played, S_1 causes a similar action with R_{214}-R_{210} and C_{207}.

The oscillator frequency is apparently quite sensitive to the value of applied d.c. grid keying voltage, since variation of this voltage is used for fine tuning of the individual notes; this is the purpose of variable resistors R_{212}-R_{213}-R_{214}. Vibrato is produced by varying this same voltage with an oscillator on the amplifier-power-supply chassis.

The basic voltage is set by a gaseous regulator tube. For coarse tuning of the entire group the inductor L_{201} is variable, its core being movable and the unit being very similar to the coils used by Conn. Only eight duo-triodes are used in the entire tone-generator section of the instrument to cover the four-octave range.

Tone coloring is controlled by the five knobs to be seen nearest the right on the nameboard in Fig. 15-21, one knob each for horn, string, reed, flute, and diapason. The two types of oscillator output are fed to simple formant filters, each of which has a potentiometer to regulate its output; any total tone color is the result of the player's mixture of the five basic colors. The third knob from the left is the "solo" control, which has three positions and which, by resistive networks between the oscillators and the filters, makes it possible to emphasize the lower or upper part of the manual at will to bring out solo notes in either right or left hand. The second knob is the vibrato control, which is used to select the desired vibrato depth. The first knob is the on-off switch and pedal volume control. The thirteen pedal tones are produced by a single oscillator which is tuned by the pedal switches in conjunction with tuning potentiometers. No provision is made for varying the tone color of the pedal notes.

A two-manual spinet model H2 is also available, with 3½ octaves per manual and a much greater variety of registration. Each stop in the spinet is operated not only by a rocker tablet but also by its own individual volume control. The oscillator arrangement is practically identical for the two models, the spinet using the sharing principle of the single-manual.

Chapter 16

Installing and Servicing Instruments

A few of the commercial electronic musical instruments such as the Solovox, Organo, and spinet organs can be installed nearly as simply as a radio or phonograph just by plugging in the power and perhaps attaching a keyboard. Installation of an organ with one or more separate tone cabinets is not so simple and to a very large extent the quality of the sound and the usefulness of the organ depends on the care and skill employed in its installation. In almost every case the local representative of the organ manufacturer makes the installation or at least is present to give advice. Outside metropolitan areas, however, these people are often organ dealers only as a sideline and they do not have the knowledge and experience of their confreres who have training and spend full time on the job. For that reason, the purchaser of an organ should read the comments in this chapter; he should apply them himself if he does the installation job, and he will at least be in a better position to see that he is getting what he should if the manufacturer's representative is in charge.

SELECTING AN ORGAN

The minute we get away from the concept that an "organ" is nothing but a bunch of oscillators we must fall into the much more reasonable and correct view that an organ is a musical instrument which, in the hands of a capable player, must furnish means for subtle and artistic expression of music, an appeal to the emotions. In addition, many electronic organs are used to replace the function of a pipe organ and for that reason must be made to sound as much as possible like one. To achieve these results it is necessary to consider carefully the selection of type and model in relation to the size and type of room or auditorium, the exact function the instrument is to fulfill, and the available budget.

The organ has traditionally been used in religious worship and even today most organs go into churches and synagogues. In the smaller church its principal function is to accompany the hymn-singing of the congregation, and the available organist, usually a member of the congregation, is not often a professional organist. Rarely if ever would the organ be used for concert work of any kind. The budget for small-church organs is usually small – though sometimes surprisingly small congregations take a most unusual pride in their organ and procure an elaborate one.

For the small church with minimum budget a spinet organ is often very satisfactory. They are made by Baldwin, Hammond, and Wurlitzer. Conn also makes the Connsonette but its usefulness for church work is questionable. For a very minimum expenditure a Lowrey Organo may be procured and attached to a piano, but here again this is not really a church instrument and it will probably not be found satisfactory for the purpose in most cases. All of these spinets have speakers built into the consoles, but in most cases an external speaker may be added either at installation or later. Incidentally, one of the advantages of a spinet is that if it is mounted on casters or a castered dolly frame it can be wheeled into other rooms and used for church social occasions.

The medium- and large-size church is usually desirous of something larger than a spinet. Any of the organs described in this book and other models of the same manufacturers are suitable. Any of the organs is as satisfactory for a very large cathedral as for a medium-size church from the angle of sound volume, since additional power amplifiers and speakers may be added as needed.

The main question, aside from price, is tone quality and versatility. Any electronic organ can produce good approximations of the traditional church tones – the diapasons and so on – which are useful for playing simple interludes and hymn introductions. For accompanying congregational singing reed tones are desirable and most organs can produce them. The Hammond, which is not really imitative of the full tonal range of a pipe organ (it is a distinctive instrument which is capable of many tones not to be found on pipe organs) does not produce real reed tones, though it can have sufficient sharpness of tone to ride above the singing. For more complex playing, such as may be expected of a professional organist during interludes, postludes, and perhaps in concert work, the various models may be compared by someone on the organ purchase committee who knows organs.

In this connection, it too often happens that organs are bought by people who know little or nothing about them. This is not a

reflection on the ability or honesty of these people, for after all it is the members of the congregation, represented by the purchase committee, who will pay. But it is far and away the best idea to include the organist, if he is well experienced, or an independent consultant if he is not, in the party which visits the organ showrooms. He should require the salesman to demonstrate every kind of playing of which the organ is capable and to comment specifically on the way his organ will fulfill the particular purpose in the particular auditorium. Then the organist with the purchase committee should himself spend some time playing the instrument and feeling out its resources.

The question of long-period reliability should and usually does come up. Unfortunately, it has been the practice of a few organ salesmen to run down their competitors unfairly on this point. It has been stated, for example, that vacuum-tube oscillators "always get out of tune." That is not true. They do require periodic tuning just as does a pipe organ or a piano, or reeds. When a vacuum-tube organ does require retuning any piano tuner can do the job in about one fourth of the time he would take to tune a piano (except for the Connsonata, with which tuning takes longer since many oscillators must be adjusted). If a piano tuner is not available, a Stroboconn may be kept on hand and then anyone at all can do the tuning. But except in tropical climates tuning is not a problem.

The same is true of the rest of the electronic circuits. No electronic device will operate forever without maintenance. Organs being designed for reliability, use oversize, overrated components and breakdowns are rare. When they do occur either the local representative will provide service or a good radio serviceman can do the job, with the help of this book and the manufacturer's service manual.

The question of electronic versus pipe organs sometimes comes up. No reputable electronic organ salesman today will say flatly that his instrument outstrips a really fine pipe organ in tonal quality and musical resources. He may say that it is better than the average pipe organ – and he may be right. It is certainly cheaper than any pipe organ of equivalent resources, and requires less space and fewer structural alterations at installation.

But the number of really fine pipe organs, either in this country or in Europe, is strictly limited. The average church organ is an instrument of little if any distinction and many are outright dogs (with apologies to all good canines). In addition, some are old and rundown, and cry for replacement.

The American Guild of Organists has yet to call an electronic organ an organ in its official publication. It uses the word "electrotone" so that nobody will think the AGO would dignify a piece of vacuum-tube apparatus with the same name as that traditional assembly of pipes. In the early days of electronics there was reason for this rather self-conscious distinction. Today there is not. Nobody who is musically minded and listens to organs a great deal will confuse the Aeolian-Skinner in Salt Lake City with a Baldwin electronic. But the same Baldwin – or Minshall, etc., etc. – compared with a *standard* organ will come out even or better *as an organ,* not as a novelty or an "electrotone." Couple this with the tremendous price differential and it is easy to see why electronic organs are making their way into more and more churches, concert halls, auditoriums, and homes, despite the fact that nobody claims the pipe organ will soon be dead.

ACOUSTICS

An organ must be installed correctly to give the most in tone and coverage. The first consideration is power, which must be adequate to insure, not only that the entire church or auditorium will be covered, but that the listeners will be able to hear the organ over other sounds such as congregational or choir singing.

The manufacturer's representative will usually estimate the needed power on the basis of his experience, since factors enter into the question which make calculation difficult or impossible. A very "live" auditorium of plain rectangular shape with a flat ceiling requires the minimum of power for its cubic content. A dead room, with drapes or sound-absorbing walls and carpets and with irregularities of shape such as alcoves requires the maximum.

A very important factor is the occupancy of the auditorium. For practical purposes an open window is considered to absorb 100 per cent of the sound striking it. A single person absorbs about as much sound as 4 square feet of window area. So it can be calculated roughly that when an auditorium is filled with 1,000 people the effect is as though 4,000 square feet of open window had been cut into the walls, compared with no windows with an empty auditorium. Thus, in judging the effect of a completed installation, the job should not be given a final approval until the organ has been used under typical conditions with an audience of maximum or near maximum size.

Even distribution is another important point. People sitting in one part of the auditorium ought to hear the organ about as well as those in any other part. For long, narrow rooms this is not a great problem. But for square or wide rooms, it may necessitate at least two sound sources. Irregular rooms and those with alcoves or small chapels, etc., may require low-power booster sources locally.

Reverberation is almost essential for any organ except one used entirely for popular music. This means that the sound should not reach the ear directly from the loudspeaker with no other path. It should bounce around walls, ceilings, and other surfaces and be a little bit "muddled up" for best effect. The pertinent quality of any room is estimated by making a rough measurement of the reverberation time, which is the time required for a tone to die away when the actual sound source has stopped emitting tone. For churches a reverberation time of about one second is the minimum desirable and it should reach at least that figure when the auditorium is at least two-thirds full.

Reverberation chambers are very commonly used when the auditorium itself does not have sufficient reverberation time. This is an

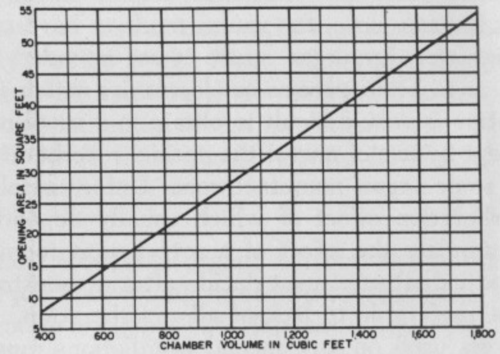


Fig. 16-1. Suitable dimensions for chamber volume and related opening size.

enclosure lined with some rigid material such as concrete or tile, with an opening into the auditorium. The speakers or tone cabinets are placed within the chamber and the hard walls make the sound bounce back and forth many times before emerging through the opening. The key factors in the design of a reverberation chamber are the volume of the chamber and the area of the opening. The graph of Fig. 16-1, furnished by Hammond, shows typical dimensions for these two figures for chambers with dimensions in the ratio of 2:3:4½. In practice the chart will also be found usable with other ratios. The opening should, if possible, be on the largest-area wall of the chamber and as near the center of the wall as possible. The tone cabinet or speakers should be placed in the chamber so that the sound is directed to a wall or ceiling of the chamber rather than toward the opening.

Considerable latitude for originality in laying out sound sources is possible. The standard tone cabinets sold by the manufacturers need not be used. Depending on the auditorium and the ingenuity and the ambition of the people involved, speakers may be mounted on large flat baffles at various points and in various combinations. In every case, however, it is especially vital to see that construction is exceptionally solid, with very heavy wood used throughout and anchored as firmly to the structure as though the building's stability depended on it. Nowhere in music reproduction do we come across sustained bass tones as we do in this kind of organ installation and if it is not solid beyond the shadow of a doubt it will rattle and may even eventually destroy itself.

Installing an organ in the home or in a very small auditorium is much less of a problem. Most homes do not allow for a reverberation chamber and standard tone cabinets are used, simply being placed wherever convenient. It is always desirable to place the tone cabinet in someplace other than the same room as the player and listeners; if the speakers are too close, the sound tends to have an unfortunate "electronic" character, for organ music is not intended to come from a point source or to be as precise as electronics makes it.

Reverberation is very difficult to obtain in a small room – usually impossible except by use of one of the special reverberation tone cabinets made by some organ manufacturers. Unfortunately, these cabinets, the reverberation effect of which usually depends on springs, do not really simulate the effect of a reverberant room to any great degree. The individual user must decide after demonstration whether or not they are preferable to lack of all reverberation.

The speakers used on any special installations must be as large and heavy as possible. Three to six 15-inch speakers are not too many and a larger number of good 12-inch speakers is even better in some cases. Treble is not often a great problem, but if tweeters are used, they should be experimented with and a separate volume control placed on them to avoid emphasis of treble beyond the organ's normal tone. If special amplifiers are used much attention should be paid to power handling rating *at the lowest bass frequencies.* Most standard amplifier power ratings are given at a medium frequency and the bass power-handling ability is usually far less.

SERVICING

It is, of course, impossible to show how to service all the electronic organs in one chapter. There is one trait they all have in common, however – they tend to analyze their own troubles. If only one tone is bad, the trouble is probably in one easily identified oscillator or frequency divider. If the organ uses dividers this diagnosis is made certain if all notes of the same letter below a certain octave

are bad. This same kind of reasoning can be carried on to find almost all faults with the exception of those which are in circuits like the main amplifier, and even then the fault can at least usually be traced to the amplifier. The key to trouble evaluation is simply a knowledge of how the organ or instrument functions (to be gained from a study of this book) plus a little common sense.

Most organ manufacturers provide some sort of service for their instruments and some conduct factory service schools which are attended for a short time by all field technicians. When such factory service is available it is usually preferable to locally engaged servicemen, who too often are simply radio and television repairmen who depend on past experience and second sight rather than any real knowledge of electronics. If, however, an owner has some technical knowledge, or a church or organization member who has such knowledge can be found, it is an excellent idea to have him read and understand how the instrument works and observe the regular serviceman (if any) in his work. Most repairs are simply tube replacements which can be made in three minutes if someone with a clear head and an understanding of the instrument is on hand to point to the right tube. Spare tubes should always be kept close by.

TUNING

Tuning most electronic organs is, from the physical standpoint, a much easier job than tuning most other instruments, for it is simply necessary to rotate a knob. The Hammond and other rotating-wheel organs cannot be tuned and depend on constant power-line frequency for constant tuning. An organ which uses reeds, such as the Wurlitzer, is a more complex job, since the generator must be disassembled to some extent and the reeds filed. This is a job which should be avoided as far as possible by novices, but fortunately the reeds do not often have to be touched up.

The musical aspects of tuning an organ with locked frequency dividers, such as the Schober, Baldwin, Minshall, or Kinsman, are easily taken care of without a professional tuner, though if the owner wishes, he can call any competent piano tuner, who will be able to do the job in less than 30 minutes.

Two methods are available for the amateur tuner and either is quite satisfactory. The first is to employ the demonstration record sold by mail by The Schober Organ Corp., 2248 Broadway, New York 24, N. Y. for $2.00 postpaid. This record contains twelve tones played for about 40 seconds each. When a good, steady turntable is used, the organ is simply tuned to the recorded pitches by the old standard zero-beat method, which anyone can do without training or a "musical ear." If an oscilloscope is handy it will speed up the job and may make it somewhat more accurate; one scope input is connected to the record player and the other to the organ output, while each tone is tuned until the screen pattern stands still.

The second method is an extension of the method used by most tuners – the counting of beats between musical fourths and fifths. Because of the tempering of the modern musical scale, the musical note which approximately corresponds to the third harmonic of one below it, is actually slightly flat; for instance, the second G above middle C, at 784.0 cps is slightly flat of an accurate third harmonic of middle C at 261.7 cps. When middle C and middle G are played together, the actual third harmonic of the C and the second harmonic of the G, which are about 1.1 cps apart, cause a beat or periodic pulsation of volume. If we can assume that the C is properly tuned, then the G can be tuned by adjusting it until the number of beats per second – 1.1 – is correct (and being sure that the G is flat rather than sharp of zero beat). As a matter of practice for the amateur, greater accuracy is obtained by using a stopwatch and counting the number of beats per 10 seconds.

The table below shows the correct number of beats in 10 seconds for each of the tones when it is played with one already tuned. Begin by tuning the middle A at 440 cps to unison (zero beat) with a tuning fork or with WWV. Now hold down the A and the E below it, and tune the E first for zero beat. Then flatten the E until you can count just 15 beats in 10 seconds. Next hold the E and the B above it, tune the B to zero beat, then flatten it until you get 11 beats in 10 seconds. Continue down the line in the same way. Use only the notes in the middle octave C through B. Use an 8-foot or unison stop, preferably one with some, but not too much, harmonic content, such as a flute which is not a sine wave or a very muted string. When you finish, the D-A combination should beat at close to 10 beats in 10 seconds. If it does not, do not retune the A; start over again.

TUNING TABLE

Sound	Tune	Beats in 10 Sec.
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	check only	10

The organ should, of course, be thoroughly warmed up before you start, and the vibrato must be off. An oscilloscope can be used to facilitate beat counting. Simply connect the vertical input to the organ output and reduce horizontal gain to zero so that you have a narrow vertical line. Each beat will cause the line to lengthen and shorten. An a.c. voltmeter with a scale suitable to measure the organ output will indicate as well by a pulse of the needle for each beat.

You can, of course, do the job even more quickly and accurately with a Stroboconn or Strobotuner. However, these instruments are generally too expensive for the organ owner to purchase and are often not available even to organ servicemen. They are particularly advisable where each note must be tuned individually, as in Conn, Allen, Thomas, and similar organs. However, such organs can be tuned by the methods given above; the middle octave is first tuned as shown, then each note in the other octaves is brought to zero-beat with the corresponding note in the middle octave. The job is rather long, however, and the time necessary for the tuning may allow some frequency drift which may upset the final results somewhat.

Chapter 17

Constructing An Instrument

THIS third and final section of the book is devoted to the construction of electronic musical instruments. The first and second sections gave some general principles of music and musical electronics. Now in the following chapters we shall use the knowledge gained from the study of principles and commercial practice to apply some actual elbow grease – and perhaps some of the reader's own originality, imagination, and design skill – to bring electronic music into three-dimensional life in the living room or the community church or auditorium.

Chapter 18 describes the Thyratone, a monophonic (one-note-at-a-time) instrument designed by the writer. Chapter 19 gives full instructions for making the Electronorgan, a complete electronic organ potentially equivalent in musical quality to some of the finest instruments on the market. For the construction of these two instruments the reader needs little originality and not a great deal of money, but time and hard work are essential ingredients, plus the familiarity with reading schematics and handling the tools of the trade required by any instrument builder.

The remaining chapters are provided for those readers whose skill extends beyond constructing according to fully laid out plan. They give individual ideas and circuits concerning tone generators, tone coloring, amplification, control, and instrument layout which can be used in any of a thousand or more combinations to yield exactly the design which will most nearly fulfill a particular need and desire. The task of selecting individual circuits and tying them together is left to the designer. Most of these circuits have never been published before; many of them come from the fertile brains of the thousands of inventors whose ideas are patented every year.

It is not surprising that among the many devices made possible

by electronics, musical instruments rate high as construction projects. The possible reward is very attractive – a means of musical expression far more versatile than any standard instrument except the pipe organ – and there seems to be a highly intriguing quality about the idea of making music from resistors, capacitors, and tubes. The reward, however, in common with most other rewards in this world, must be earned. And the earning process is long, painstaking, and expensive. In an electronic organ – the most popular project – the basic fact that confronts one is that – at the very least – 61 separate tones must be generated, and they must be available in any combination. No matter how hard we may try to circumvent it, 61 or more individual tone generators must be built. True, there have been a few patents on ways of making fewer generators do the job, but they are not particularly practical. Witness the fact that no commercial instruments (except the simplified Hammond Chord Organ) have taken this path to economy. Sixty-one of anything involving vacuum tubes is a lot, especially when this only starts the job – there is still the problem of what to do with the tones once you have them.

The keying system is one of the worst headaches, and not just because of the minimum of 61 switches per manual. Audio circuits have the perverse quality of clicking and popping when keyed, and of sounding like code-practice oscillators when keying is practically instantaneous.

All in all, building an organ is not a task to be considered lightly. No matter what the design, it entails many hours (hundreds of hours is more like it) of hard labor, much head-scratching in design, unavoidable hours of experiment, and enough dollars to cover the cost of an enormous list of parts.

The writer has given this warning to all the many correspondents who have written "Please send me a complete set of instructions for building an electronic organ." Until recently no "complete set of instructions" has ever been available. But now that one is available (The Schober Organ Kit, Chapter 19) the warning still goes.

DECIDING ON REQUIREMENTS

The series of articles on which this book is based began with a statement to the effect that simply assembling some oscillators of the requisite frequencies does not yield a musical instrument. That has been amply demonstrated by the techniques used to obtain real musicality in the commercial instruments we have described in such detail in the second section of the book. Having decided now to construct our own instrument, let us try to determine what the end results should be and what means are best used to attain them.

The first requirement is a set of tone generators. It is desirable to have at least five octaves (plus the top *C*) of keys, establishing 61 notes as a minimum. It is best to have an extra octave – a 16-foot register – at the bottom to give a good bass foundation, whether or not a pedal clavier is included. It is possible to get by without an extra octave on top (as in the Baldwin model 5) and to include a 4-foot register by having the last octave of keys repeat the notes of the fourth octave in the high register. But if it adds no excessive complication, the extra octave is useful. With a total of seven octaves and top C we have 85 notes to generate; without the top octave, 73.

It is not easy to use 73 individual, unsynchronized oscillators, as in the Connsonata, because of tuning difficulties and drift. Usually the logical decision is octave strings, with twelve sets of frequency dividers or multipliers, each set generating as many notes as there are octaves of generators in the organ – from five to seven. More than seven (used to provide 2-foot ranges and higher) rarely contribute enough to the musical results to justify the effort.

Each divider string must begin with a sure-fire self-excited oscillator stable within ¼ of 1% (0.25%) accuracy or better. Design should begin with laboratory-bench experiments with various oscillator types. A reference source for tone pitch is a necessity, since no variable-frequency audio test generator is accurate enough. A newly-tuned piano is good; a Hammond organ is even better. A Stroboconn is best but not generally available. A good-quality harmonica can be used if it is first checked with a well-tuned piano. This reference source will provide the essential accuracy for final tune-up. For the design procedure it need only be stable, not necessarily exactly on pitch.

After a sample oscillator is built (breadboard style is recommended) it should be tuned to coincide with a note from the reference source. Tune very carefully, either by ear – counting beats until the new oscillator is right on the nose – or with an oscilloscope Lissajous pattern. Let the oscillator run for several hours, then check its frequency again. Turn it off and let is cool, then turn it on; after five minutes or so check it again. In each case the pitch should be so close to the reference that the note is musically acceptable. The figure of ¼ of 1% accuracy is the *outside limit,* and it means 1 cycle permissible error at 400 cycles or 10 cycles at 4,000 cycles, which is not much. If the first design does not measure up, discard it and try again. Never skip this tryout procedure. You may copy someone else's design electron for electron and still have unstable oscillators.

Before deciding on the system of dividing or multiplying frequencies, the other organ requirements must be outlined. First, what method of tone coloring will be used? If harmonic synthesis

(Hammond-Organ style) is wanted, then each generated tone must usually have sinusoidal waveform, which is not easy with oscillators that must be synchronized. Perhaps the most practical way to obtain sine waves with synchronized oscillator strings is to generate some other kind of waveform and then use a low-pass filter in the keying system between each octave group of tones and the rest of the system. This, incidentally, will kill key clicks as well.

You may decide to use a phonic-wheel system, as in the Hammond, but here again bench experiments must be made with sample tone wheels and pickup magnets to determine the best metal

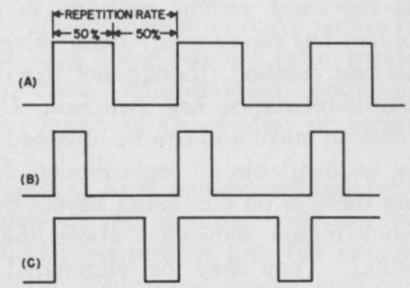


Fig. 17-1. Square waves should not be symmetrical.

and the precision with which each wheel must be made. This method is suitable only for people with marked mechanical-design ability and the necessary tools and facilities.

If the *formant* method (used in the Baldwin and Minshall organs and, in simpler form, in the Solovox and Organo) is chosen, the tones must have a complex waveshape, the more complex the better. Sawtooth waves are best because they contain a smooth progression of both odd and even harmonics, but square waves are useful too. The pulse width should be greater or less than 50 percent of the repetition rate to avoid giving all tones a stopped or woodwind quality. (See Fig. 17-1.) Square waves can be obtained from multivibrators and flip-flop circuits, while sawtooth signals usually entail relaxation oscillations of one kind or another.

You may prefer a formant and mixer system like the one in the Connsonata. The Connsonata uses individual oscillators which yield both sine and pulse signals. This can be done with synchronized strings by using octave filters as suggested above to obtain sine waves after keying, and by using separate key switches followed by differentiators (high-pass filters, effectively) to obtain sharp pulses. Don't use frequency dividers or multipliers which will work only at a single frequency. Any system, even though it is for only one note of the twelve, should function well over a range of at least three semi-tones. Otherwise it may as well be discarded as not stable enough.

The keying system is an important decision. In a pipe organ each stop is normally available in a certain register – 4-, 8-, 16-foot, etc. – and couplers are provided to give the same stops in other registers as well. The Baldwin organ is an electronic example of this practice; so is the Electronorgan. It is also feasible to key all tones at one or more registers and have stop filters passing tones of any register selected, as in the Minshall.

Key click is the most serious problem in almost any keying design. The problem has never really been solved in an ideal way, but probably the best method (though not the easiest or cheapest) is Baldwin's variable-resistance key switches. Unfortunately, these switches are not easy to make and can be obtained only from Baldwin, though they may be available as replacements from some Baldwin dealers. Assuming there is no d.c. being keyed, the clicks take place in a high-frequency region somewhat above the frequency of the highest fundamental. They may be eliminated by octave filters, as in the Minshall, though this also does away with some of the harmonic content of the signal waveform. Vacuum-tube keying is practical from the schematic viewpoint but means the addition of at least one tube for each section of each key switch assembly. Vacuum-tube keying requires at least a two-section *R-C* delay network at the grid. The rise time must be rather slow to avoid the thump caused by plate-current rush unless well-balanced push-pull tubes are used.

Tone-coloring methods are described in this book and the one chosen is a matter of preference. The designer should listen to good pipe organs and recordings and judge results by ear.

Amplifiers should have exceptional power-handling capacity in the low-frequency region. This means good output transformers and plenty of negative feedback. Twenty watts is about the minimum power output for any organ, even in small rooms. Speaker systems should have good bass characteristics; it is usually best to use one to four heavy speakers with at least 12-inch cones. Tweeters are rarely necessary, though they may add something to a good instrument. Intermodulation distortion must be kept low, for sustained multiple tones produce very annoying beat tones in a nonlinear system. Frequency non-linearity, of course, will change the tone colors of any kind of organ, since it has the effect of a formant filter.

MATERIALS AND CONSTRUCTION

In choosing any design or making one out of fragments from commercial instruments and/or other ideas, always consider whether the parts are readily available or manufacturable. The number of letters received by this writer inquiring as to the number of turns on the coils in the Organo or where to buy multiple transformers like those in the Baldwin is staggering. While the ambition of the correspondents is to be commended, it is simply impossible to produce such special items as the multiple transformers; and the number of turns on the coils was not given because the information was not available – manufacturers give out information for purposes of description, not to enable readers to make copies of their products. While this may not stop some readers from making the copies – and more power to them – consideration should have been given before choosing the design to the matter of making some of the special parts.

Coils of any ordinary kind can be wound by anyone and the right number of turns arrived at either with an inductance bridge or simply by cut-and-try connection into the circuit. Special transformer cores are harder, but some people with adequate facilities can make them. The same is true of large multiple switches such as those used for couplers in the Connsonata, and for other parts. But if the constructor has only average facilities, he had better stick to a design requiring nothing special or else bypass special parts with design modifications. The Electronorgan was a good example of this. The principle of operation was chosen, then a design was evolved using strictly standard parts, though the use of special ones might have made for a slightly more compact or versatile instrument.

On the whole, the parts required for an electronic musical instrument are the same old standbys used for amplifiers, receivers, and so on – resistors, capacitors, tubes, transformers, etc.

Every organ includes a number of purely mechanical complications, the most obvious of which are the keyboards or manuals. The simplest solution is to get a second-hand pipe-organ console and build everything inside it. The keys, action, pedals, housing, stop tabs, and so on, are already provided. Consoles are not particularly hard to find, especially in the larger cities. Look up organ repairmen in the telephone book and ask them to notify you when they get a console. Most of them will have one sooner or later. Another source is a local dealer in electronic organs; he often will sell or give away the console of a pipe or reed organ he replaces.

The console should be examined carefully to see that the action is in working order or easily repairable. Damaged key ivories are easily replaced either with new ones or those peeled from another used keyboard. The pedal action should also be examined. If possible, obtain a console with at least as many stop tablets or knobs as your design calls for. Extra ones are not easy to add, especially if good appearance is a factor. A combination action is a desirable feature; the pneumatic actuators usually can be removed and solenoid actuators substituted. Reed-organ consoles are acceptable, but usually require more work for adaptation. The keys are usually much longer than necessary, since they had mechanical work to do rather than simple electrical contacting, and the case may be deeper than required. The case can be cut down, of course.

A console can be built by a handy woodworker, the only outside requirement being the keyboard. Here again an organ repairman may be able to supply a keyboard from an old organ or harmonium. Be careful to get a C-to-C keyboard, since some harmoniums have F-to-F manuals. Figures 17-2 and 17-3 show top and bottom views of a Steinway piano keyboard purchased for very little from a piano-repair concern. The keys are in almost perfect condition, and were cut, after the photo was taken, to eliminate the hammers and pins, leaving only enough behind the pivots to actuate switches.

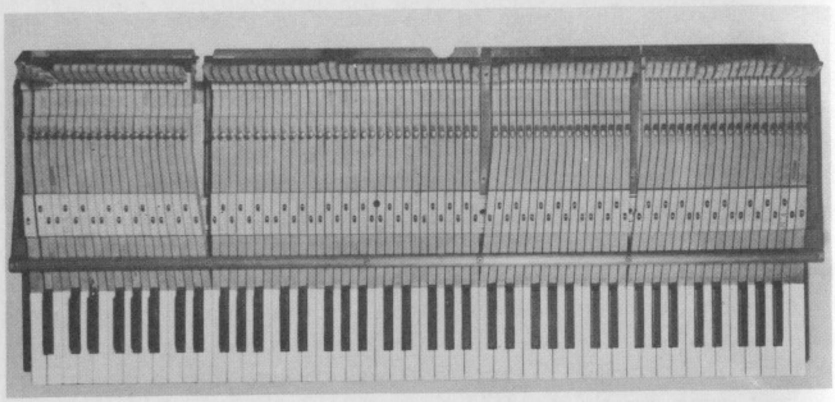


Fig. 17-2. A piano keyboard which can be used for an organ manual after suitable alteration.

Key switches may be made in several ways. Possibly the easiest is to purchase flat relay-contact blades and assemble them in the desired positions, either under the key fronts or above the rear ends of the keys. Guardian Electric Manufacturing Co. makes a contact parts kit (No. 200-3) containing an assortment of blades, contact rivets, washers, and spacers, which is useful for this purpose. Springs and contact strips also may be used, as in the Thyratone. Nickel and

phosphor-bronze contact spring wire, available from organ repair men, also may be used to make light and small contacts.

Generator systems should be built and mounted so they can be removed in sections for servicing and adjustment. The logical arrangement is to put the string for each note on a separate chassis, making twelve in all. The chassis need not have sides and skirts; often a simple flat piece of metal makes a fine chassis for this purpose (see Chapter 19). It is especially worthwhile to mount the parts in such a way as to make it possible to reach them while the organ is operating. Many experiments with values and circuits undoubtedly will be made after the instrument is complete, especially in the tone-coloring section.

The EXPRESSION pedal or pedals and the CRESCENDO pedal, if any, are important design points. Ordinary radio volume controls are not

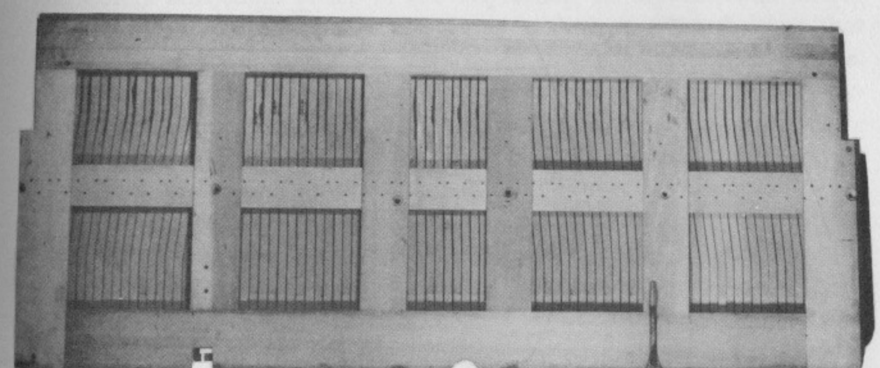


Fig. 17-3. Bottom view of the piano keyboard.

suitable, even though it may be possible to link them to the pedals. The EXPRESSION pedal undergoes hundreds of times as many operating cycles as radio or amplifier controls and must be especially rugged. Whatever control is used must at least be a heavy wire-wound unit. The best type is the contact-stud variety, like a broadcast attenuator, available from General Radio, Daven, Cinema Engineering, and other companies. High-impedance types are usually undesirable because the necessary lines cause high-frequency loss and because any contact disturbance is more obtrusive in a high-impedance circuit. The expression control must not bring the volume down to zero in the minimum position. A range of 30 db or so is sufficient. At the same time the instrument must have some kind of preset volume control so that the maximum volume can be adjusted to suit the room and the listeners.

Separate tone cabinets are superior musically to having the speaker in the console. A speaker cabinet can be designed for good baffling and preferably should be placed in a corner of the room or auditorium, with the speakers facing the corner with good sound dispersion. A better installation method is to have a hard-walled organ chamber, perhaps ten feet square or more, with the speaker cabinet concealed within it and louvers opening into the auditorium. Excellent results may be obtained also by placing several separate speakers at intervals along the front of the room, but they must be correctly phased to prevent interference.

Chapter 18

Building the Thyratone

THIS chapter describes in detail the design and construction of the Thyratone, an experimental instrument designed by the writer. Its name is derived from the fact that the tone generator tubes are thyratrons operating as relaxation oscillators exactly as in most oscilloscope sweep generators. The Thyratone has certain weaknesses, principally the fact that the master oscillator, being a thyratron, is not as stable as would be desirable. However, it could easily be replaced by a hard-tube oscillator.

The Thyratone's primary virtue as a construction project is that it is a truly *musical* instrument. It not only provides a series of tones of the correct pitches, but it includes genuine tone-shaping circuits to produce uncannily realistic imitations of orchestral instruments and organ qualities. Its voices are available both singly and in combination, so that the range of tonal effects is very large. In addition, it has separate stops for 8, 16, and 32-foot registers.

WHAT IS THE THYRATONE?

The Thyratone is a monophonic or solo-type instrument, appearing, at first glance, to be much like the Hammond Solovox. It is similar in that its three-octave keyboard of foreshortened keys (Fig. 18-1) may be fastened to the front of a piano and the instrument can be played at the same time as the piano. Another likeness is that only a single key may be played at a time. But there the resemblance ends.

The block diagram of Fig. 18-2 gives an over-all view of the instrument. There are three tone generators operating at octave separation. The 8-foot generator produces three octaves of tones ranging from C^{40} (middle C-261.7 cycles – see frequency chart on page 7 to C^{76} (2,093 cycles). The 16-foot generator produces a three-

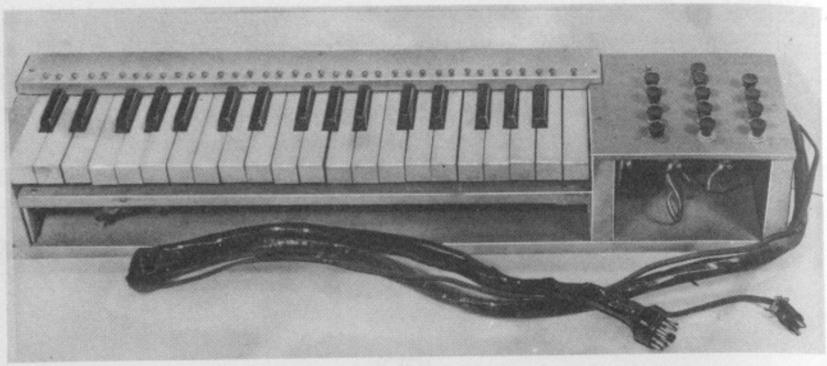


Fig. 18-1. The Thyratone keyboard.

octave range from C^{28} (130.8 cycles) to C^{64} (1,047 cycles). The 32-foot generator produces pitches from C^{16} (65.41 cycles) to C^{52} (523.3 cycles). The nomenclature for the ranges is taken from organ practice for convenience.

When the Thyratone keyboard is fastened to the front of a piano the lowest key coincides with the position of middle C on the piano. Therefore, in the 8-foot range, sounding this note will produce an actual pitch of middle C. Pressing the same key but using the 16-foot range, the tone heard is one octave below what one would normally expect from that key. The 32-foot pitch is an octave below that. The Thyratone therefore has a total range of five octaves (plus one note – the top C).

The generators are all keyed simultaneously so that pressing any one key produces three notes an octave apart. The tones from each generator are fed to a series of *L-R-C* filters which alter the waves in such a way as to give a more or less close approximation of a standard organ tone. The three ranges are filtered separately, as Fig. 18-2 indicates, so that a bourdon tone, for example, is produced only in the 32-foot range, and an oboe is available only in the 16-foot register.

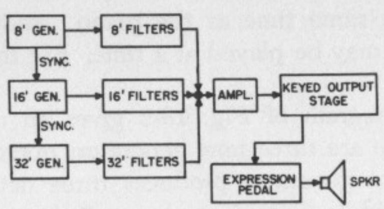


Fig. 18-2. Block diagram of the Thyratone.

There are four 8-foot tone stops, six 16-foot stops, and two 32-foot stops, a total of twelve tone qualities or stops in all. One or several may be in use simultaneously to give any type of mixture desired, just as in an organ. The tone colors will be described later in the discussion of playing.

The tones from the outputs of all the filters are mixed and amplified, then fed to a push-pull output stage. This stage is normally biased to cutoff. The cutoff bias is removed each time a key is pressed and an *R-C* time-constant network provides keying delay to eliminate clicks and thumps and give a good musical attack and decay. An expression pedal, consisting of a foot-operated 8-ohm T-pad, is placed

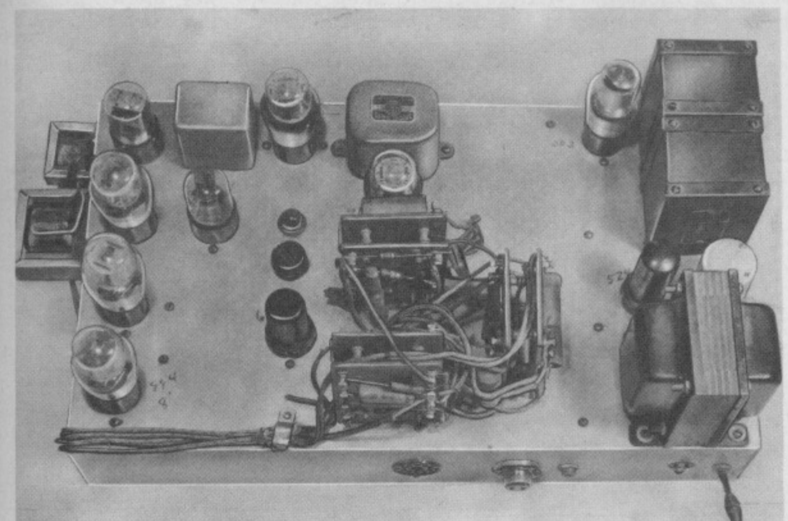


Fig. 18-3. The electronic chassis.

between the output transformer secondary and the speaker. A preset volume control in the amplifier section allows the player to set maximum desired level so that the pedal can be operated over its entire range.

WHAT IT LOOKS LIKE

The physical appearance of the Thyratone is illustrated by the photographs. Figure 18-1 shows the keyboard unit. There are three octaves of keys and, at the right, a control board. On the control board there are fourteen push-button switches to control the stop combinations and the vibrato. At the upper left is the a.c. power switch. In this experimental model the front has been left open for access; in the finished product, of course, it will be closed. The wood will also be finished and the hole at the upper right will probably be filled with another push button. The entire keyboard unit may be fastened to the front of a piano with metal brackets in the same way as the Solovox is mounted. Because of the nonavailability of compactly built keyboards the writer did not bother to keep the keyboard unit especially small. Other constructors should try to do better in that respect, as long-legged players may find that there is not enough room underneath.

The chassis appears in Fig. 18-3. The entire electronic equipment, with the exception of the generator-tuning capacitors, is mounted on it, and the keyboard unit serves only for control. Normally the chassis is mounted within an ordinary loudspeaker enclosure along

Fig. 18-4. Expression pedal in position on the floor.

with the speaker. Figure 18-4 shows the expression pedal in position on the floor. A cable from it plugs into the chassis. The keyboard unit connects to the chassis through a 20-conductor cable terminating in a standard Amphenol 20-pin plug. An additional two-wire line serves for the a.c. power switch.

THE TONE GENERATORS

Because of the type of tone-color filters used in the Thyratone, sawtooth waves are required from the tone generators. An additional requirement is three generators which will synchronize easily in exact octave relationships without having any of the synchronizing frequency appear in the output. In an experimental mood, 884 thyratrons (from which the instrument gets name) were chosen. Gas-filled

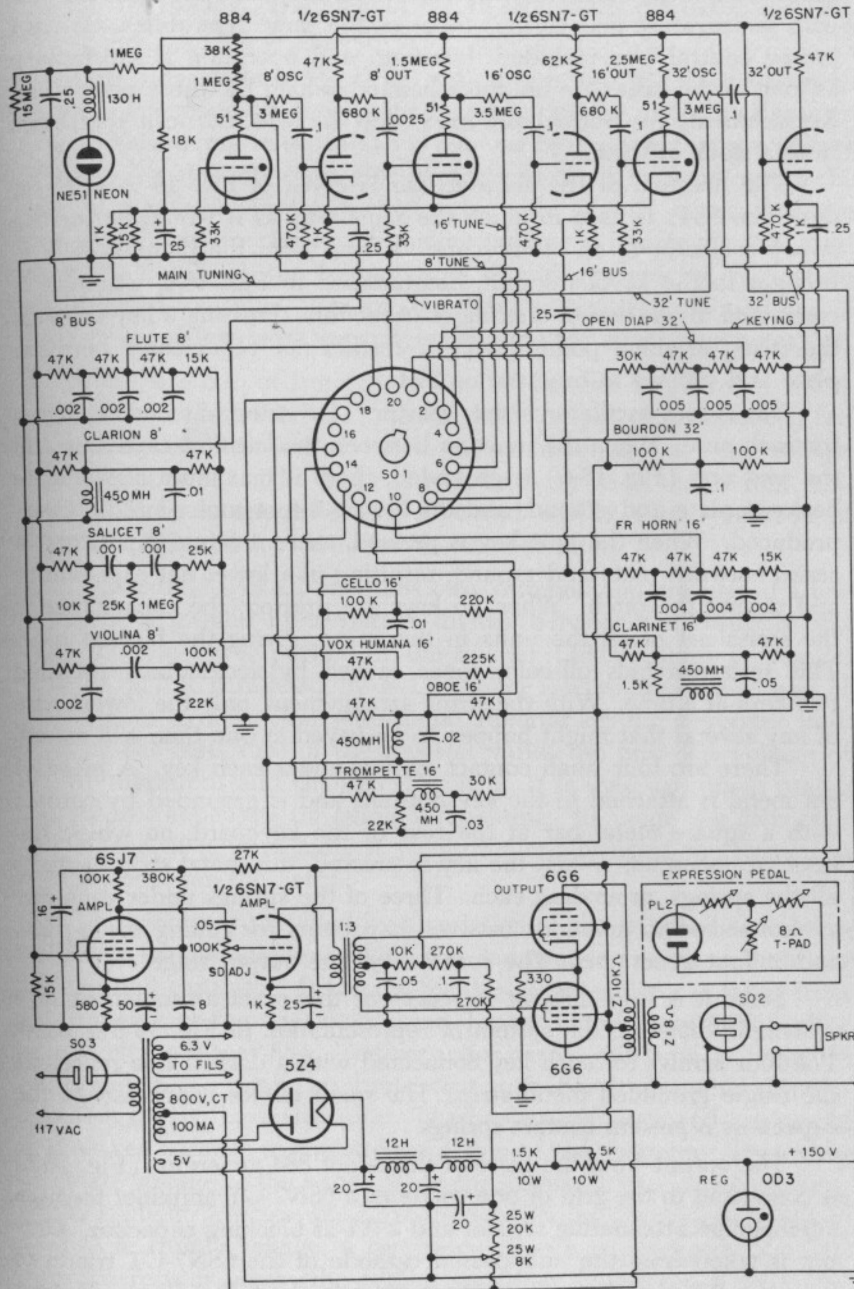


Fig. 18-5. Main chassis diagram.

tubes are not the most stable oscillators, as they vary in characteristics with temperature and various tubes of the same type differ. A main tuning control was included, however, and operation is satisfactory as long as the tubes are not interchanged among the three generators. Replacements for burned-out 884's must be selected on a trial basis from a stock of them.

The diagram of the main chassis is given in Fig. 18-5. Each of the three 884's is used in much the same way as it would be for the sweep oscillator of an oscilloscope. A lead from the plate of each is brought to the keyboard unit, diagrammed in Fig. 18-6, where it is connected to one end of a string of capacitors. Pressing a key grounds the string at some point, giving a certain net capacitance between plate and ground to tune the oscillator.

The 8-foot oscillator is the "master." Its string has one capacitor for each note. When the junction between the leftmost capacitor and the next one (Fig. 18-6) is grounded, there is maximum capacitance between plate and ground, and the lowest 8-foot tone (middle C) is produced. When the next key is pressed, there are two capacitors in series between plate and ground, resulting in a lower net capacitance and raising the pitch. When no keys are pressed, the capacitance is the series net of all the units in the string, giving the highest tone. This system avoids off-color tones caused by accidentally pressing two keys at a time. With the string arrangement, only the lowest note of any several that might happen to be keyed at one time will sound.

There are four small contact springs under each key. A piece of flat metal is attached to the key bottoms and is grounded by contact with a square metal bar at the rear of the keyboard, on which the keys are mounted. When the key is pressed, this metal strip contacts all the springs, grounding each. Three of the springs under each key are connected to junctions between capacitors for tuning the 8-, 16-, and 32-foot generators. The fourth keys the output stage.

Since it is very difficult to represent this exact arrangement in a schematic diagram, the system of representation in Fig. 18-6 is used. The four arrows for each key connected with a dashed line represent the single grounded metal strip. The small circles connected to the capacitors represent contact springs.

The output from the plate of the 8-foot 884 generator (Fig. 18-5) is connected to the grid of one triode of a 6SN7-GT amplifier through a 3-megohm attenuating resistor and a 0.1-μf blocking capacitor. Output is taken from the unbypassed cathode of the 6SN7-GT triode to form the 8-foot bus carrying 8-foot tones to the 8-foot filters. Output from the plate of the 6SN7-GT triode is fed through a 680,000-ohm attenuating resistor to the grid of the 16-foot 884 to provide synchronizing voltage. The plate of the 16-foot 884 is carried to the keyboard and a string of capacitors for tuning.

Referring again to Fig. 18-6, note that in the 16-foot string there is not a capacitor for every note but only one for every six or seven notes. This saving in capacitors is allowed by the fact that the 16-foot generator is synchronized. For each group of six or seven notes, the natural frequency of the oscillator is made slightly lower than the lowest note; when the synchronizing voltage is fed to the 884 grid from the 6SN7-GT the frequency is brought to exactly one octave below the 8-foot tone. For greater stability, constructors may find it wise to use a few more capacitors, say one for every four notes.

The plate output from the 16-foot 884 is fed to the other triode section of the first 6SN7-GT. The cathode output of the triode is fed to the 16-foot filters and the plate output is applied as syne voltage to the 32-foot 884. The latter is tuned exactly as is the 16-foot generator. It feeds one triode of a second 6SN7-GT, the cathode of which provides 32-foot tone for the 32-foot filters.

The cathodes of all three 884's are common and are placed a few volts above ground by a voltage divider between the B-supply and ground. The lower portion of the divider (15,000 ohms in parallel with 25 μf) is paralleled by a 1,000-ohm resistor, the lower end of which goes to a 50-ohm wirewound rheostat on the keyboard unit. Varying the resistance of the rheostat varies the grid-bias voltage of all the thyratrons and changes the pitch. It is used as a main tuning control to compensate for aging and heating. Its range is a little over a half-tone; if it were more it would materially upset the frequency spacing between the notes.

The vibrato is provided by a variation of the standard neon-lamp oscillator, which includes a high-value inductance as well as the usual resistor and capacitor. B-voltage for the neon oscillator is taken from the junction of the 38,000-ohm and 1-megohm resistors in the plate circuit of the 8-foot 884. The capacitor across the lamp is normally ungrounded. To produce vibrato, that capacitor is grounded through a switch on the keyboard unit. The oscillations produced are nearly sine waves because of the storage action of the inductor. They vary the plate voltage of the 8-foot generator slightly at the oscillation rate, which is about 7 cycles; since gas-tube oscillators change pitch with a change in supply voltage, a frequency vibrato is produced. The 15-megohm resistor across the 0.25-μf capacitor discharges it after the bottom end is ungrounded (when the vibrato switch is turned off). Without the discharge the neon will not oscillate again when the switch is closed.

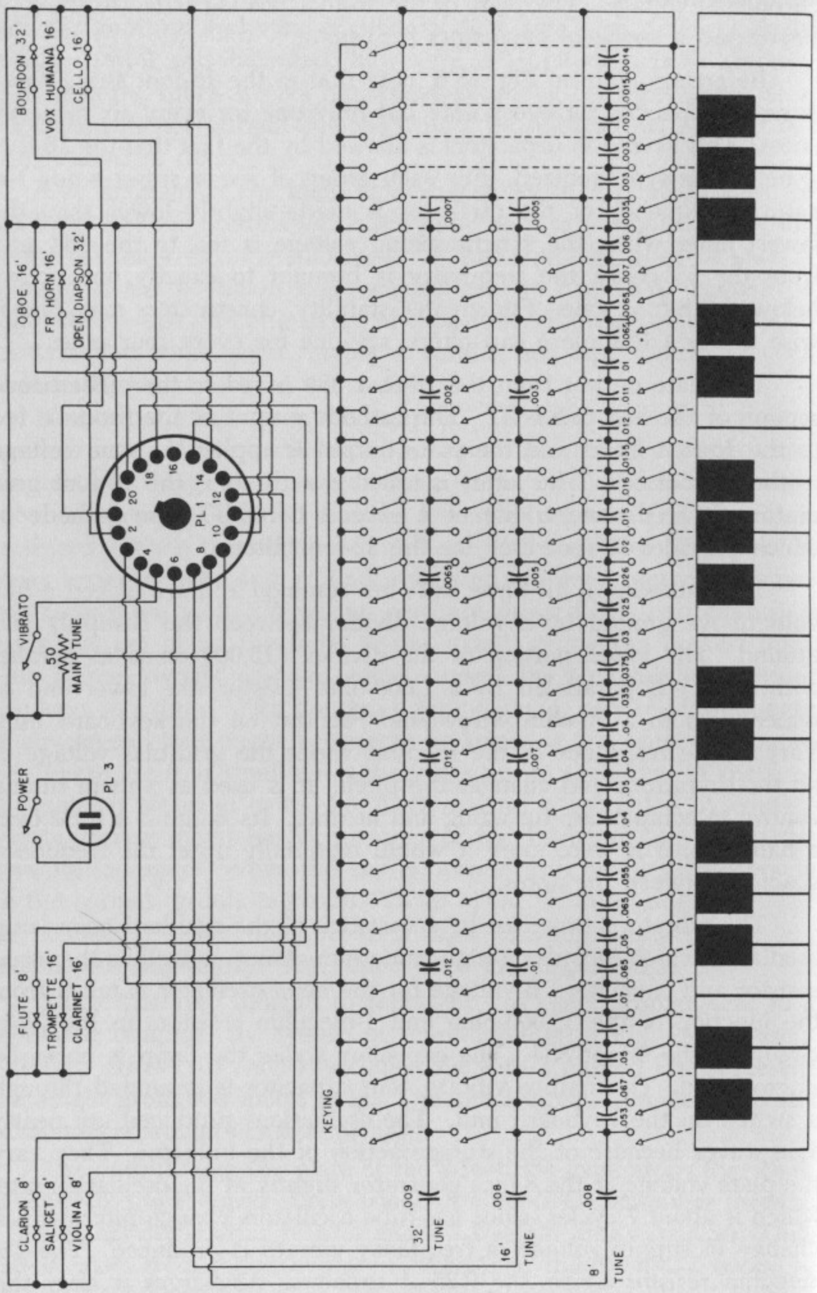


Fig. 18-6. Diagram of the keyboard unit.

TONE FILTERS

The formant filters are of the same type as those used in the Baldwin organ and in the Electronorgan. Their capabilities are not realized fully in the Thyratone because of the desire to avoid unnecessary complexities, but they do produce uncannily realistic imitations of many of the tones of a pipe organ.

The inputs of all the filters in each register are obtained from the corresponding bus and all filter outputs are paralleled. Because of the long lines involved in the Thyratone as it now exists, switching outputs to select tone colors or stops was found impractical. The stop buttons therefore are normally closed switches which short the filters. Punching a button removes the short on the corresponding filter and allows the tone color to come through. The short circuit is made in the "middle" of the filter so that it will not appreciably affect input or output busses.

AMPLIFIER

The outputs of all filters go to the grid of a 6SJ7. The amplified tones go to a volume control which is preset for the desired maximum level. From here the tone goes to the second triode section of the same 6SN7-GT used to feed the 32-foot bus. The plate is transformer-coupled to a push-pull 6G6-G output stage.

There are two reasons for keying this output stage. First, when no keys are pressed, all oscillators are tuned to their highest pitches by the capacitor strings. Second, a slow attack and decay must be provided so that the instrument does not sound like a code-practice oscillator. It is difficult to key a single-ended stage because, unless the attack is too slow for musical comfort, the rush of electrons from cathode to plate when cutoff bias is removed – even with a delay circuit – makes a thump in the speaker. This hazard is removed by using a balanced push-pull stage; the rush of electrons is in the same direction in both tubes and the two cancel in the output transformer (if the tubes are fairly similar).

To provide a negative bias, the bleeder of the power supply is tapped and the tap grounded. Thus the lower end of the bleeder is more negative than ground. In the model shown the power transformer produced insufficient d.c. voltage at the filter output (about 250 volts). Other constructors should use transformers with at least 400 volts each side of center-tap. This allows the tap on the bleeder to be moved up higher, giving enough B-voltage for reliable operation of the 0D3 (which provides regulated voltage for the thyratron tone generators).

The negative end of the bleeder is wired to the center-tap of the driver transformer through an *R-C* network. A lead from the network goes to the keying contact springs on the keyboard unit. When a key is pressed the junction of the 100,000-ohm and 270,000 resistors in the delay network is grounded. This removes negative bias from the output tubes. It takes a certain time, however, for the .05-μf capacitor to discharge the bias voltage, and the tubes do not conduct fully at once. When all keys are released the bias voltage is applied again, but delay of the sound decay is caused by the 0.1-μf capacitor, across which the bias voltage must build up.

The delays are a little shorter than might be musically optimum, particularly the decay. The reason is that the sound must have disappeared by the time the key rises enough to take ground away from the tuning capacitor contacts. The keying contact spring is purposely adjusted under each key so that it is the last to be grounded when the key is pressed and the first to be removed from ground when the key is released.

The expression pedal operates an 8-ohm T-pad which is placed between the output transformer and the speaker. This is the simplest way to control volume without running a high-impedance line to a grid potentiometer. A high-impedance line could easily pick up enough hum or other undesirable noise to be objectionable, even if well shielded. The pedal itself is simply two hinged boards with the attenuator sandwiched in between. A cable strung around its knob rotates the attenuator when the pedal is depressed. Pushing down on the pedal increases the volume.

HOUSINGS

The instrument is in two principal parts, the electronic chassis and the keyboard unit. The chassis is a 10 x 17 x 3 inch aluminum unit which, after completion of all the construction and adjustment, can be mounted in a suitable wooden cabinet used for the speaker.

The keyboard unit was especially constructed of wood. While it serves the purpose well, it is a trifle large and heavy; that could probably be cured by using lighter wood.

A three-dimensional drawing of the keyboard unit appears in Fig. 18-7. This sketch, handed to a cabinetmaker, will result in a satisfactory unit. If the reader builds his own, the list of wood pieces in the figure should be helpful. Comparing the drawing with the keyboard unit photograph of Fig. 18-1 should make the scheme clear.

Referring to the photo of the chassis top, Fig. 18-3, the power supply—transformer, filter capacitor, rectifier, two chokes, and voltage-

regulator tube – are at the right end of the chassis. At the left end, from front to rear, are the three tone generator 884's, 8-, 16-, and 32-foot, and the 6SN7-GT which amplifies the outputs of the 8- and 16-foot generators. To the right of the latter is the vibrato choke, and in front of it the second 6SN7-GT. Next to the right, from front to rear, are the 6SJ7, the preset volume control, the neon lamp, and one 6G6-G. The output transformer is at rear center, with the other 6G6-G in front of it. All the tone filter components (except for the inductors) are mounted on the three double terminal boards atop the chassis. The inductors had to be kept far from the power supply, so are mounted on the left side of the chassis.

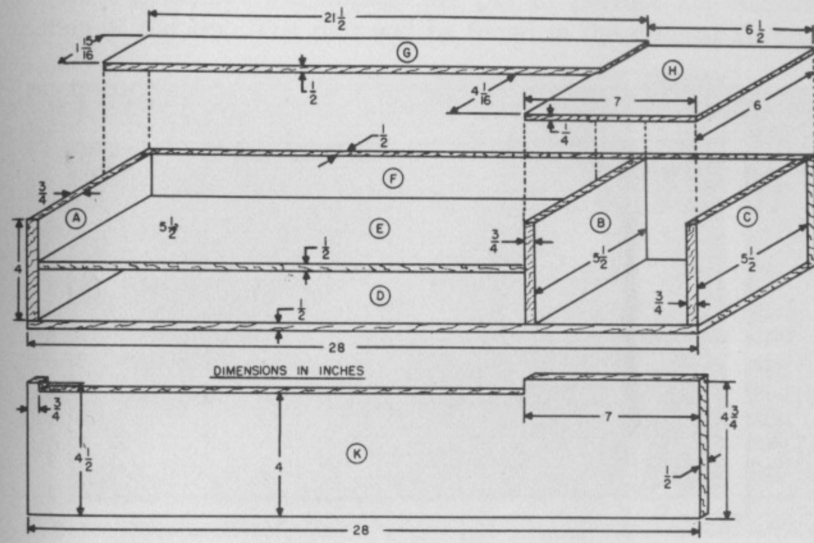


Fig. 18-7. Perspective exploded drawing of the keyboard unit.

On the front chassis apron are four connectors. The leftmost is a 20-pin female for the cable running to the keyboard unit. Next is a 2-prong (and grounded shell) connector for the expression pedal. An ordinary phone jack follows for the speaker, and at the right is a 2-prong female for the a.c. switch on the keyboard unit.

Figure 18-8 shows the underside of the chassis. The wiring under the tone-generator side is rather complex and crowded, which is one reason a change in placement of some components (as discussed later) may be advisable.

CONSTRUCTION STEPS

The first order of business is to prepare the keyboard unit so that as each portion of the chassis circuit is completed it can be tested and made final.

After obtaining the wood specified in Fig. 18-7 and cutting it all to shape and to fit, assemble the rear and the two sides and partition, pieces A, B, C, and F. The keys used in the original model were obtained from an old reed organ which was scrapped after being removed from a church to make way for a Baldwin electronic. The individual keys were removed and cut down as in Fig. 18-9, so that each was about 4¾ inches long. The raised portions of the black keys also must be cut down about another ¾ inch, so that when the black keys are in place with the white ones,

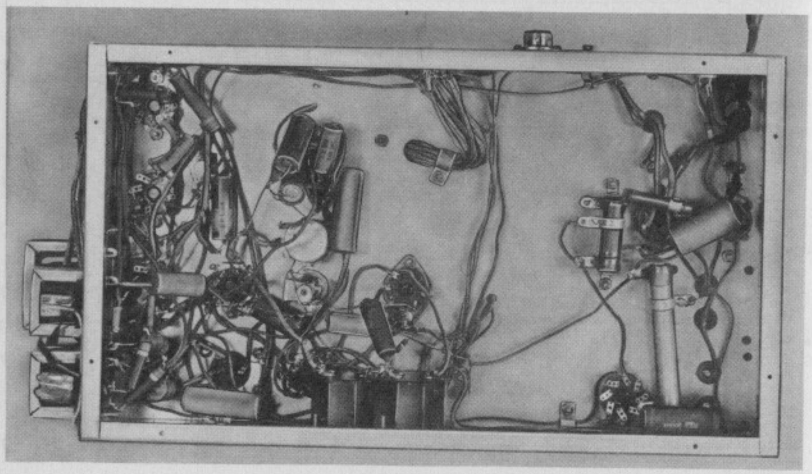


Fig. 18-8. Underside of the main chassis.

the raised portions stop 4 inches back from the tips of the white keys. This allows the key mask (piece G in Fig. 18-7) to be put in place eventually, holding all the keys at the same height.

Now refer to Fig. 18-10, showing the undersides of some of the keys. Along the bottom of each key is a piece of spring metal extending about ¾ inch out from the rear. This is a springy metal and serves two purposes. It acts as the contact which electrically connects all the contact springs under each key to ground when the key is pressed, and it is the method by which each key is mounted to the ⅝-inch square brass tube which extends the length of the keyboard (20¼ inches). The bar can be seen in Fig. 18-9. It is drilled and tapped at each key location (or self-tapping screws can be used) and the metal extension under each key is fastened to it. When the

key is pressed, the metal contacts the four springs; when it is released, the springiness of the metal brings it up again. The writer used transformer laminations for the job.

Now assemble the lowest and highest keys, with their metal strips, and fasten them to the ends of the brass bar. Hold the assembly so that with the keys perfectly horizontal the key tops are even with the top edges of the sides (pieces A and B in Fig. 18-7). Set in place the contact board (piece E) so that it will hold the bar in this position. Then remove the bar and keys.

The next job is to place the guide pins at the front of each key so that when the key is in place the guide slot underneath it (see Fig. 18-10) will engage the pin to prevent any sideways motion of the key. The pins will be found in the original keyboard

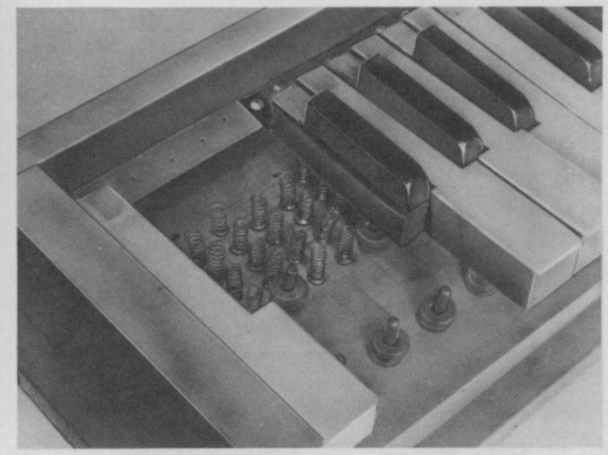


Fig. 18-9. The keys were cut down.

assembly and can usually be pulled right out with a pair of gas pliers. Each key must be held in place, the hole in the brass bar drilled and threaded, the key fastened to that, then the guide pin position marked. Drill a hole for each guide pin, then push it in place. When the keys are mounted permanently, a two small rubber grommets are slipped over each pin so the keys will hit bottom without a thud and will not go down too far.

After all the keys are mounted and working mechanically, detach them one by one and mark the contact board to show the area covered by the metal strip under each key where the contact springs will be located.

The contact springs used in the original model were cut down from copper motor brush springs obtained at motor repair shops. A hole was drilled through the contact board at each spring location and the spring held down with a round-head machine screw. Small solder lugs were placed between the nut and the board underneath. Quarters are close, so careful measurement is necessary.

The keys must be remounted next, and the contact springs adjusted by bending, pulling, and cutting them off, so that an ohmmeter shows positive contact when the key is pressed and no contact when it is released. The key mask (pieces K in Fig. 18-7) should be mounted before the final ohmmeter check is ended.

One very important point is to adjust the springs so that with each key the rearmost spring is not contacted until the key hits almost the very bottom. This spring controls the keying of the output stage; that stage should remain inoperative until all the tuning contacts have been made and should cut off again before any tuning contacts are broken.

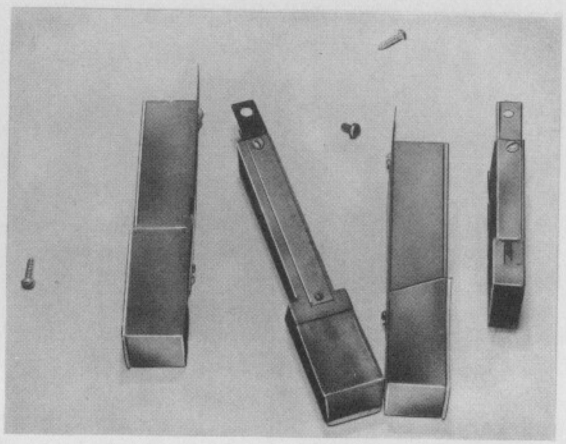


Fig. 18-10. Undersides of keys shows metal spring strips.

Before going back to the chassis, the stop buttons may be mounted on a thin board (piece H in Fig. 18-7). The cable leading to the chassis should be made up. It consists of twenty shielded wires bound together with Scotch electrical tape and terminated in a 20-prong male Amphenol connector. Numbered connections are indicated in Fig. 18-6. The keying (pin 3) and ground (pin 20) leads may be connected to the terminal lugs under the springs as indicated in that diagram. For that purpose, all rear contact springs are connected together. The cable comes out of the keyboard unit through a hole in the rear in the right-hand compartment, as shown in Fig. 18-11.

CHASSIS ASSEMBLY

The construction of the chassis assembly is next on the agenda. Begin by wiring all the filaments. Then complete the 8-foot generator and its amplifier (but not the synchronizing connection from the plate of the amplifier to the grid of the 16-foot 884), the 8-foot flute stop filter, the 6SJ7 and ½ 6SN7-GT voltage amplifiers, and the 6G6-G output stage. Short out the expression pedal receptacle SO-2, and plug a speaker into the phone jack.

In the keyboard unit, wire temporarily the first 8-foot tuning capacitor, using the .008-μf value shown in Fig. 18-6. Now, when the lowest key is pressed, some tone should be heard in the speaker. Wire up the vibrato circuit and test it. Press the key several times

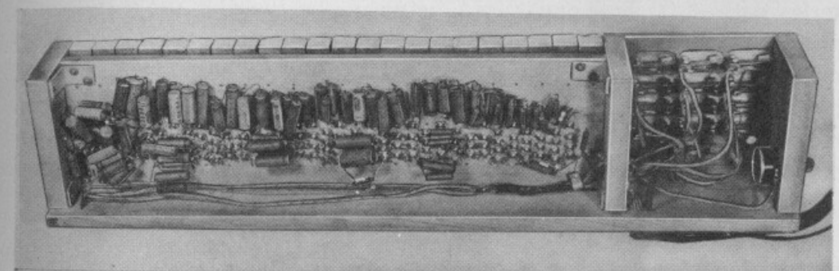


Fig. 18-11. How cable emerges from keyboard unit.

to see that there are no clicks or pops. If there are, or if there is too much delay in tone buildup, experiment with the tap on the 8,000-ohm, 25-watt section of the power supply bleeder, but do not place this so far up that the 0D3 does not glow. The 5,000-ohm, 10-watt series regulator resistor also requires adjustment for this purpose; if the supply voltage is low, it will probably have to be shorted out entirely. If attack characteristics are not perfect, experimenting with all the resistors and capacitors of the delay network at the center-tap of the interstage transformer will supply the answer. The arm of the present 100,000-ohm potentiometer may have to be moved down to prevent the audio tone from overriding the cutoff bias on the final stage with the key up.

Next wire to rest of the generators and amplifiers, but do not connect amplifier plates to 884 grids for synchronization. One each of the 16- and 32-foot filters also may be wired.

The next step is to tune the 8-foot range. *The capacitor values suggested in Fig. 18-6 are not exact and will vary with different 884's.* It is simply a matter of baving a good stock of capacitors on hand and substituting for each note until the right value is found.

Begin with the lowest note. It is essential to have a piano or a well-tuned organ for this job, and not the least of the required equipment is a good ear. As the correct capacitor is found for each pitch, wire it in place and proceed to the next note. Every few minutes, recheck the pitches of the preceding ones, readjusting the main tuning control if necessary. There will not be significant drift while the Thyratone is in operation, but the initial tuning should be very exact, which is why any drift at all should be corrected.

After tuning the 8-foot generator throughout the range, tune the 16-foot generator. Pull the 16-foot stop which has been wired and select the capacitor value which, when the lowest key is pressed, will tune the *16-foot* generator to a shade *lower* than an octave below middle C. Then connect temporarily a 1-megohm potentiometer (in series with a .0025-μf capacitor) between the plate of the 8-foot amplifier and the grid of the 16-foot generator. Pressing the lowest key, reduce the resistance until the tone just pops into synchronism and hits the C exactly. Now press the next keys up through F, each time reducing the resistance until the 16- and 8-foot tones synchronize an octave apart. In each case, recheck the lower tones to see that they are still where they should be.

It may or may not be possible, depending on the tubes and the wiring, to go as far as F with one resistance value, keeping all tones in sync. If not, stop at a lower note. Insert a fixed resistor and go on to the next group of notes in this same way. Tuning the 32-foot range duplicates this same process. When it is finished, pushing any key should produce three octavely related notes. The one or ones heard will depend on the stop buttons pulled.

The rest of the filters may now be wired up. There is an opportunity for the individual to express himself here, for by experimenting with values the tone quality of each stop can be altered to suit a whim. The ones shown in Fig. 18-5 use values found in W. E. Kock's Patent No. 2,233,948 and are fairly well imitative of the organ stop qualities with which they are labelled. The woodwind stops really should be fed square waves as is provided in the patent but we have not bothered with that in the Thyratone. The inductors in the filters were made by removing cores and windings from old audio chokes and transformers and checking values with a bridge. Exact inductance values are not essential and if there is no bridge available the inductors can be trimmed by ear – until the stops sound right.

In the keyboard unit the square brass tube acts as the common ground connection so that when a key is pressed the strip of metal under it grounds all four contact springs.

EXPRESSION PEDAL

Figure 18-12 is a side view of the expression pedal. It consists principally of two pieces of wood 4 x 10 inches and ½ inch thick. A pair of angles attached to the center outside of each and a threaded ¼-inch bar going through all four pivot it. A wire-wound, 8-ohm T-pad is mounted on an angle on the bottom piece and is turned by a string-pulley arrangement with screw-eyes. Three turns of the string around the knob is sufficient, but one end of the string arrangement should be terminated in a spring to keep tension fairly constant. A pair of angles at the ends of the bottom board provide stops to prevent the pedal from being pushed too far in either direction.

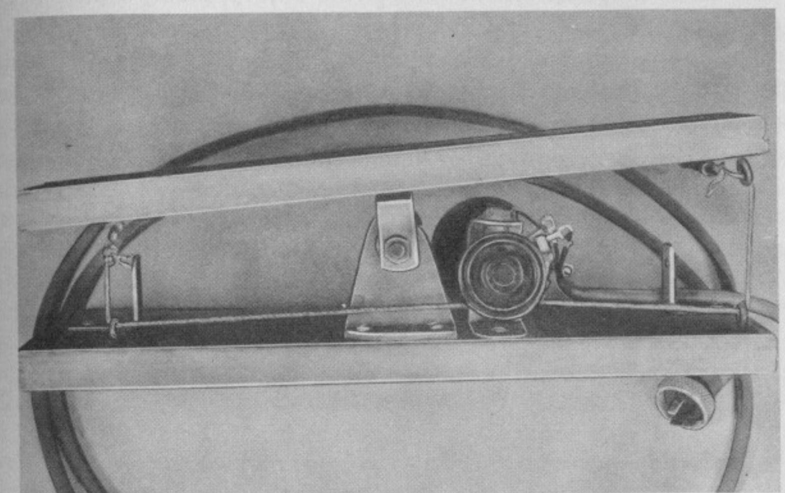


Fig. 18-12. Side view of the expression pedal.

Though this arrangement can be made to turn the pad through its full rotation, that is not ideal, since the pedal should not be allowed to cut volume down to zero. Selecting a knob with the correct diameter is the easiest way to control the amount of total rotation.

MODIFICATIONS

The Thyratone, as described here, is an experimental instrument. While it was first designed on paper, many changes were made during the course of construction and there is no doubt that ingenious constructors will have many more excellent ideas to contribute to their own versions.

One good idea, for example, might be to alter the keyboard unit so as to place in it the tone generators. This would eliminate a large amount of wire and make for less crosstalk.

Addition of the "woodwind device" explained in the Kock patent would add to the realism of the woodwind tones. This is similar to the outphaser used in the Baldwin organ and the Electronorgan.

There is a slight amount of gliding when a key is pressed; that is, the tone is not sounded squarely but slides up to pitch. That is caused by the use of a thyratron as a master oscillator (the 8-foot tube) and it could be eliminated – and over-all stability improved – by using some other kind of oscillator for the 8-foot range. For good tone shaping, however, the oscillator should have an output waveform bearing some resemblance to a sawtooth, which dictates a relaxation-type oscillator even if a vacuum tube is used. A separate tube for the triode amplifier which follows the 6SJ7 probably would help to eliminate crosstalk.

The present design does, in any case, give the electronic music enthusiast and designer a few interesting ideas and certainly provides a satisfactory solo musical instrument.

Chapter 19

The Schober Organ Kit

Now that electronic organs can be designed and built to sound just about any way the designer wishes — up to and including a really fine imitation of pipe-organ sound — the future appears to contain a genuine boom in organ sales among the general public. Electronic organs are nothing new, however, to a large group of people (most of them technically inclined) who have for years wanted to build one and many of whom have spent countless hours trying to evolve successful designs and/or duplicate commercially built units. Unfortunately, experience is the best teacher in this kind of work and there is little practical experience to draw on; this, and the necessity for some very special parts not available from electronic supply houses, has kept the number of successful amateur organ efforts down to a very small handful.

The Schober Organ Corporation, 2248 Broadway, New York 24, has come out with what appears to be the answer to the organ constructor's prayer – a full concert organ which can be built by anyone, with or without technical experience – from several kits of parts accompanied by exceptionally clear and informative instruction material. The kits are expected to attract not only the people who have long yearned to build an organ, but also those who may never have thought of it before but would like an organ and find attractive the idea of acquiring one for less than half the price of its commercial equivalent. There are 24 separate kits which, when assembled, make up the complete organ, leaving the constructor with nothing to seek out on his own except solder and ordinary radio tools. To increase the price advantage and make things easy for the constructor of modest income, each of the kits may be bought separately when and as the money becomes available.

There is nothing toy-like about the Schober Organ, nor is it an abbreviated or cut-down version of anything. It has two full 5-octave manuals, full 32-note pedal keyboard, 19 stops giving as many authentic-sounding pipe-organ tone colors, and six inter- and intra-manual

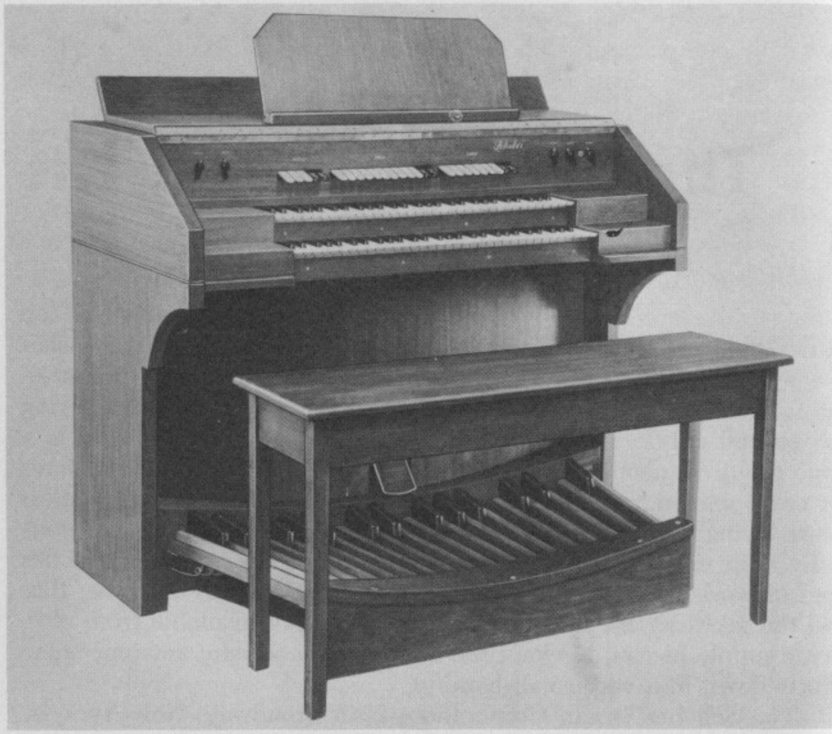


Fig. 19-1. The complete Schober Organ in the console designed for it.

couplers. Playing dimensions conform to those set up by the American Guild of Organists for pipe organs. The instrument is played exactly like a pipe organ — including registration — and sounds so much like one that the first question from many nontechnical people is, "Where are the pipes?"

DESIGN STANDARDS

Designing a fine electronic organ for home construction is a bit more difficult than designing one that is to be sold in stores as a complete unit. In addition to assuring that the instrument will be as good musically as its commercial counterpart, special approaches and techniques must be used in physical design to make certain that the average (and also the less handy than average) constructor will be able to understand the job and do it with a minimum of error and profanity. The detailed descriptions later in this article will show

how this has been done. Suffice it to say at this point that three principal factors are important:

1. *Unitized construction.* In the electrical equipment there are functionally 20 separate sections. In a commercial organ these might be combined into a few major groups, with each group lumped together on one chassis with many interconnections within the group. The Schober approach has been to keep them separate, so that each section is more easily understood (though theoretical understanding is not necessary), easily assembled, and easily serviced. This also makes possible purchase of the components in small "lots" so that little money need be spent at any one time.

2. *Extensive use of etched-circuit panels.* There are actually 168 "printed" circuits in the organ. Wherever used, they keep "wiring"

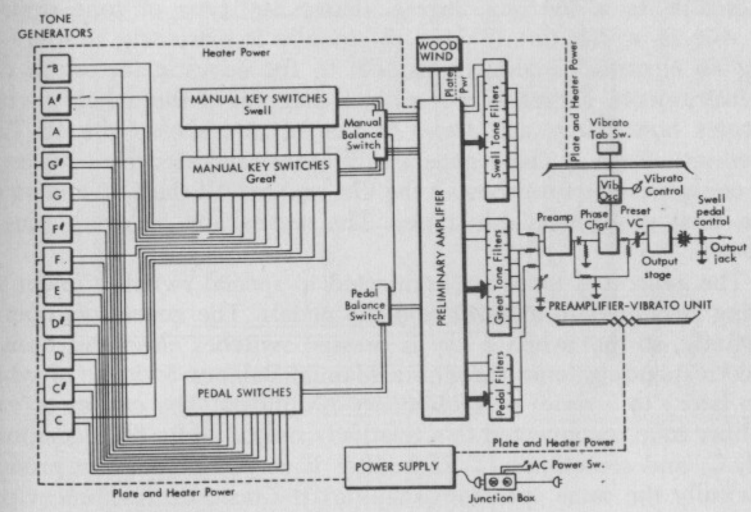


Fig. 19-2. Block diagram of the complete organ.

uniform, minimize errors, mount all components in the open easily accessible for service and test, and vastly reduce assembly time.

3. *Detailed and thorough instructions.* The instruction sheets for assembly, adjustment, and service of the various sections are easily among the features of highest importance. With every step detailed in full — even to such points as which way to wind a wire around a particular tube-socket lug — the constructor has a hard time going wrong and is sure to finish with a duplicate of the original design which will work as intended. Instruction sheets — many pages — accompany each section kit and give theoretical explanations as well for those who are interested, together with many illustrations. In con-

junction with the physical design, the instructions make it possible for entirely nontechnical people to do a perfectly satisfactory job of building an organ.

In the basic electrical design, there are several choices and the one chosen depends on which is felt to be most suitable both musically and for home construction. The Schober (like the Baldwin) is a "formant" organ which is an electrical analog of both ordinary orchestral instruments (see Chap. 4) and the so-called "straight" pipe organ, the type which is musically the most resourceful. Unlike instruments of the Hammond and Estey type, the player does not have to learn entirely new registration or playing techniques; the stops are controlled by standard tablets, each engraved with the stop name. These, as well as other console features, can be seen in the photo of Fig. 19-1, which shows the completed organ.

The Schober Organ functions solely on the formant system and can imitate to a startling degree almost any type of tone desired. The way in which this is done electrically is extremely simple – by using an electrical analog or parallel to the acoustic factors in normal instruments. Figure 19-2 is a block diagram of the Schober which indicates how things are done. At the left are shown the 12 Tone Generators. Each of these generates 7 tones separated by octaves, so that one generator furnishes all the C's, another all the D's, and so on, for a total of 84 available tones. The waveshape of every tone is sawtooth.

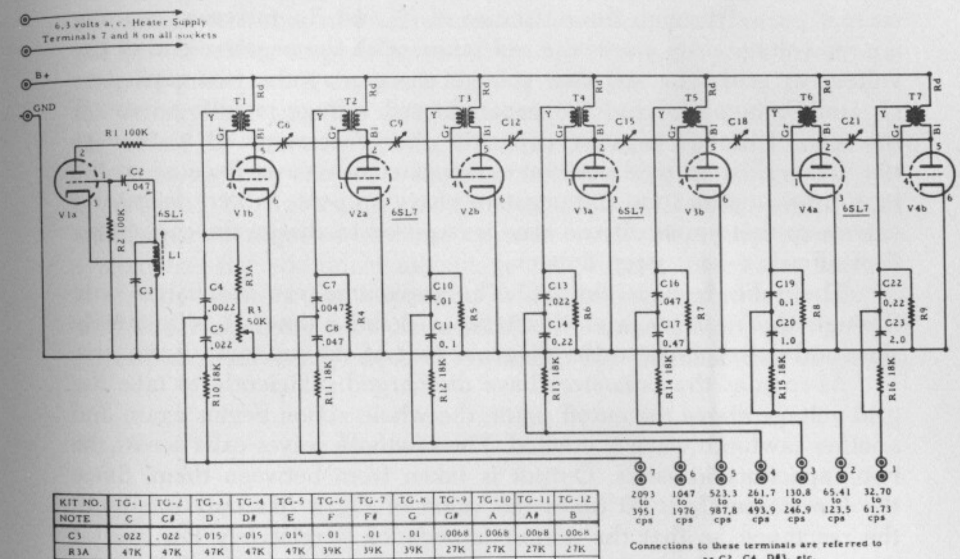
The generated tones are connected to special switches under the playing keys of both manuals and the pedals. The generators operate constantly, so that when a key is pressed switches close and transfer the corresponding tones through a Manual Balance Switch (of which more later) to a series of Preliminary Amplifiers. The output of each amplifier tube is connected to a relatively simple audio filter composed of R, C, and sometimes L. Each filter is so designed as to produce electrically the same spectrum shape as the acoustic instrument produces mechanically, and there are 19 such filters, each creating the tone color of a particular instrument or group of organ pipes. The filter outputs are passed through a vibrato circuit, preamplifier, and pedal-operated volume control to an output jack from which a cable is connected to whatever power amplifier and speaker is used.

The block diagram cannot, of course, show all the sections adequately. The coupler system is interwoven with the Preliminary Amplifier so that tone can be coupled from one manual to another or to pedals, or octave tone may be had on each manual. The Woodwind Circuit transforms the sawtooth into a symmetrical waveshape to simulate certain instruments whose harmonic content is almost entirely odd. The Manual Balance Switch enables the player to make one manual predominate over another if he wishes regardless of the stops in use; the Pedal Balance Switch adjusts the volume of pedal tones

relative to those from the manuals. A preset volume control is provided, so maximum volume can be set at installation time in accord with the amplifier, speaker, and room size. The power supply furnishes all power needed for the console.

TONE GENERATORS

Figure 19-3 is a complete schematic diagram of one of the twelve Tone Generators, with a table of parts values for all of the twelve. V_{1a} is the master oscillator, a modified grounded-plate Hartley, which



KIT NO.	TG-1	TG-2	TG-3	TG-4	TG-5	TG-6	TG-7	TG-8	TG-9	TG-10	TG-11	TG-12
NOTE	C	C#	D	D#	E	F	F#	G	G#	A	A#	B
C3	.022	.022	.015	.015	.015	.01	.01	.01	.0068	.0068	.0068	.0068
R3A	47K	47K	47K	47K	47K	39K	39K	39K	27K	27K	27K	27K
R4	100K	100K	100K	100K	100K	100K	100K	100K	68K	68K	68K	68K
R5	100K	100K	100K	100K	100K	100K	100K	100K	68K	68K	68K	68K
R6	100K	100K	100K	100K	100K	100K	100K	100K	68K	68K	68K	68K
R7	100K	100K	100K	100K	100K	100K	100K	100K	68K	68K	68K	68K
R8	100K	100K	100K	100K	100K	100K	100K	100K	68K	68K	68K	68K
R9	100K	100K	100K	100K	100K	100K	100K	100K	68K	68K	68K	68K

Fig. 19-3. Schematic of one tone generator. All twelve are physically identical, though certain of the part values differ.

operates at the frequency of the highest note and determines the tuning of all the notes on the chassis. The design is an exceptionally stable one so that once tuned, the organ will stay in tune for long periods.

The top note actually used for output and all the other, lower notes, are generated by locked frequency dividers consisting of the seven remaining triodes. This system is desirable because it means octaves will always remain in tune with each other and tuning the entire organ requires only twelve adjustments. Each of the seven stages is a blocking oscillator. Assuming for a moment that neither R_1

nor C_6 is connected to the plate of V_{1b}, we can easily explain the action.

The transformer T_1 is a special pulse transformer with low inductance; it is so connected that a positive pulse at the plate will put a negative pulse on the grid and *vice versa.* When the circuit is first turned on, a small random positive signal may appear on the grid. Because of the high tube amplification this will result in a much larger negative signal on the plate. This, fed back to grid by the transformer, increases the grid's positiveness and this action builds up so fast that the result is a very sharp and high-amplitude positive grid pulse. This makes the grid draw a large, sharp pulse of current. The current passes through the resistance of R_3 and R_{3a} in series, creating a large voltage drop across the resistance, with the negative end of the voltage at grid. The negative voltage charges up the two capacitors C_4 and C_5 in series, and the negative grid voltage resulting cuts off the tube. This fast negative capacitor charge corresponds in time to the "flyback" or vertical position of our sawtooth wave. Because of the high tube amplification the negative charging pulse is very large and narrow so that practically no time is required to charge the capacitors. This affords a very steep flyback.

The tube is now cut off. The capacitors can discharge only through the resistors, and this takes a definite time; it is relatively slow and corresponds to the diagonal part of the sawtooth wave.

As soon as the capacitors have discharged sufficiently to take the grid voltage above the cutoff point, the whole action begins again and another sawtooth wave is created. The sawtooth waves exist across the two capacitors in series. Output is taken from between them. Since the lower one C_5 is 10 times the value of C_4, it has only one-tenth the reactance, so that the output impedance is relatively low and at the same time the output lead is well isolated from the tube grid. Isolation is so good that the output lead may be almost shorted to ground wihout any material effect on operation of the generator.

So far, we have described the generator stage as free-running — constantly oscillating at a frequency determined by the values of resistance and capacitance (because they determine how soon the capacitors will discharge). What we actually want is to synchronize the frequency of this first blocking oscillator to that of the master oscillator so that both will run at the same frequency, with the master oscillator controlling. This is done simply by feeding d.c. to the master oscillator plate through the plate winding of T_1, and setting the R and C of V_{1b} for a free-running frequency somewhat below the final frequency desired. When a negative oscillator pulse passes through the winding it adds to the negative pulse beginning to build up because of the blocking-oscillator action and fires the blocking oscillator. Since the blocking oscillator fires once per master oscillator cycle, the frequency outputs of the two are identical.

The next blocking oscillator V_{2a} operates in the same way as the first with two exceptions. First, its R and C are chosen for a free-running frequency slightly less than half of the first stage. Second, synchronizing pulses are fed to its plate capacitively from V_{1b} through C_6. This stage, therefore, produces a frequency exactly half that of V_{1b}, or one octave below it. The other blocking oscillators divide in the same way.

Selected components are impractical for kit construction since in order to make the selection the generator would have to be assembled before shipping. All the components, therefore, are standard values — 10 per cent tolerance resistors and capacitors and normal production tolerances on tubes and transformers. The necessary adjustment for the first blocking stage is provided by R_3, which adjusts the free-running frequency until the stage locks in with the master oscillator. For the remaining stages the variable factor is the amplitude of the sync, for which the trimmers C_6, C_9, C_{12}, C_{15}, C_{18}, and C_{21} are used. If the free-running frequency of a stage is far below that desired, more sync is obtained by tightening the trimmer; if it is almost correct, only small sync is required and the capacitor may be nearly open. This system has the additional advantage that future corrections may be made for resistors and capacitors which change value with age, as well as for changing tube characteristics, without major surgery. Such adjustments are rarely needed normally, but the fact that they can

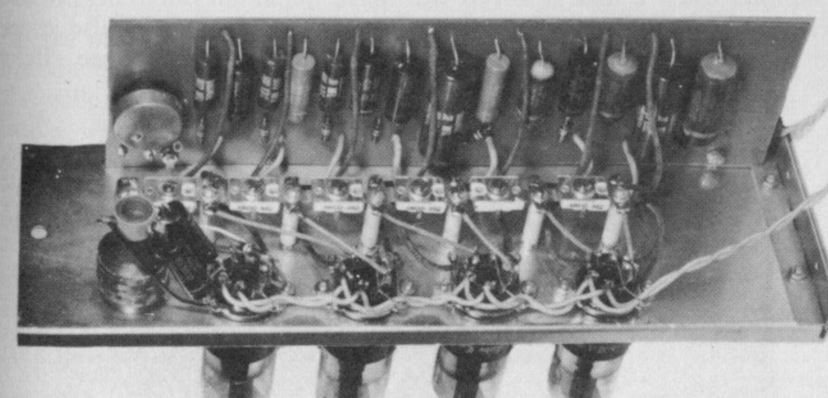


Fig. 19-4. "Inside" of completed tone generator chassis.

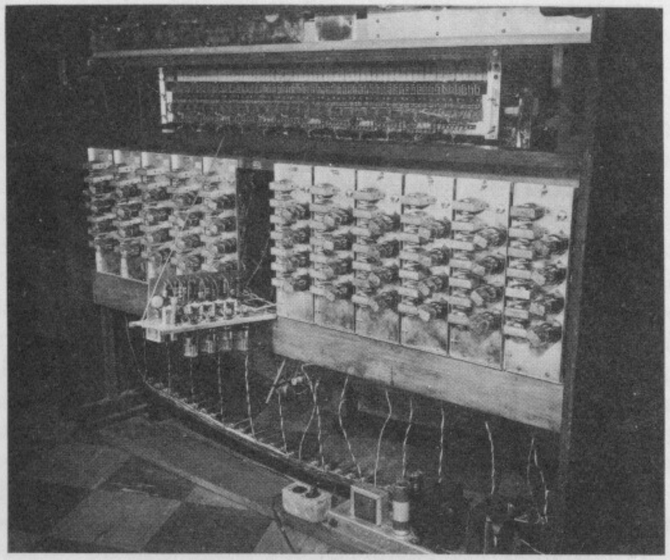


Fig. 19-5. The twelve generators mount on a rail along the rear of the console.

be made solves a possible problem. The design also cheapens the kit since perfectly standard parts can be stocked and shipped with no labor necessary for special selection.

The 18,000-ohm resistors R_{10} through R_{16} merely place a constant load across the output so that the changes between no load and that imposed when a tone is keyed will make minimum difference in the circuit and lower-frequency tones will not be affected. They also serve to keep the capacitors discharged so that they will not cause clicks or pops when the tones are keyed.

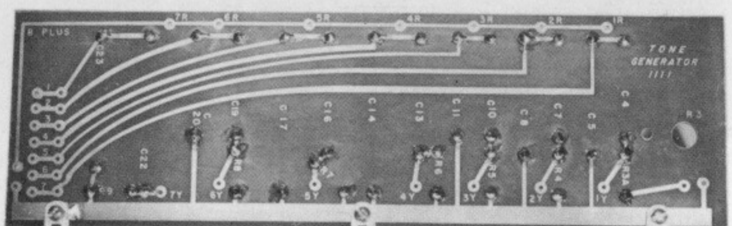


Fig. 19-6. Etched circuit panel used in the tone generators.

Figure 19-4 shows the inside of a completed generator chassis. There is a metal chassis to hold the tubes and transformers and a large etched-circuit panel mounted on it at right angles to hold almost all the other components. Figure 19-5 shows how the generators look when mounted in place in the rear of the console shell. There is a hinge on the bottom of each so that it can be lowered for inspection or adjustment without being disconnected. A screw and thumbnut at the top holds each in place in normal operation.

Figure 19-6 is the etched circuit used on each generator. All grounds are made automatically when the panel is fastened to the chassis with the metal angles. Not only the "wiring" appears on the panel; copper foil is also left in the form of markings showing exactly where each resistor, capacitor, and connection goes so that constructional errors are almost impossible. Figure 19-7 shows the rear of the panel with components mounted. Note how each component of the generator is easily accessible.

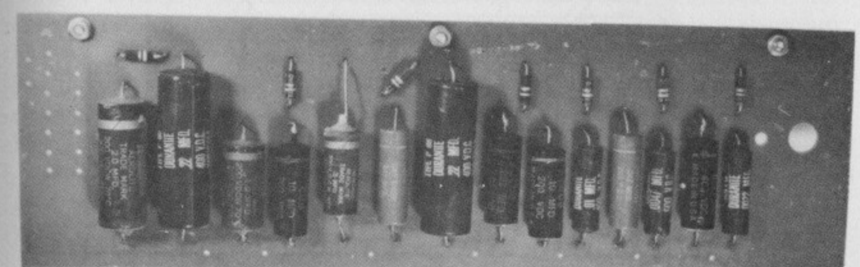


Fig. 19-7. Component side of printed circuit used in tone generators.

The organ is tuned with the variable iron core of each of the twelve master oscillator coils, and the generators need not be swung out on their hinges for the purpose. To facilitate initial adjustment the coils are furnished pretuned. Final tuning can be done by anyone, either with a simple system given in the instruction sheets for final assembly or with the aid of the company's demonstration record which most constructors will have and which contains twelve recorded tones on one side especially for this purpose.

The tone generators described furnish 84 sawtooth tones beginning with low C at 32.70 cps and going up to the high B at 3951 cps. Let us now look at the key-switch assemblies which channel the right tones to the right places when the organ is played.

When any key is pressed, three different tones are switched, corresponding to either 4-, 8-, and 16-foot or 2-, 4-, and 8-foot pitches. These pitch registers are normal in pipe-organ work. The 8-foot or unison pitch corresponds to the normal pitch of the key pressed; the 4-foot or super register gives a tone one octave above unison and

2-foot two octaves above; and 16-foot pitch is one octave lower than normal for the key.

Figure 19-8 shows the schematic diagram of the switching system for five G notes of the great or lower manual. Each switch consists of three horizontal fingers normally in the down position so that they do not strike the bus wires which are at right angles to them and run the length of the organ (left-right). When the G3 key (G just above middle C) is pressed the three fingers of the G3 switch rise, each striking one of the lengthwise busses. The center finger carries tone from the generators at a frequency of 392 cps, which corresponds to the normal pitch of G above middle C, and this tone is thus introduced into the center or 8-foot bus. Simultaneously the lower finger, which carries tone from the generators an octave below puts this tone on the 16-foot bus, and the upper finger puts tone an octave above 8-foot on the 4-foot bus.

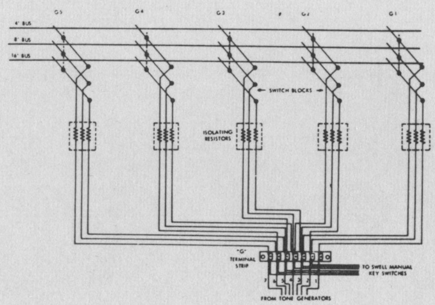


Fig. 19-8. Schematic of the G switch connections on the Great. A 4' tone for one key is the same as the 8' tone an octave above; a 16' tone is the 8' of the octave below. The bus leads go to Preliminary Amplifier.

The same generator tone is obviously used for more than one key switch. The same tone carried to the G3 center finger for 8-foot register is also used by the G2 upper finger as 4-foot tone and by the G4 lower finger as 16-foot tone. Between the generator and each switch finger there is an isolating resistor which prevents interaction and any "robbing" or lowering of the volume of one tone when another is also used, through additional loading on the generator.

The actual circuitry is exactly as shown in Fig. 19-8 except that there are twelve times as many 3-finger switches as shown, one set of five for each of the 12 notes of the chromatic scale. There is also an extra switch at the top for the highest C, since there are six C's on the standard organ manual. Each of the three output busses carries a complete rendition of the selection being played, the only differences being that if an amplifier were connected to the 4-foot bus the music would be heard an octave higher than if connected to the 8-foot bus and two octaves higher than if connected to the 16-foot bus. Pre-

Fig. 19-9. Keying a sawtooth wave only adds another vertical transient to the many already existing.

cisely the same scheme is carried through on the swell (upper) manual, except that there is a 2-foot bus and no 16-foot bus.

It is interesting to note that on the Schober Organ no system for eliminating key clicks is necessary. Ordinarily, when audio is keyed there is a click when the switch is closed; this is very destructive to music. Figure 6-8, chap. 6, shows how the click arises. Assuming that, for instance, a sine wave is being switched from its source to the grid of a tube, the switch may be closed during some part of the time when the wave is not at its zero axis. (The statistical probability of this is extremely high, since the wave passes through zero only at two brief instants.) Grid voltage then changes from its quiescent value instantaneously to some other value, and of course plate current does the same. The almost infinitely steep rise time of this sudden change is in effect a portion of a wave containing an infinity of high-frequency components. These components are heard as a click. In other organ designs it has been found necessary or desirable to place capacitive low-pass filters across the switching systems, use gradual resistive keying, or to key plate voltage on an oscillator rather than direct audio to eliminate or reduce the clicks.

The secret of the Schober's clickless keying is simply the generated waveform. The sawtooth has a very fast flyback — almost perfectly vertical, as can be seen in the drawing of Fig. 19-9. The filter system is designed to take care of this — it can take advantage of the high-frequency components in imparting brilliance to stops that require it and can roll off the highs for less brilliant stops. When the sawtooth tone is keyed at some point in its rise, the vertical rise added by the keying is just like the vertical part of the sawtooth itself,

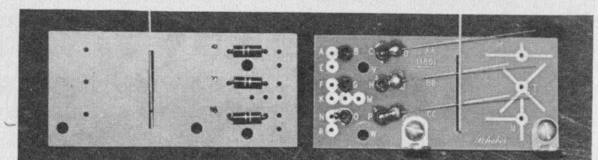


Fig. 19-10 (right) and Fig. 19-11 (left). Front and rear views of key switch.

and the filters treat it the same. To say it another way, each wave of a good sawtooth has a built-in key click. When you add another by closing a switch, neither the circuitry nor the ear can tell the difference.

From a puristic point of view there is always, of course, a transient when a switch is closed in that there is a change in the output signal from zero cps to some finite frequency. This frequency transient or discontinuity is not subject to remedy, but fortunately it is practically inaudible.

One of the manual key switches is shown in Fig. 19-10. Each switch is an etched-circuit card, die cut and pierced for accuracy, on which three fingers are soldered. These are made of a special spring silver alloy which has great resiliency and a high fatigue point to resist breakage despite many millions of operations, and which resists corrosion which would render it nonconductive.

The fingers are moved by a phenolic actuator which is held in a slot in the card and which in operation is *under the rear ends* of the playing keys. When the keys are pressed the actuators, normally held down, are allowed to rise. Each finger then contacts a gold-clad bus bar running through one of the three holes at right. Because of the points at which the actuator and bus are placed there is a small wiping action which cleans bus and finger. In case the softer gold wears somewhat after a long period, the busses can be moved to expose new surfaces.

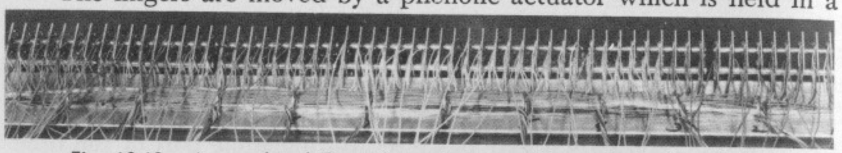


Fig. 19-12. A completed Great key-switch assembly ready for installation.

The rear view of the switch card in Fig. 19-11 shows how the isolating resistors are mounted. The remaining holes are used for wiring.

Figure 19-12 shows a completed great-manual key-switch assembly. All the cards are mounted on a steel channel, with connections between cards made before installation in the console. The 12 groups of twisted wires will go to the generators and the single wire from each switch will go to the swell assembly after installation.

The numbered metal loops are on the channel before wiring begins; they keep the wires in place as the wiring proceeds and eliminate any need for lacing.

The entire channel is mounted in the console with two end brackets. These are screwed to the woodwork and the channel is then mounted to the brackets with screws and wingnuts. Slots allow wide adjustment horizontally and vertically.

These assemblies may be used with any standard organ manuals

having keys pivoted somewhere at center so the rear ends rise when the playing surfaces are pressed. Thus used consoles may be used by those who prefer not to buy the Schober console in order to save money.

Wiring the key-switch assemblies is, of course, one of the most time-consuming aspects of the organ construction. However, the instructions have been worked out so that a simple and unmistakable procedure and a chart make the work compound – a number of small, simple, similar operations – rather than complex.

The pedal key switches are of the more rugged flat-blade type and have only two output busses, 8- and 16-foot, but the scheme is electrically the same. This is placed permanently in the console. The pedal clavier is a self-contained mechanical unit and it is simply pushed into place so that the ends of the pedals overhang the switch blades. In this way necessity for extending wiring outside the console is eliminated. Novel methods of providing mounting legs for the pedal isolation resistors and the use of large screw-eyes as a substitute for lacing make pedal-switch wiring again a succession of simple operations.

The metallic material used for key switches is a problem in organ design. Low-level audio is always critical to key since the slightest uncertainty of contact results in either noise or complete failure. For this reason silver-against-silver contacts are simply unusable, as is, in fact, any normal coin-silver contact. In the manual key switches the special silver alloy of the fingers does not corrode; neither does, of course, the gold used for the bus wires. In the pedal switches special palladium contact points are employed.

Each of the keying output busses is terminated in a maximum of 1800 ohms to ground. The resistors for termination are on the balance controls, and adjustment of these switches effectively varies the terminating resistances. In the normal position of the Manual Balance Control, all great and swell busses terminate in 1800 ohms. In the clockwise positions the swell termination is reduced so that the great produces relatively more output. In the counterclockwise position the reverse is true. This arrangement takes the place of separate swell shoes for the two manuals, reducing cost and making the organ easier to play. The pedal balance control varies the terminations on the pedal busses so that the pedal volume can be balanced with any combination on the manuals without having to depend on the present levels of the pedal stops for the purpose. The idea in both cases is to add flexibility to the playing.

PRELIMINARY AMPLIFIER

The Preliminary Amplifier is an assembly of fourteen 6SL7's which performs three important functions. First, it raises the levels of tones from the keying busses to keep the signal-to-noise ratio high. Second, it isolates the filters from each other positively so that the stops are as independent as in a pipe organ. Third, it is used in the coupler system so that coupling is without effect on other controls or sounds.

Of the 19 stop filters, 16 are fed by one triode plate each, each triode carrying 2-, 4-, 8-, or 16-foot tone from the proper source according to the purpose of the filter. Three filters, the diapasons, carry both 8- and 4-foot tone, the 4-foot tone present in small quantity to give the diapasons the necessary life and carrying power.

Figure 19-13 shows the scheme behind the system; the entire diagram is too large to print here. The great 8-foot bus output goes to the grids of all the tubes which feed 8-foot great filters through a single 47,000-ohm resistor. The amplified 8-foot great sawtooth tone at the plate of each of these tubes goes to one great 8-foot filter. There is a group of tubes operated in the same way for all the other registers.

Coupling on an organ lends great flexibility — if it is not a substitute for meager resources. The Schober has the following couplers:

Great to Great 4′	Swell to Pedal 8′
Swell to Swell 4′	Great to Swell 8′
Great to Pedal 8′	Swell to Great 8′

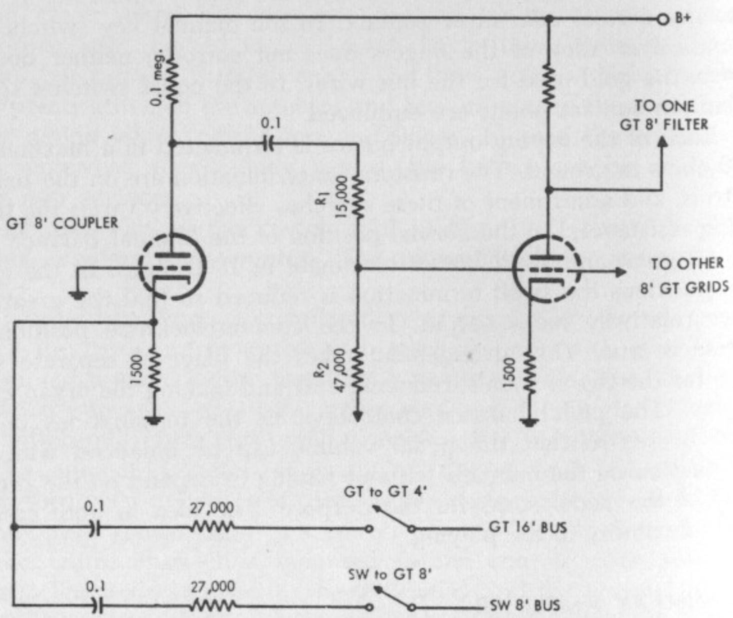


Fig. 19-13. Simplified schematic of coupler circuit which permits interconnecting manuals or buses.

The meaning is simple. Ordinarily only tones produced by playing on the great manual will go through the great filters. By using the Swell to Great 8′ coupler, the great keyed tones are made to go through any swell filters in use as well so that the tonal varieties of both great and swell stops are available on the great. The reverse is true for the Great to Swell 8′ coupler. When the Great to Great 4′ coupler is used, all tones heard due to great stops in use are also heard one octave higher as if the player were fingering everything in octaves. The same is true on the swell for the Swell to Swell 4′ coupler. By pulling the Swell to Pedal 8′ or Great to Pedal 8′ couplers, pedal tones will also pass through swell and/or great couplers, adding those tonal resources to the ones normal to the pedals. The flexibility of registration this adds is obvious.

The coupler system works electrically in a simple way. Refer again to Fig. 19-13. If the Swell to Great 8′ coupler is used, the corresponding switch is closed. Now swell 8-foot tone goes through a 27,000-ohm resistor to the cathode of the great 8-foot coupler triode, operated as a grounded-grid amplifier to keep output phase unchanged. The plate output is fed to the grids of the triodes feeding the 8-foot great stops, right along with the regular 8-foot great tone. The level of the coupled-in swell tone is kept to the same level as the 8-foot great tone by the resistor network. R_1 is the series and R_2 the shunt leg of a voltage divider for the coupler tube output. (The great 8-foot bus terminating resistor at the balance switch is also a part of the shunt leg but since it has a maximum value of 1800 ohms its effect is negligible.) For the 8-foot great tone R_2 is the series leg and the resistance of R_1 plus the coupler-tube plate-ground resistance is the shunt leg of a similar divider. Both operate so that the levels at the amplifier-tube grids are the same.

As many busses can be coupled into the cathode of the coupler tube as necessary, and each bus can be connected to as many couplers as necessary. The 27,000-ohm series resistors do not load the keying busses and, in conjunction with the low dynamic resistance of the tube cathode they isolate the busses from each other as well as keeping coupler-tube output identical to bus output. In the actual circuit each coupler switch has two sections so that, for the Swell to Great 8′ coupler, for instance, not only is 8-foot swell tone coupled to the 8-foot great filters but 4-foot swell tone is also coupled to the 4-foot great filters.

Five small etched-circuit panels are used on the preliminary amplifier chassis to carry the R_1, R_2, and associated capacitor components of Fig. 19-13. Other components are mounted directly on tube pins and tie points, so that this assembly is merely another case of making a number of simple connections rather than the complex assembly it appears to be at first glance.

WOODWIND CIRCUIT

Two of the stops on the organ — the Clarinet and Stopped Flute — imitate pipe-organ and orchestral tones which have almost no even-harmonic content. A symmetrical wave for this purpose is produced by the woodwind circuit shown in Fig. 19-14. To obtain a symmetrical wave it is necessary to invert a 4-foot signal and reduce its level by 6 db, then mix it linearly with 8-foot tone.[1] Eight-foot tone is taken from the swell 8-foot keying bus and applied to the grid of V_{1B}. It appears in phase on the cathodes of both tubes and thus effectively on the grid of V_{1A} in reversed phase, then on the plate of V_{1A} in the original phase. Swell 4-foot tone is applied through a voltage divider directly to the V_{1A} grid, and appears on the plate reversed. The mixing takes place in this manner and the result, again reduced in ampli-

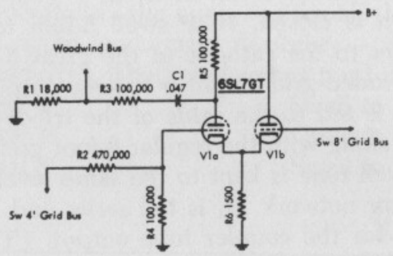


Fig. 19-14. Schematic of woodwind circuit, which converts sawtooth wave output from generators to square wave for certain tone effects.

tude, is the woodwind output, which is applied to the grids of the two preliminary-amplifier triodes feeding the Clarinet and Stopped Flute filters. The woodwind circuit is built on a small chassis with all the components but the tube on an etched-circuit panel.

As already explained, each of the nineteen stop filters in the Schober Organ is fed tone of the correct pitch register or registers by one or two preliminary-amplifier triodes. Several of these filters are presented in detail in Fig. 19-15. Discussing one or two examples will show the scheme of operation.

The pipe organ has normally four types of tones — flute, string, reed, and diapason: there is of course, considerable variety available within each class. All these tone types are reproduced in the Schober.

The Great 8′ Flute is typical of the first class. Here the incoming 8-foot sawtooth is passed through a three-section low-pass filter designed to roll off the upper harmonics so as to give a smooth, round tone which is not quite a sine wave. As with all the filters there is a blocking capacitor at the input to prevent d.c. from the tube plate

[1] Acknowledgment is due to Baldwin for this basic idea, though not for the circuitry.

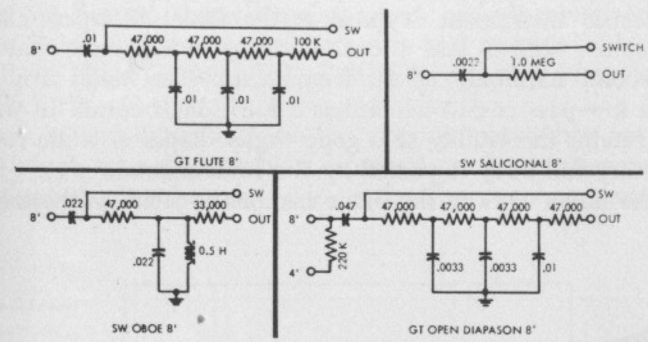


Fig. 19-15. Schematics of four typical filters used in the organ and described in the text.

from passing into the filter and following circuits. There is also a resistor at the output to set the comparative volume level of the stop. Stop switching is done by shorting the filter input (after the blocking capacitor) to ground.

A typical string filter is the Swell 8′ Salicional. String tone is characterized by comparatively little fundamental and large high-harmonic content; it is very "buzzy." In this filter there is little filtering action other than slight differentiation due to the small blocking capacitor, and the sawtooth comes through unchanged except for some flattening of the sweep with consequent transformation of the flybacks into near-spikes.

Reeds are used in pipe organs not only as imitation of orchestral reed instruments but also to simulate brasses. A typical true reed is the Swell 8′ Oboe, with its thin, nasal, penetrating tone. In this filter the input wave is first slightly differentiated by the blocking capacitor and the 47,000-ohm resistor and tuned circuit to ground. Then it is shunted by the tuned circuit which greatly emphasizes a portion of the spectrum in the neighborhood of 1600 cps. All reeds are charac-

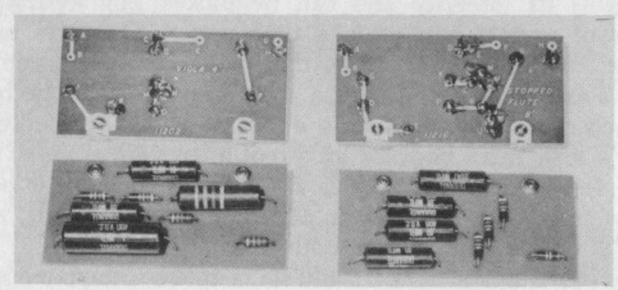


Fig. 19-16. All stop filters are constructed on uniform-sized printed circuit panels.

terized by a definite formant of this type.

Diapason tone is peculiar to pipe organs and is not imitative of any orchestral instrument. Typical is the Great 8′ Open Diapason which employs both 8- and 4-foot tone to give it a somewhat accentuated second harmonic at all frequencies. The tones are passed through a low-pass circuit which has a fairly high cutoff. In this way the tone retains the vitality of a good organ diapason while retaining the necessary full body imparted by the fundamental.

All the filters work in this same manner to yield synthesized pipe

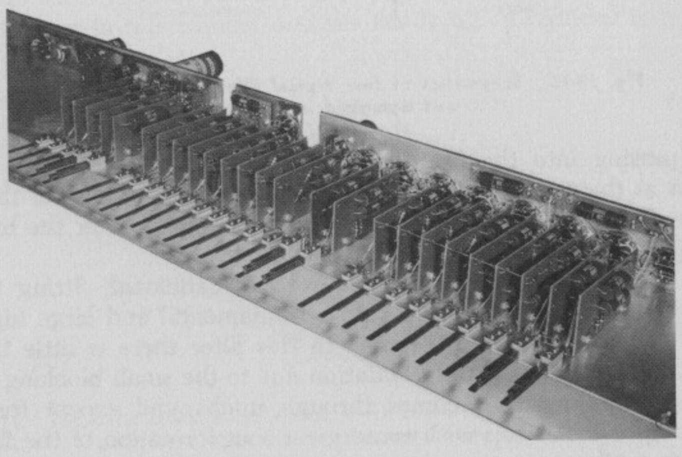


Fig. 19-17. A portion of the filterboard, showing mounting of the stop filters. Switches at the front are engaged by the stop tabs.

and orchestral tones reasonably close to the originals. All the filter outputs for each manual and the pedals are connected together permanently. Because the stop-switching ground shunt is at the input of each filter, there is no change in the loading on the output bus when additional filters are switched on. For this reason, the tone of each filter retains its integrity and can be picked out separately in an ensemble of several, just as in a pipe organ. They can be used in ensembles of many kinds, and the pitch register systems of the Schober is especially useful here; a solid 8-foot tone can be brightened by a touch of 4-foot tone from a 4-foot filter, a 4-foot ensemble can be given a little more body by an 8-foot filter, and so on *ad infinitum.* The couplers add greatly to this flexibility in making it possible to mix stops from both manuals or add them to the pedals, as well as to add the octave on either or both manuals.

Each of the nineteen filters and six couplers is mounted on a 2 x 4-inch etched-circuit panel, some of which are shown in Fig. 19-16.

While there are too many connections on these panels to allow marking component placement, each connection point is lettered and the instructions contain charts showing which components go into which lettered holes, making construction easy and unmistakable.

All the filters are mounted on the Filterboard, a long metal channel which runs the length of the organ, as shown in Fig. 19-17. On the front of the filterboard are the stop switches; they are actuated by the toggle-type organ tablets mounted on the wooden nameboard of the console. On the rear edge of the filterboard are mounted the

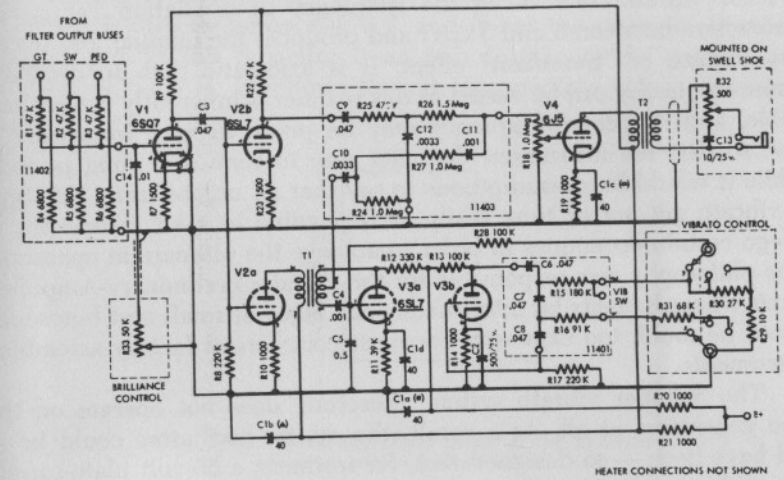


Fig. 19-18. Schematic of preamplifier-vibrato unit.

chassis of the preliminary amplifier, woodwind circuit, and preamplifier-vibrato unit. All connections are made by wires "swept under the rug" to eliminate cabling by passing them through holes and running them under the filterboard.

PREAMPLIFIER-VIBRATO UNIT

The Preamplifier-Vibrato Unit is the final electrical assembly through which all signals go before reaching the power amplifier; the unit is diagrammed in Fig. 19-18.

Each set of stop filters — great, swell, and pedal — has a common output bus. These busses are terminated by R_4, R_5, and R_6 and from each bus one of the resistors R_1, R_2, and R_3 goes to the grid of the 6SQ7 first preamplifier tube. These isolating resistors are included to avoid the necessity of commoning all busses, which would increase the loading on the output with reactive components. Although such a common would not cause the switching in of any filter to affect the tones of filters already in use (this is due to the shunt-type stop switching), the output of almost every filter is partly reactive and commoning all of them would slightly complicate the voicing problem.

R_{33} and C_{14} to ground constitute a variable low-pass filter which is adjustable from the stop panel (or *nameboard*, as it is called) to add flexibility by giving control of over-all brilliance. Tone colors are normal with the potentiometer arm in center position.

Two 6SL7's are used in the novel phase-shift vibrato which is unique in the Schober Organ. It is common practice in electronic organs to vary the voltage of some electrode of all the master oscillators for vibrato; this varies the frequency of oscillation (at a rate somewhere between 5 and 8 cps) and produces the familiar and necessary vibrato or "tremulant" effect. It is axiomatic that an oscillator whose frequency can be varied in this manner is inherently not entirely stable, and its center frequency may be prone to vary from time to time so that the instrument does not stay in tune over long periods. While it would be presumptuous to say that all organs using this type of vibrato get out of tune easily, it is possible in a kit instrument to forego certain economies in order to achieve the ultimate in operation. This philosophy was responsible for the 14-tube Preliminary-Amplifier, a unit whose benefits kit constructors can enjoy at small cost but which would be much too expensive for most commercial factory-assembled instruments.

The Schober vibrato system, therefore, does not operate on the tone generators at all. As a result, the master oscillators could be — and have been — so designed that, for instance, a 50-volt plate-supply change has negligible effect on tuning, truly an unusually high degree of stability. The vibrato operates on the tones after they emerge from the 6SQ7 stage in Fig. 19-18 by varying the relative phase of the signal over a wide range at a vibrato rate. Such phase variation is to the ear the equivalent of frequency modulation (which is what vibrato is) just as many FM broadcast transmitters generate FM by phase modulation.

Phase modulation of the necessary type is not such an easy trick and the circuit employed was developed by the writer especially for the purpose. Figure 19-19 shows a phase-shift circuit of the usual type in which the capacitive reactance is equal to or greater than the resistance. If either element is varied the relative phase of the output signal changes. The two faults of this circuit for vibrato use are:

1. Change of phase in this way also causes change of amplitude, and amplitude changes are different at every frequency.

2. The amount of phase shift differs at every frequency, the maximum attainable at the best frequency being somewhat less than 90 deg.

Figure 19-20 shows a circuit of more practical value, first published for use in phase-angle measurements.[1] The input signal is in push-pull form — two signals of equal amplitude and opposite phase. With connections as shown, variation of either impedance component will cause changes in relative output phase over a maximum possible angle of about 175 deg. While the phases of outputs of different frequencies with respect to either half of the input signal are different, varying a component shifts the phase of a signal of any frequency within a wide band over a considerable range with respect to its resting phase.

The action of the circuit can be seen superficially with the aid of the theory of extremes. If the resistor value is reduced to zero, the output is directly across *GEN 2*, and it has the phase of that half of the input. If the capacitor is at zero reactance, the output has the phase of *GEN 1*. It follows that at intermediate values of resistance

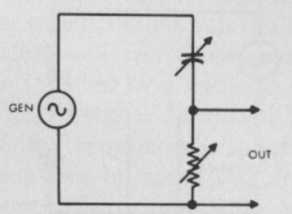


Fig. 19-19. Basic equivalent circuit of the phase-shifting operation of the preamp-vibrato unit.

and reactance the phase of the output is somewhere between these two 180-deg. extremes. The crowning — and a most important — point is signal output amplitude does not change during phase shift, nor is the input-to-output path frequency discriminating under any conditions.

To translate this to an actual vibrato circuit several things are necessary. It is obvious that either output lead could be grounded in Fig. 19-20; that shown is deliberately chosen for our purpose. The input signal must be translated from single-ended (as it emerges from the 6SQ7) to push-pull; this is easily done by feeding it through V_{2A} to T_1, which has a centertapped secondary. Next, either the resistance or capacitance must be replaced by an element with the same characteristics but which can be varied electronically. The solution to this is to replace the resistor with the plate resistance of a tube — V_{3A} in Fig. 19-18. With the grid of V_{3A} driven by signals from the phase-shift oscillator V_{3B}, which operates at a vibrato frequency of around 6 cps, the plate resistance of V_{3A} varies at a vibrato rate and the signal out-

[1] Robert C. Moses, "Phase-angle measurements at a.f.," *Radio & TV News*, Radio Electronic Engineering Section, July 1953.

put between the transformer centertap and ground is phase-modulated. The capacitance of Fig. 19-20 is the series value of C_4 and C_5.

Some further refinements are necessary. First, the vibrato rate and amplitude are made variable and controlled by S_1 on the organ nameboard. This is simply a matter of controlling the frequency of the oscillator and the signal input to the grid of V_{3A}. Both are done simultaneously by the switch, so that in the first position vibrato is slow and narrow, in the second position slow and wide, and in the third position fast and wide. Position 2 is used normally; position 1 gives an unobtrusive vibrato suitable for serious music, while position 3 is used for some popular music or where a rather ethereal and novel effect is desired.

Because it is not convenient for the player to use a three-position switch for control of vibrato in the middle of a selection, on-off control

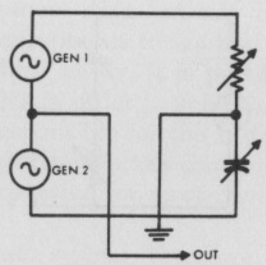


Fig. 19-20. Simplified circuit of the phase-changer which introduces an effect resulting in vibrato. GEN I and GEN 2 are actually the same tone, but with a 180-deg. phase difference.

of vibrato is assigned to a tab similar to those used for stops and marked in red. With this tab in the up position, the connection marked VIB SW in Fig. 19-18 is grounded and the vibrato oscillator stops. When the tab is flicked down, the ground is removed and the vibrato begins quickly but smoothly due to the finite time required for oscillation to build up.

One problem with this system is that if used without further modification the vibrato-frequency signal applied to the grid of V_{3A} will come through to the output as a series of thumps. This is easily cured, however, and the cure solves another requirement — that for good musicality the bass tones should have much less vibrato than others. Output from the transformer centertap passes through a high-pass filter C_{10}–R_{24}–R_{27}–C_{11} on its way to the final preamplifier stage V_4; this reduces the thumps to inaudibility and also greatly reduces the level of bass tone from the vibrato circuit. The missing bass, unvibrated, is then supplied to the final tube by V_{2B} through the low-

pass filter R_{25}-C_{12}-R_{26} in its output circuit. The two networks are complementary, so that the final tube receives all input frequencies in correct proportion but with bass obtained mostly from V_{2B} and therefore low in vibrato.

Although the vibrato circuit in the Schober Kit is furnished as shown in Fig. 19-18, it is obvious that enterprising individuals could use separate vibratos for the two manuals to achieve additional musical flexibility as well as a chorus effect.

R_{18} is a potentiometer with a short or screwdriver shaft located on the Preamplifier-Vibrato Unit chassis. It is preset at the time of installation to limit the maximum output volume to that desirable in the room, so that the expression pedal can be used fully. V_4 is an ordinary transformer-output voltage amplifier which furnishes a maximum of 2 volts in 600 ohms to the high-quality carbon potentiometer R_{32} which is mechanically coupled to the expression pedal or swell shoe. C_{13} adds additional impedance at low frequencies so that when the volume is made low the over-all tonal balance between bass and treble does not alter appreciably — a sort of loudness control.

Output taken from J_1 is fed by a cable to any power amplifier the constructor wishes to use: none is furnished in kit form. Any of the good modern "high-fidelity" amplifiers is satisfactory, though it is recommended that the amplifier be rated at a minimum of 20 watts to take care of the high-level 32-cps tone of the pedal division. Common practice in homes is to use the same amplifier employed for the home music system, with the organ line coming up on the same switch in the regular control unit which selects phonograph and tuner.

The same loudspeaker as in the home music system is usually used, but it must be a good one, particularly from the bass standpoint. Tweeters are not really necessary, though they add some brilliance, the important point being one — and preferably more than one — good bass speaker in a solid enclosure and able to withstand large cone excursions.

PERCUSSION TONES

Percussion tones are typified by those of such instruments as the harpsichord, piano, guitar, celesta, and so on. They are characterized by a rather sharp attack at full volume, followed by a fairly rapid fade to a lesser volume and then a much more gradual fade until the tone dies away. The range of tone color possibilities is as wide as for sustained organ tones, and the colors are produced in the same way — by formant filters.

The Schober Percussion Kit provides a selection of 6 switched percussion effects, each with its own tone color and appropriate decay characteristic and pitch registers. Each effect may be produced without vibrato even though the accompanying organ tones have vibrato.

In operation, when a key is played the percussive attack is obtained and the tone immediately begins to decay. If the key is held, the decay will last the normal time for the selected 'stop.' If the key is released before completion of the decay, the tone will cease. Only with this kind of operation can actual music be played, since the cessation of tone on release of the key corresponds to damper action on a piano or harpsichord.

No special style of playing is necessary here, as is true of some organ percussions, to allow the circuit to 'recover' between notes. Ordinary legato playing is feasible as well as any other normal style. And if keys are held down, any additional key struck will cause a new percussive attack, repeating all the notes held. (Patents have been applied for on essential features of the system, invented by the writer.)

The circuit operates by changing the level of a direct voltage (using the existing key switches) whenever an original or additional key is struck. This level change is differentiated to convert it into a pulse which triggers a univibrator. The uniform pulses from the univibrator charge a time-constant circuit, which is then immediately disconnected from the univibrator and allowed to discharge at a controlled rate, the two different decay curves being obtained automatically with nonlinear resistors. A control stage through which the signal passes is normally biased off; it is turned on by the fast charge of the time-constant circuit and goes off again as the voltage in this circuit decays. Signal is obtained directly from the keying busses and is passed through its own group of formant filters, so that the percussion system is quite independent of the other stops and may be used alone or in ensemble.

The Percussion Kit may be added to a completed organ without any changes in the organ. It consists of an etched-circuit panel with 5 miniature tubes and most components plus a switch bracket. Two holes must be drilled in the organ nameboard and 6 in the Filterboard.

The 6 fairly imitative 'stops' are: harpsichord (Hawaiian guitar), guitar, piano, chime (vibraphone), celesta, and steel guitar. Parenthesized names indicate effects obtained when vibrato is used.

A few years ago, most people would have laughed at the idea of constructing a full-scale concert electronic organ at home. The Schober Organ Kits are evidence that the phenomenal advance of electronics has not stopped at the fine reproduction of sound typified by the modern high-fidelty era, but has gone on to make possible the synthesis of traditional music by electronics entirely practical and artistically satisfactory.

Chapter 20

Electron-Tube Tone-Generator Circuits

In THE remaining chapters of this book it is the author's aim to supply a number of typical and possible circuits of various sorts which might prove useful in the design of future electronic musical instruments.

The job of doing this is embarked upon with a sort of hopeless feeling. One would like to be comprehensive and categoric. Yet the number of various and different possible methods available in electronics and its associated branches of science for tone production, tone coloring, keying, control, amplification, and so on, is so very great as to make any one collection small by comparison.

In these chapters, therefore, the author contributes for the reader's use more as brain ticklers than anything else whatever circuits have occurred to him and those which have been proposed in the past. The latter are far too numerous to cover in anything like useful detail, so the Appendix is provided. It lists and briefly describes a large number of patents on electronic (and electrical) music dating from 1893. This chapter begins by discussing the basic need of every instrument – basic tone generation. This and the remaining material, together with the thorough descriptions of the commercial instruments in preceding chapters, may very well be taken as a summation of present-day knowledge and achievement in the field and constitutes the building blocks which may be used and added to by every future designer.

NEON-LAMP OSCILLATORS

Probably the most truly electronic method of tone generation is the use of space-discharge tubes – hard and soft electron tubes – in which nothing moves except electrons and perhaps ions. Of these, the simplest types are gas-tube relaxation oscillators.

The basic circuit is shown in Fig. 20-1, using a neon lamp. When the switch is closed, the capacitor starts charging. The lamp cannot ionize or fire until the voltage across the capacitor is above the lamp's breakdown point. The rush of electrons from one capacitor plate to the other passes through the resistor, the voltage drop across which reduces the voltage available to charge the capacitor. As the capacitor continues to charge, however, the electron flow diminishes and the drop across the resistor becomes smaller until finally the capacitor is fully charged to the voltage of the battery.

In practice, the capacitor is never allowed to charge fully. When the charge reaches the breakdown voltage of the neon lamp (somewhat below the battery voltage), the neon ionizes suddenly and the lamp becomes a fairly low resistance. This near-short across the capacitor quickly discharges it. There being now little or no voltage across the

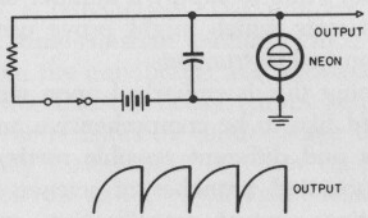


Fig. 20-1. The basic neon relaxation oscillator.

lamp and capacitor, the lamp deionizes and the capacitor once more starts to charge. The cycle is repeated indefinitely, its time per cycle (and thus its frequency) depending on the battery voltage, the resistance, the capacitor, and the striking and extinction voltages of the lamp. Output may be taken across the lamp or across any part or all of the resistor. The output wave is a sawtooth like that shown in Fig. 20-1.

The frequency of a neon-lamp relaxation oscillator is not stable. Even if all components and voltages are held constant, the discharge may not always take place between the same points along the two electrodes in the lamp. (This can be fixed by the "aging" process described in Chapter 19.) The gas is somewhat temperature sensitive, too, and the replacement problem is important, since no two "identical" lamps ever oscillate at the same frequency.

THE TRAUTONIUM

A single neon lamp has been used in a solo instrument with continuously variable frequency by (among others) the German inventor,

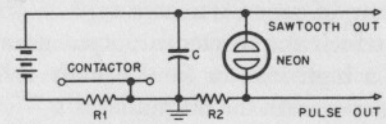


Fig. 20-2. Basic idea of the Trautwein instrument.

Friedrich Trautwein, in his Trautonium and similar instruments. The basic idea appears in Fig. 20-2. The circuit is standard except that R_2 has been added as an output load. Its value is a fraction of that of the main tuning resistor R_1. A sawtooth wave is obtained from across the lamp and a pulsed output, the result of the periodic neon-lamp discharges, from across R_2

R_1 may be a long coil of resistance wire or a long composition element like those used in potentiometers. The contactor is a strip of flexible conducting material suspended slightly above R_1 and topped with a nonconducting (insulating) layer. The player presses his finger on the contactor at any point to short out part of R_1 and change the frequency.

Typical values for R_1 and C with a 1/25-watt neon lamp experimented with by the author are 1 megohm and .005 μf. R_2 may

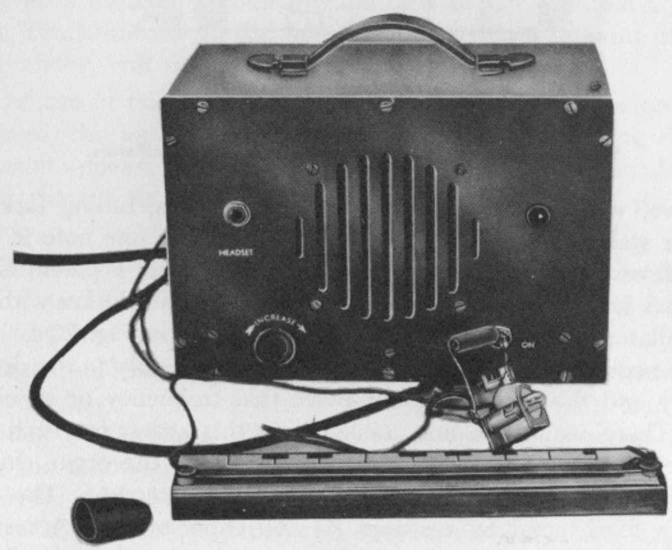


Fig. 20-3. An experimental version of the Trautwein frequency-control idea.

be anywhere from 1,000 to about 100,000 ohms. The lower values are safer because the succeeding circuits will affect the tuning less, but of course more amplification is necessary.

The circuit to which the sawtooth output goes must have a very high impedance or a high resistor (and usually a blocking capacitor) must be inserted in series with the output.

Feeding the two outputs to different amplifiers and mixing the results gives various tone qualities. The actual Trautonium circuit is more complex.

Figure 20-3 shows an experimental unit built by the author after Fig. 20-2. The long resistor is made by lead pencil mark and the piano-key drawing is a guide for fingering. The rubber fingertip insulates the player.

SYNCHRONIZED OSCILLATORS

Neon-lamp oscillators can be used in a complete organ if they are synchronized by a stable source of frequency. For a full scale, this requires twelve such sources, one for each of the notes in the

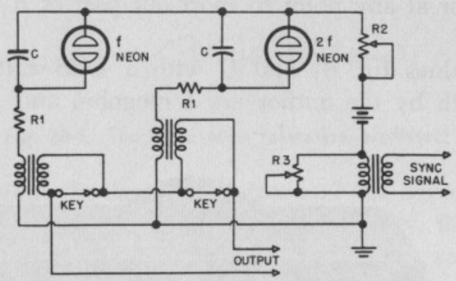


Fig. 20-4. One method of synchronizing neon oscillators.

top octave, which may be vacuum-tube oscillators, tuning forks, or any other stable devices. Each source synchronizes one note in each lower octave. A typical way of doing this appears in a patent issued to Nicholas Langer, who is probably the most prolific worker with gas-tube oscillators for music use. The idea is shown in Fig. 20-4.

Two neon oscillators are shown, one tuned roughly to the desired frequency and the other to about twice that frequency or an octave higher. There would be one oscillator in this string for each note and one for each of its octaves which appear in the organ, for example, six if the note were C, on a 61-note instrument. The neon lamps are fixed-tuned by resistors R_1 and capacitors C. Across the common resistor R_2 appears a small part of the variations in current caused by each oscillator. This is a synchronizing voltage, which

locks all the oscillators together in octave relationship. R_2, which should have a maximum value of about 10,000 ohms when 1-megohm resistors R_1 are used to tune the oscillators, is varied so that locking is obtained with a minimum of resistance in the circuit.

R_3 is across the secondary of a transformer which is fed by a primary sync source – a tuning fork or vacuum-tube oscillator of high stability with a frequency equal to or one octave above the highest note wanted. By adjusting R_3 enough primary sync signal is brought in to lock all the oscillators to it.

Output is taken through the transformers associated with each oscillator. The primaries are in series with the lamps and the secondaries are in series with each other. Each secondary is normally shorted. When a key is pressed the short is removed and the desired tone goes through to the output. The output transformers should have fairly low-impedance primaries to avoid effects on oscillator frequency and somewhat higher-impedance secondaries to minimize disturbances fed back from other circuits. Ordinary 3-to-1 audios do the job very well.

The principal problem in using neon-lamp oscillators as musical tone generators is that the frequency can be held constant only by adding a synchronizing voltage from another tone source whose frequency is stable. The drawback to that is that with most syncing systems, the sync frequency itself appears in the output of the tone generator being synchronized.

A neon oscillator producing the tone of C^{40} (middle C) for instance, synchronized by the injection of a certain amount of C^{52} an octave above, will show both frequencies in the output.

The use of neon oscillators being very tempting because of their cheapness, the author tried to eliminate the synchronizing difficulty. The result appears in Fig. 20-5. No claim of originality is made for the circuit, though its counterpart was not found in any of the patents studied.

Three neon-lamp oscillators are shown; they produce the tones C^{40}, C^{52}, and C^{64} – middle C and the two octaves above it. A synchronizing tone of C^{76} provided.

Each oscillator is tuned in the usual way by R_3, which is comparatively large -- 1 megohm or so – and C_1, R_4 is a load resistor of around 10,000 ohms. The sawtooth wave appears across it and is fed to the following stages or keying circuits. The values of R_4 and C_2 (merely a blocking capacitor,) as well as the resistance and reactance of the circuit to which the outputs are fed, have some effect on the tuning constants, but if they are not varied too widely during operation, the sync will keep the frequency constant. C_3 is a high-value electrolytic bypass, which prevents coupling between oscillators through the common B-supply. As in all oscillator strings of this type, the impedance of the supply should be extremely low to prevent coupling. Especially at the low-frequency end of the scale, decoupling networks may be used in series with the B-supply to each oscillator, though the author did not find that necessary.

THE SILENT SYNC

The important component in this circuit is R_2, across which the sync voltage is fed. The sync source may be the output of any stable

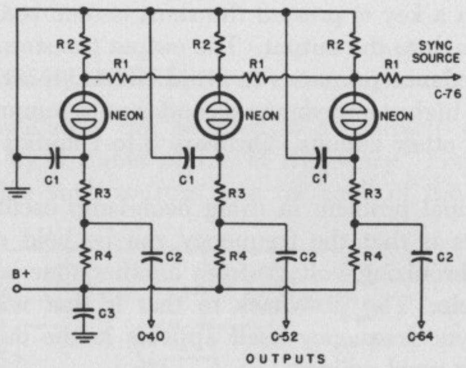


Fig. 20-5. A neon synchronizer devised by the author.

tone source whose frequency is twice that of the highest-frequency neon oscillator. As with all gas-tube syncing, best results come when the sync source produces fairly sharp pulses, but a sine wave will do the job almost as well.

The value of C_1 and R_3 are chosen so that the C^{64} oscillator will produce the correct frequency within a tone or so, without sync. (Free-running frequency should be on the low side.) Any exact values given here would not apply for various lamps and B-voltages, but any constructor can find them by experiment in about two minutes. R_2 can have almost any value, as long as it is less than approximately 10% of R_3, but lower values – 10,000 ohms for example – help in keeping the sync tone out of the output, though they require a little more sync voltage.

The neon lamp is practically an open circuit when the voltage across it is not high enough to ionize the gas. Thus, any sync voltage that appears across R_2 does not reach the R_3-R_4 section of the circuit and the output while the charge on C_1 is building up. When that

charge gets near the tube's breakdown voltage, however, a pulse from the sync source provides a little extra voltage and the tube breaks down just when the sync pulse comes. The breakdown, of course, discharges C_1 so that the next sync pulse finds the tube an open circuit and accomplishes nothing. The third sync pulse finds C_1 nearly charged again and again makes the tube break down. And so on.

If the correct values for C_1 and R_3 are used, every other sync pulse "kicks off" the neon lamp. Those that do not kick it off are not heard in the output. As a result, only a single frequency appears in the output, equal to one-half that of the sync source.

Any number of consecutive octaves can be synced. The discharge of the C^{64} neon lamp creates strong pulses across its R_2. These are coupled through an R_1 to the next lower tube, on which they act as a sync source, and so on down the line. Actually, some amount of all higher octaves appears across each of the lower-frequency R_2's; but because of R_1 the predominating frequency is that of the adjacent higher-frequency oscillator. The presence of frequencies several octaves higher would not impair operation in any case.

Only two adjustments are necessary and they need never be changed. First, each oscillator must be tuned roughly to the correct frequency with C_1 and R_3 after all connections (including output) have been made, but before R_1 is in place. Then, beginning with the path between the main sync source (vacuum-tube oscillator, tuning-fork generator, photoelectric tone wheel, etc.) each R_1 should be adjusted in turn. It limits the sync voltage. If too much is applied, the oscillator will lock in at the same frequency or a fifth below the sync. If too little, it will not remain in tune.

After adjusting the R_1 between the sync source and the top oscillator, adjust each lower-frequency R_1 individually in turn in the same way. Values vary for different neon lamps, frequencies, and B-voltages, but a good tip is to start with a variable resistor of at least 3 megohms, as very little sync voltage is needed. Of course, a full organ range requires twelve strings of oscillators, each similar to that in Fig. 20-5, and each with as many oscillators as there are octaves in the range.

After designing the circuit of Fig. 20-5 on paper, the author made a haywire assembly job of four sample oscillators on the laboratory bench. It is shown in Fig. 20-6. It worked in spite of its crude looks.

GAS-TUBE SINE GENERATOR

Ordinarily, the wave-forms obtained from gas-tube relaxation oscillators are either sawtooth (from the capacitor's charge and dis-

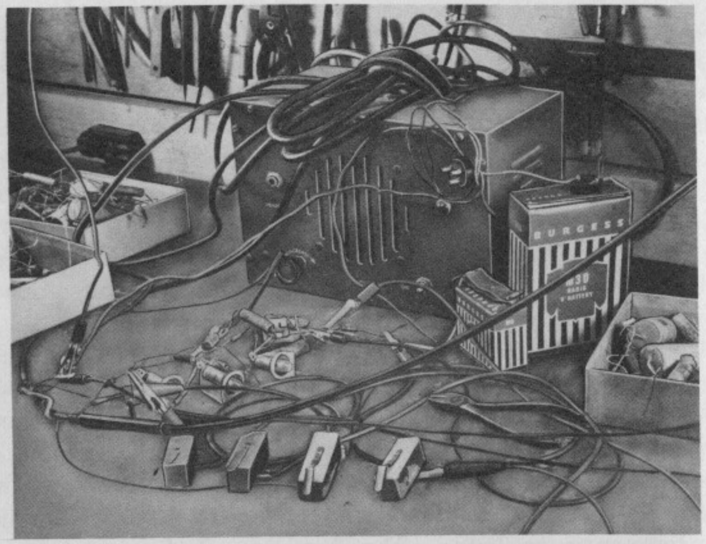


Fig. 20-6. A typical in-the-throes-of-design bench experiment.

charge) or pulsed (from the gas tube's breakdown). It is possible to obtain waves which are almost sine waves, however, by using an inductor in the circuit.

The inventor of this modification is Dr. Winston E. Kock of Bell Laboratories, who, incidentally is responsible for most of the design of the Baldwin organ (Chapter 6). The patent covering the invention is No. 2,046,463, issued in 1936 and assigned to the Baldwin Co., though not used in its organ. Not only does the inductor produce various waveshapes, but it also has an interesting effect on the frequency stability. This type of oscillator was used to produce vibrato-frequency oscillation in the Thyratone (Chapter 18).

One practical form of the oscillator appears in Fig. 20-7. R_1 and C_1 are the usual frequency-controlling elements. If the B-voltage is

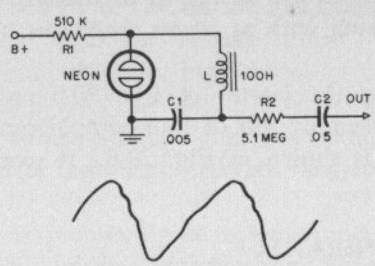


Fig. 20-7. Waveform produced by the neon oscillator using an inductance.

about 100 volts, the frequency, with the values on the diagram, is somewhere in the vicinity of 100 cycles or less, depending largely on the lamp.

The output is taken across C_1. R_2 is an isolating resistor and C_2 a d.c.-blocking capacitor. In an ordinary neon oscillator, the breakdown of the tube would short the capacitor, which would be connected directly across it. Here, however, a sudden large discharge current through C_1 is prevented by L, since an inductance tends to resist sudden changes in current. Instead, C_1 is allowed to discharge at a relatively slow rate. The current buildup through an inductor being logarithmic, just as is the voltage buildup across a capacitor, the discharge begins slowly and gradually increases in speed.

During the second part of the oscillation, when the capacitor is discharging, the inductor alters the curve, rounding it off. The resulting wave appears in Fig. 20-7, as drawn by the author from a scope pattern. It is somewhere between a sine and a triangular wave, with an undulating edge.

INDUCTOR ADDS STABILITY

Most relaxation oscillators increase frequency with increased supply voltage. So does the inductive oscillator of Fig. 20-7 – up to a point. After that, increasing voltage has less and less effect on frequency, until a point where varying the voltage has practically no effect at all on frequency. At 200 volts, for example, the circuit

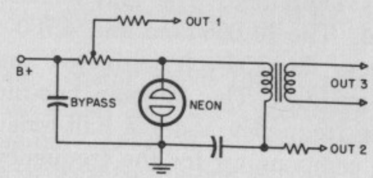


Fig. 20-8. Three places where output can be taken from the inductive relaxation oscillator.

remains stable at the frequency of middle C, and ordinary supply variations have no appreciable effect. The waveform clears up, too, the undulations disappearing from the leading edge. The frequency, when the supply voltage is high enough to make the oscillator stable, is about the resonant frequency of L and C_1.

At least three different waveshapes can be obtained from one of these oscillators, as illustrated in Fig. 20-8. The sine-wave output can be taken from across the capacitor and a near-sawtooth from a tap on the limiting resistor, both through isolating resistors. If the inductor is made the primary of a transformer, a third output is available from the secondary. All the waveshapes are different and they can be reproduced separately or combined to cause differing tone qualities. The only practical difficulty is to find a transformer whose primary has the correct inductance and a very small ohmic resistance (for reasonably good Q). The author was unable to find any; possibly it would have to be made to order. Outputs 1 and 2 are most practical.

THYRATRON OSCILLATORS

The synchronizing difficulty that appears with neon lamps can be avoided entirely by using thyratrons – gas-filled triodes such as the 884, 885, and 2051.

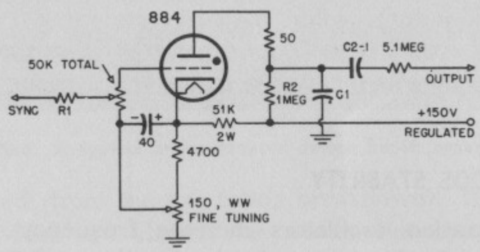


Fig. 20-9. A thyratron relaxation oscillator.

The circuits are almost exactly the same as those used in oscilloscope time-base generators. Figure 20-9 shows one similar to those used in the author's Thyratone. The grid is biased by the cathode-resistor arrangement. The 51,000-ohm and 4,700-ohm resistors are a voltage divider across the 150-volt regulated supply. The cathode is around 13 volts positive. The 150-ohm fine-tuning resistor varies the bias to vary the frequency about a half-tone or a little more in either direction. It compensates for the frequency instability caused by the 884's mercury vapor, which changes the breakdown and extinction voltages of the tube somewhat with heat.

The tuning elements are R_2 and C_1. In a monophonic instrument it is best to switch in different values for C_1 rather than vary R_2 for tuning, as the latter varies the voltage output as well as causing a kind of "chirp" – the tone does not come in on the nose but slides about a half-tone, making for very sloppy playing. The output is taken through C_2, a blocking capacitor, and the 5-megohm isolating resistor.

Synchronizing voltage may be introduced into the grid circuit as shown. The tap on the 50,000-ohm grid resistor and the value of isolating resistor R_1 must be chosen by experiment. The amount of

sync voltage is rather critical, as with most neon oscillators.

Once the 884 has heated for about 10 minutes, its frequency is stable enough to allow its use as a varied- or switched-frequency oscillator for a monophonic instrument. If more than one oscillator is used at a time, however, in an octave-coupled solo instrument or a polyphonic instrument, synchronizing arrangements are essential. The sync frequency should be an octave higher than the oscillator tone, but higher multiples are also permissible, though they make the magnitude of the sync voltage increasingly critical.

VACUUM-TUBE OSCILLATORS

The types of electronic tone generators discussed so far are often less expensive and complex than vacuum-tube oscillators, but they are rarely as flexible and reliable. Those may be two reasons why gas-tube (neon or thyratron) oscillators are not used in today's commercial instruments, though they do not seem to be compelling ones.

There are two principal types of instruments – monophonic and polyphonic. In the former, a single oscillator is varied over the entire music range to be covered and only a single tone can be reproduced at a time – no chords. A typical comercial monophonic instrument is the Hammond Solovox. Polyphonic instruments must have at least one tone source for each note of the scale so that chords may be played. Two subclasses of the polyphonic instrument are illustrated by the Baldwin electronic organ, in which both manuals and the pedal clavier operate only one set of generators; and the Connsonata in which each manual and the pedal board has a separate set of tone generators.

The monophonic scheme seems the simpler of the two, but unfortunately involves about as many problems as the polyphonic. While there may be only one oscillator, its pitch must be varied over a fairly wide range, at least two-to-one to cover a single octave. Wide-range oscillators are not simple, especially when only simple switching, if any, can be used to vary frequency and each frequency must be held to a very close tolerance (about 0.25% maximum) over a long period of time. Add to this the fact that the waveform of the output must be fairly constant over the entire range.

VARIABLE OSCILLATORS

One of the simplest variable tone generators (Fig. 20-10) is the type often used for code practice. It looks like a Hartley oscillator but does not operate the same way. Basic oscillator theory tells us that a sine-wave output with its frequency determined almost wholly by the L and C of the tank circuit (L and C_3 in Fig. 20-10) is obtained when the signal fed back from plate to grid is just enough to cause oscillation.

In this oscillator, a standard center-tapped inductor is used – usually one winding of an output or interstage transformer. That means that the signal across the grid half of the winding is approximately the same as across the plate half, which is far more than needed to excite the grid and maintain oscillation. The reason, of course, is that the tube amplifies and the output is much larger than the input.

The grid is, therefore, greatly overdriven on each alternation. On the positive half cycle it draws current, creating a voltage across R_1 *and charging* C_1. The negative charge given to C_1 suddenly

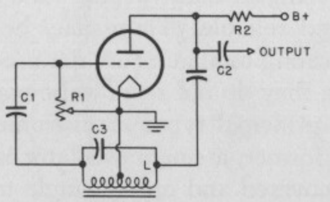


Fig. 20-10. A simple variable tone oscillator.

swings the grid to cutoff and holds it there until the charge can leak off through R_1. The time this takes does not depend much on the values of L and C_3, which would normally control the frequency of oscillation.

The frequency of the oscillator therefore depends largely on the time constant of C_1 and R_1 and the tube acts more or less as a relaxation oscillator. Varying the value of R_1 varies the time constant and controls frequency over a fairly wide range. The output waveshape may be any of a number of variations, depending on the setting of R_1, the value of C_1, and, to a lesser extent, the tank circuit. Instead of being continuously variable, R_1 may be a series of resistors selected with keyboard-operated switches.

Actual values for this circuit depend largely on what is in the tank circuit. It is very easy to make the oscillator work over any desired range, however, by a simple experiment. Select the first available inductor and use a 100,000-ohm potentionmeter for R_1. Any medium-mu triode will do–6J5, 6C5, etc. – and a supply of any voltage between 100 and 300 volts. R_2 may be about 50,000 ohms. C_2 should be large, at least 0.1 μf. Now experiment with various values for C_1, C_3, and the setting of R_1 to see how much frequency change and in what range. About 15 minutes worth of trying will

be sufficient to discover optimum values.

This oscillator will put out a sine wave but not over a wide frequency range. The simplest method to make the output a sine wave is to add a variable cathode resistor. As the cathode resistance increases, it cuts down the tube's gain, and the grid is less overdriven. When output is a sine wave, frequency stability is much better and the L-C_3 combination has much greater control of frequency. Sine-wave tone, however, is uninteresting to the ear and in a simple instrument the more complex relaxation waveshapes are considerably better.

HETERODYNE OSCILLATORS

One of the early electronic musical instruments invented by Leon Theremin in 1924, is called by the inventor's name (sometimes also called the Aetherophon). A monophonic instrument, it can produce the entire musical range. It is described in Chapter 15.

Another heterodyne instrument similar electrically to the Theremin is interesting because of the way pitch is varied. The variable

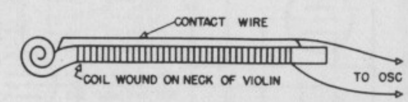


Fig. 20-11. One method of varying oscillator frequency.

oscillator is controlled by varying a tap on the tank inductor. The coil is wound on a form resembling the neck of a violin, as indicated in Fig. 20-11. Above the coil is stretched a metal wire which the player presses at various points to obtain his pitches. The wire contacts the coil at a point where it puts the right amount of inductance in the circuit for the frequency wanted. Hand capacitance is probably a problem here which the inventor, N. S. Ananiew (patent No. 1,975,220), does not seem to have solved.

W. L. Carlson is the holder of a 1931 patent (No. 1,823,724) disclosing an interesting hetorodyne instrument which can provide various tone qualities by harmonic synthesis – that is, by adding harmonics to the fundamental tones in calculated proportions as is done in the Hammond Organ. The patent was assigned to General Electric but was never, as far as the writer knows, used in a commercial instrument. Figure 20-12 shows how the instrument works.

It has the usual fixed and variable oscillators, the fixed operating, for the sake of argument, at 100 kc. It is coupled to the following tube by three separate secondaries on the tank coil. One secondary is tuned to the fundamenal frequency and the others to harmonics – 200 kc, 300 kc, and so on. (Only three secondaries are shown but more can be used for additional harmonics.) Each secondary is so spaced from the primary winding that all secondaries pick up approximately equal voltages. Naturally, the fundamental-frequency secondary has the least coupling.

All three secondaries in series are coupled to the amplifier grid, so that fundamental and harmonics appear with equal amplitudes at the output. Three series-tuned circuits are shunted across the amplifier output. A variable resistor in series with each series-tuned

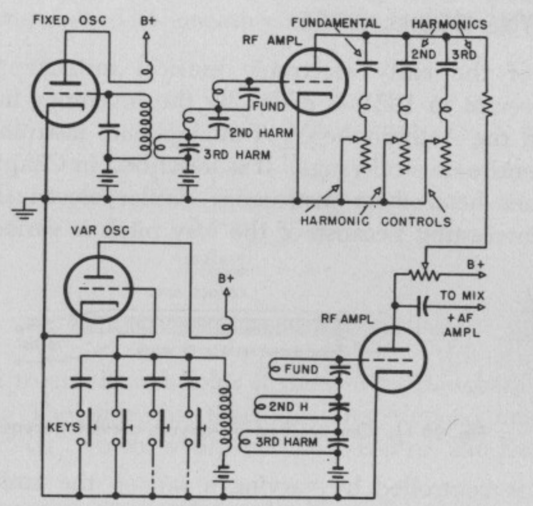


Fig. 20-12. Heterodyne instrument with harmonic synthesis.

circuit enables the circuit either to short out the frequency to which it is tuned or to pass it at the desired amplitude. Thus the stage fed by this amplifier may contain 100, 200, or 300 kc or any combination with any mixture of amplitudes.

The variable oscillator is set so that either the highest or lowest frequency is 100 kc. A series of capacitors, one for each note of the instrument, may be shunted across the oscillator coil by the playing keys. (Only one key at a time is to be pressed.) This oscillator also is coupled to a following amplifier by three harmonic-tuned secondaries. The r.f. from the two oscillators is combined in the variable-oscillator amplifier's plate circuit and the whole is fed to a mixer, amplifiers, and loudspeaker.

Let us assume that middle A, 440 cycles, is to be played. The A key is pressed, shunting the variable-oscillator coil with the correct

capacitor to produce a frequency of 100,000 plus 440 or 100,440 cycles. The variable oscillator is then also producing twice and three times that frequency or 200,880 and 301,320 cycles.

If, in the output circuit of the fixed-oscillator amplifier, there is no resistance in the second- and third-harmonic shunt-tuned circuits, only the 100-kc fundamental passes to the mixer and only 440 cycles of audio appears. But if the 200-kc shunt has resistance, it no longer shunts out all the 200 kc, which mixes with the variable's 200,880 to produce 880 cycles the second harmonic of middle A. The third harmonic is produced similarly by moving the slider of the resistor in the 300-kc shunt circuit in the direction of its maximum resistance.

The single set of harmonic shunts will take care of the entire musical range. The highest musical fundamental frequency usually

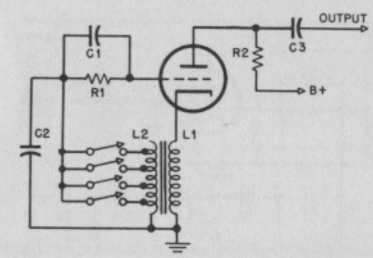


Fig. 20-13. Cathode-feedback oscillator.

used is about 4,000 cycles, which means a variation in the fixed oscillator of 100,000 to 104,000 cycles (fundamental) 200,000 to 208,000 (second harmonic), and 300,000 to 312,000 cycles (third harmonic). It is only necessary to design the secondaries of the variable-oscillator transformer so that they will tune broadly over these ranges. That is very easy, as the range in each case is only four per cent of the lowest frequency.

Heterodyne oscillators and code-practice-type relaxation oscillators can be used to generate the necessary tones for monophonic instruments.

It is more common, however, to use more or less simple *L-C* oscillators and to vary the frequency by substituting either different capacitors or different inductors in the tank circuit. The reason is that an oscillator is most stable when there is just enough feedback to sustain oscillation. Audio-frequency oscillators are not necessarily more stable (in percentage terms) than r.f. oscillators, but when the latter are used in a heterodyne arrangement, the small percentage of inadvertent frequency variation at r.f. produces a very large and intolerable percentage of variation at audio. For that reason it is seldom practical to control a beat-frequency electronic musical instrument with ordinary keys which introduce fixed frequency changes; the changes simply will not remain fixed.

At audio frequencies, almost any kind of *L-C* oscillator can be varied in frequency in fixed steps by having the keys operate switches which substitute various values of *L* or *C* in the tank circuit. Representative circuits appear in Figs. 20-13 and 20-14, which show methods for inductive and capacitive tuning.

OSCILLATOR CIRCUITS

The oscillator of Fig. 20-13 uses feedback from the cathode to the grid through a transformer to create oscillation. L_1 is the trans-

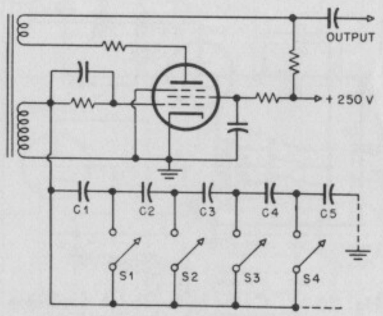


Fig. 20-14. Tuning an oscillator with a series capacitor string.

former primary and L_2 the secondary. R_1-C_1 is a grid leak bias arrangement, C_2 the tank capacitor, C_3 the output blocking capacitor, and R_2 the plate load resistor.

The frequency is varied in fixed steps by using taps on L_2. Each key controls an s.p.s.t. switch to determine how much of L_2 should appear across C_2. If two or more keys are inadvertently pressed simultaneously, the one putting the least amount of inductance in the circuit is effective and the highest note sounds.

The circuit of Fig. 20-14 is an ordinary feedback oscillator with a tickler winding on the oscillator transformer. The frequency is varied with a string of capacitors that may be tapped at any point. The method of calculating or finding experimentally the correct capacitor values for the various notes requires a little explanation. For this purpose only five notes are provided for in the diagram.

With all switches open, all the capacitors are in series and the effective capacitance across the tank coil

$$C = \frac{1}{\dfrac{1}{C_1} + \dfrac{1}{C_2} + \dfrac{1}{C_3} + \dfrac{1}{C_4} + \dfrac{1}{C_5}}.$$

This is the minimum possible capacitance and corresponds to the highest note.

With S_1 closed by pushing the appropriate playing key, C_1 is shorted and the net capacitance is the reciprocal of the sum of the reciprocals of only C_2, C_3, C_4, and C_5, which increases the *C* in the tank and gives the next lower note. With S_2 closed, only C_3, C_4, and C_5 remain, and the result is the second lower note.

It might seem simpler to use a switching system like that in Fig. 20-15, where each capacitor is the only one across the tank when

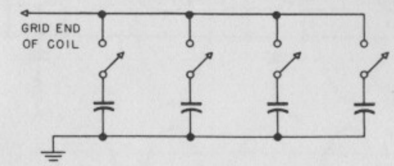


Fig. 20-15. This method of parallel capacitor switching makes playing errors likely.

its key switch is closed and series values need not be calculated. The danger, however, is that the player may accidentally hit two or more keys at a time, throwing two of the capacitors into the circuit. The result would be that the *C* in the tank would be the sum of the two and would tune the oscillator to a note entirely different from either one alone – and probably one that had nothing to do with the scale. In Fig. 20-14 that is not possible, since hitting two keys at a time would simply short fewer or more of the capacitors; the note sounded would be the lower of the two. In other words, it is not possible to put strange values of capacitance across the tank.

In designing or constructing the circuit of Fig. 20-14, start with the lowest note. Place a single capacitor across the tank and select the value that gives the lowest musical note desired from the instrument. For the next higher note, select a capacitor which, in series with the first, will give the desired note, and so on up the scale. Any but very rough calculations are entirely unnecessary, since a good deal of experiment is required anyway, due to relatively large tolerances in marking capacitors.

The waveform of the oscillator output depends largely on the amount of voltage back from the tickler winding, which, in turn, depends on the relative number of turns in tickler and tuning coil and on the value of the resistor in series with the tickler. Experiment also with grid-leak values.

MULTIRANGE INSTRUMENTS

In general, it is difficult to construct an oscillator which will be stable and have the same waveform over a very large frequency range. For that reason, some designers of monophonic instruments have used oscillators with a relatively narrow range – one to three octaves, using frequency multipliers or dividers to provide the other octaves as they are needed.

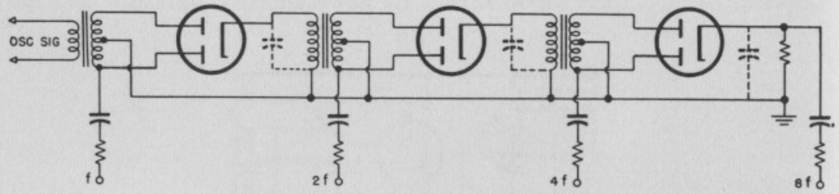


Fig. 20-16. Full-wave-rectifier frequency multiplier.

Figure 20-16 shows a simple arrangement for a string of frequency multipliers which needs no d.c. power for its operation. It is designed as a series of simple full-wave rectifiers identical to those used in receiver and amplifier power supplies.

Every radioman knows that the output of a 60-cycle full-wave supply contains 120-cycle ripple because the rectifier takes every negative alternation of the original a.c. (dotted lines in Fig. 20-17) and places it above the base line to make it positive. There are then twice as many positive alternations as before. Now when the d.c. pulsating at the doubled frequency is taken through a capacitor or transformer which will pass only a.c., the base line moves to the effective center of the waves and we get the a.c. shown at (B) in Fig. 20-17, which is twice the frequency of the original sine wave signal.

In Fig. 20-16 we have three full-wave rectifiers in cascade. If the primary of the first transformer is fed a frequency corresponding to C^{16}, 65.41 cycles, each rectification will provide the next octave C above. If the master oscillator is variable over one octave beginning at C^{16}, then the topmost note available from the third rectifier will be C^{64}, two octaves above middle C. In the diagram of Fig. 20-16 the outputs are labeled *f*, *2f*, *4f*, and *8f*, to indicate the appropriate multiple of the master oscillator frequency *f*. Each frequency

is taken from the secondary of a transformer through a capacitor and a "leveling" resistor. Since there is some power loss through each rectifier and transformer, the values of the leveling resistors may be adjusted to have the output voltage at all points the same. The load on the last rectifier is a resistor.

As (B) in Fig. 20-17 indicates, the output of each rectifier is not the original sine wave. The presence of the sharpened negative alternations will introduce even harmonics, which may or may not be desirable. If not, a fairly large capacitor may be placed across the primaries of the second and third transformers and across the ter-

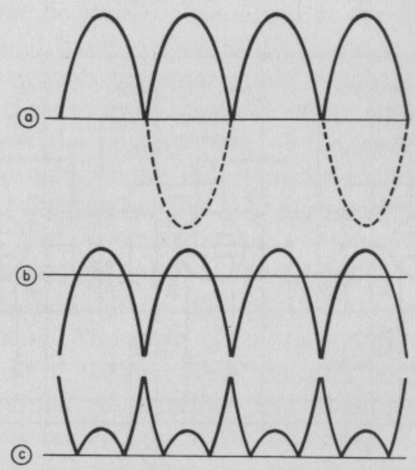


Fig. 20-17. First few output waveforms of the rectifier multiplier.

minating resistor, as indicated by dashed lines in Fig. 20-16. If these capacitors are large enough, they will round the negative peaks fairly effectively. They will also, of course, reduce the available output voltage, but that is not serious if the level is high enough in the first place. Alternatively, a pi-section low-pass filter may be inserted in each output following the leveling and isolation resistor.

If the waveform remains unaltered after the first rectifier, the shape of the output of the second rectifier will be different again, for the sharpened negative peaks will have been transferred above the baseline. The second rectifier output will look something like (C) of Fig. 20-17, and subsequent rectifier outputs will be even more complex. Again, the designer must determine whether this is desirable for the particular instrument in which the circuit will be used.

Obviously, the number of rectifiers actually used may be greater or smaller than in Fig. 20-16. IN34 crystal rectifiers may be used instead of vacuum tubes (but watch the currents through these) so that no power supply of any kind is necessary. The transformers may be interstage units with 1-to-1 ratios or perhaps slight stepups. If level is sufficient, the primary of the first transformer may be used as the inductance of the oscillator tank circuit, depending on the type of oscillator circuit that is used.

NONRESONANT DIVIDERS

Very few electronic musical instruments of the monophonic or melody type are manufactured commercially. The best known is the Hammond Solovox, which uses a system of master oscillator and fre-

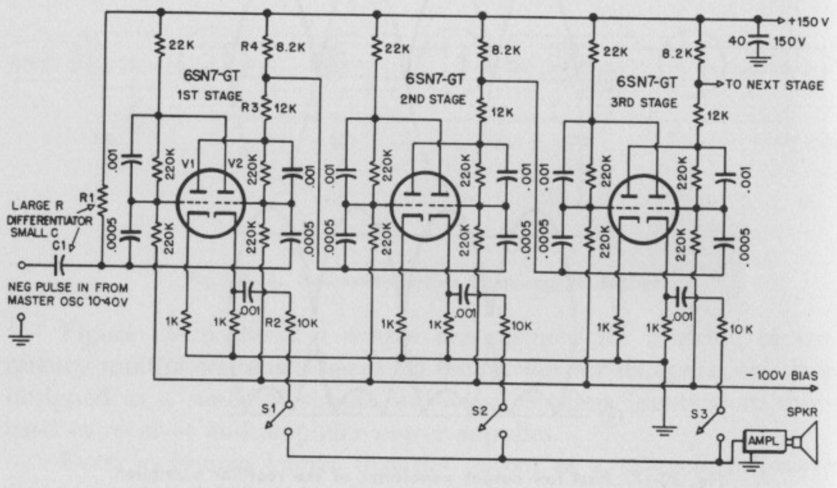


Fig. 20-18. An aperiodic, three-stage, flip-flop frequency divider.

quency dividers to obtain its musical range. Another frequency divider of interest was invented by Nicholas Langer (of neon-tube organ fame) and assigned to Central Commercial Co., maker of the Lowrey Organo. The patent describing the dividers is No. 2,486,039, granted in October, 1949.

The complete circuit diagram of a three-stage divider appears in Fig. 20-18. This circuit takes oscillations generated by a master oscillator at a frequency one octave above the highest note desired in the instrument and derives from that oscillator notes of the top and two successive lower octaves. Additional stages may be added to give as many octave divisions as desired. In a monophonic instrument, the master oscillator would have a variable frequency over at least one octave. Depending on the switching system used, the cor-

responding notes of the lower octaves would then be heard, either one octave at a time or several octaves in unison.

Unlike many divider circuits, this one is aperiodic or nonresonant, and is very nearly independent of actual frequency. It will work over almost the complete audio range without any changes. Circuit constants of the three stages are alike, so that a description of the first suffices also for the others.

The input signal is furnished by a master oscillator which should generate a nonsinusoidal waveform. A sawtooth oscillator would be suitable, as would any other furnishing a wave high in upper harmonics. A neon oscillator would do, but like any other source used, its frequency must be stable. The signal at the input to the circuit of Fig. 20-18 should have predominantly negative pulse form, which in the case of a sawtooth generator would merely require that polarity be appropriate. If necessary, a crystal rectifier could be placed in the circuit to short out the positive parts of the wave.

C_1 and R_1 form a simple differentiator circuit. Essentially it is a highpass filter which, when fed a complex wave tends to remove the fundamental and lower harmonic components and leave only the higher harmonics; the result is a rather sharp waveform.

The triode sections of the first 6SN7-GT form a pair of cross-connected amplifiers. The plate of the second feeds the grid of the first, and the plate of the first feeds the grid of the second. When the unit is first turned on, therefore, and there is no signal, the first slight electron flow in one tube starts the action. Let us assume that this takes place in V_1 and that the grid becomes slightly positive with respect to the cathode. There is, then, a small plate-current flow through V_1, which makes its plate become more negative. This negativeness is transferred to the grid of V_2, decreasing V_2 plate current and making the plate more positive. The positiveness is transferred to the grid of V_1, adding to the positiveness there at the start. The whole action snowballs in a very short time until V_2 is completely cut off and V_1 is conducting strongly. At that point, the grid of V_1 is receiving the full 150 volts positive which the 150-volt supply is placing on the plate of V_2 in the absence of plate current. At the same time it is polarized 100 volts negative by the bias supply. Its net voltage is therefore plus 50 volts, and there the circuit is in equilibrium – V_1 conducting and V_2 cut off.

Now let us apply a negative pulse from the master oscillator to the input terminals. Passing through the C_1-R_1 network, the pulse is sharpened and transferred through the two 500-$\mu\mu$f capacitors simultaneously to the grids of both triodes. Since V_1 is already cut off it has no effect on V_2 plate current. The negative pulse, when applied to the V_1 grid, however, reduces the positive potential there, and as a result V_1 plate current instantly decreases. The plate of V_1 thus produces a positive pulse; transferred to the grid of V_2 the positive pulse starts plate-current flow, making the V_2 plate negative. The negative voltage at the V_2 plate is applied to the V_1 grid, assisting the original negative pulse from the master oscillator. Again the action snowballs, until V_1 is cut off and V_2 is fully conductive.

The next negative pulse from the master oscillator reverses the situation again. It is apparent, then, that each time a negative pulse is applied, the two triodes change states.

The cathode resistors of 1,000 ohms do not change this explanation appreciably. They are there merely to provide an output point of reasonably low impedance which can be connected to outside circuits without harming operation of the divider.

A full cycle of output from a single triode includes one "on" and one "off" of plate current. Since it takes two negative pulses from the master oscillator to make a triode go through this one cycle, obviously the output of the divider is at half the frequency of the master oscillator. Output is taken in Fig. 20-18 from the cathode of V_2 through a d.c. blocking capacitor and a 10,000-ohm isolating and leveling resistor R_2.

An additional output is taken from the plate of V_1 by effectively tapping down on the plate resistor R_3-R_4. This output is fed to the grids of the next similar frequency divider, and so on.

Each of the cathode outputs is of substantially square waveshape and contains large amounts of odd harmonics. It may be fed to following amplifiers and tone controls, in which rectifiers may be provided to give some even harmonics.

SWITCHING SYSTEMS

There are three principal ways in which switching may be arranged with the Langer frequency-divider circuits, though not all of them are suggested in the patent itself.

If the master oscillator itself can be varied over a range of several octaves without losing stability, the scheme of Fig. 20-18 may be used. Here S_1, S_2, and S_3 select the ranges which are to be sounded. If, for instance, the oscillator covers the top three octaves of the piano keyboard, notes 25 through 88, then, if S_1 is closed, notes 40 through 76 will be heard. S_2 will bring in notes 28 through 64, and S_3 will cover notes 16 through 52. The range in which the instrument will work at any given time may thus be selected with the three register switches. Since two or all three of them may also be

closed at a time, the player may bring in unisons with a maximum of three octavely related notes at once.

Figure 20-19 shows a switching circuit to be used with master oscillators that cover only one octave. Each playing key actuates two s.p.s.t. switches. The lower switches substitute different tuning components in the master oscillator (shown in the figure as a neon lamp). The upper ones connect the output of the correct divider to the amplifier. This is not a very flexible system, as each key will

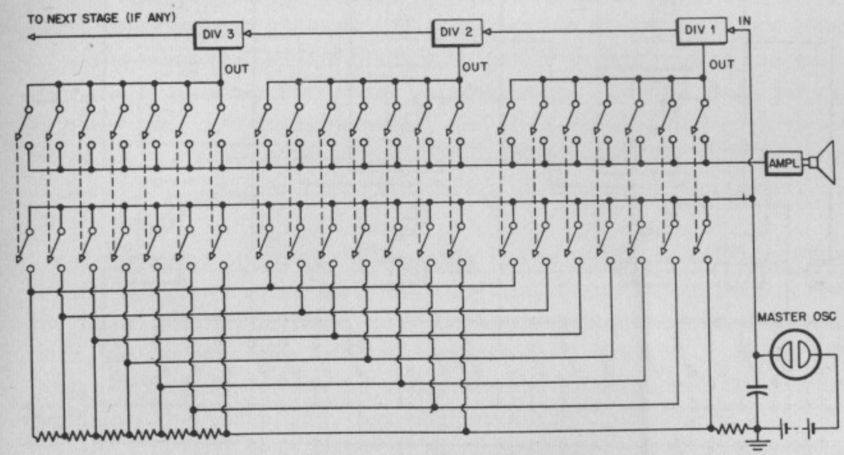


Fig. 20-19. A switching circuit for master oscillators tunable through only one octave.

only sound one note at all times. If there are to be more than the three octaves shown in Fig. 20-19, as many keys as there will be notes in the instrument must be furnished.

A third system, shown in Fig. 20-20, utilizes nine relays to make a three-octave keyboard effective over a five-octave range when the oscillator is kept within a one-octave variation range. (This latter condition is often necessary for stability with many types of oscillators. Large tuning range and stable operations do not usually go together.)

As in Fig. 20-19, the lower bank of keying switches tunes the master oscillator over its one-octave range. The top bank, however, energizes relay coils. Three relays each are provided for high, low, and middle ranges. With the high-range selector switches (the lower three-ganged units) closed, pressing any key draws sound from its own frequency divider. With the middle-range switches closed, each key draws sound from a divider giving a tone an octave lower. And the low-range switch sounds the lowest-octave dividers. One, two, or all three switches may be closed at once for unison effects. A system like this will give much the same effect as the Solovox, and as a matter of fact an earlier model of the Solovox did use a relay system somewhat like this one.

The relays must, of course, act quickly and quietly. The diagram provides for 6.3-volt a.c. coils, which might well be amateur

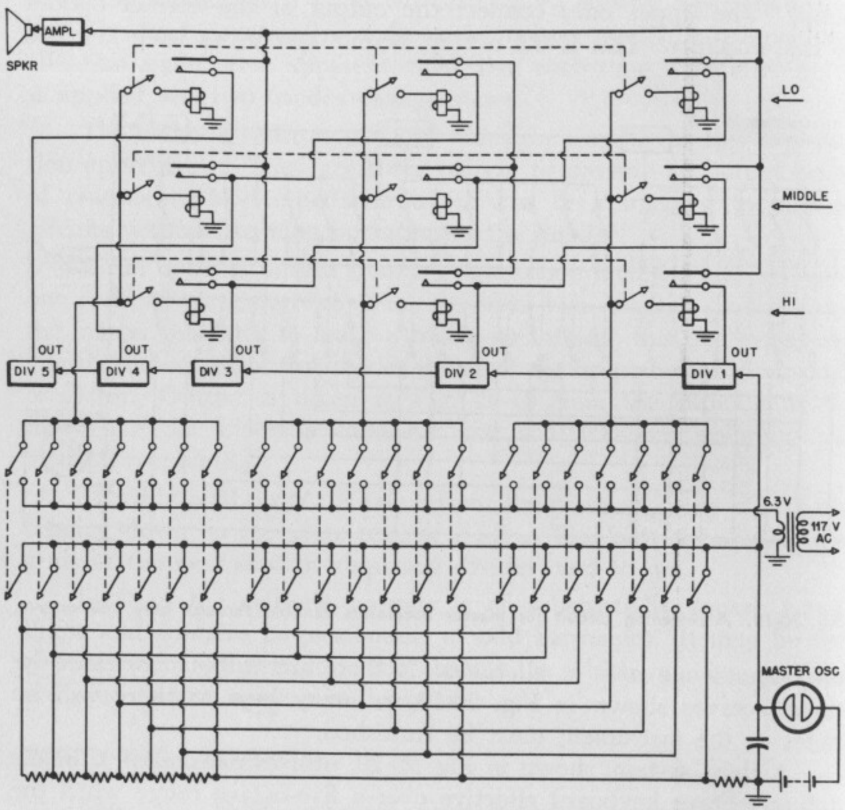


Fig. 20-20. A relay-type switching system.

keying relays. These may be too noisy, however, and it may be better to substitute some small sensitive relays. These normally operate in the plate circuits of vacuum tubes and require very little actuating current. The gaps being small and the armatures light, they make almost no noise.

POLYPHONIC INSTRUMENTS

Constructing a polyphonic electronic musical instrument is a big job compared to building many other electronic devices. Sixty separate tone frequencies must be available for even a single-manual, five-octave instrument. So the number of components will not be small. And after the bare tone frequencies have been provided, the tones must be shaped — varied in quality — and volume-controlled. Keying delay should also be provided; vibrato or tremolo is needed; couplers may be desired between manuals; octave couplers may be wanted; manuals (and pedals) and a console must be secured or made; and so on.

Building an electronic organ is not a job for the novice, nor is the design for such an instrument a decision to be taken in five minutes. But strangely enough, initial generation of the necessary tones, which is what this section deals with, is the easiest part of the job.

There are only two basic requirements. The first is to provide as many tone sources as necessary and the second is that each should be of the correct pitch. These requirements indicate that (initially, at least) the designer may consider all of the oscillator circuits discussed so far.

Depending on what the designer has in mind, several other conditions may have to be met, and they may be no less important than the basic ones. The tones must be keyed somehow, and it may be desirable to key the oscillators themselves without running into "chirp" or clicks. A keying delay should be included, even if the oscillators themselves may have to provide it. The desired waveform may be anything from a sine to the most complex and the oscillators rather than following tone-shaping circuits may have to provide it. Frequency stability may be important unless there is some provision for synchronization. And other points may crop up, not the least of which may be the necessity of keeping costs and space requirements down.

For frequency stability it is desirable to have all oscillators tuned to the octaves of each of the twelve notes synchronized so that they will "lock in" with each other. But synchronization has two disadvantages. First and most important to the tonal effect, it eliminates chorus effect since all the oscillators operate in phase. Second, if sine waves are desired, very few oscillators will be found which can be synchronized and which will produce sine waves. Oscillators naturally lock in, of course, when both are at the same frequency, but synchronization is a touchy matter when sine-wave oscillators are required to lock with a harmonic or subharmonic.

Figure 20-21 illustrates six representative feedback oscillators, all of which are suitable for electronic organs (though not all have been used in commercial instruments). Figure 20-21 (A) is the tuned-grid (plate tickler) oscillator, in which the grid circuit is tuned and a secondary winding on the inductor feeds back energy from the plate. As in all feedback-transformer arrangements, the connection polarity of primary and secondary must be correct for positive feedback; if the circuit does not oscillate, reverse the connections to primary or secondary (not both).

Figure 20-21 (B) diagrams the tuned-plate (grid-tickler) oscillator, which is exactly the same as that of (A) except that the plate circuit is tuned. In this case the values of the tank inductor and capacitor play the greatest part in determining frequency, though the tube characteristics, plate current, tank current, and other factors do influence it. The paralleled resistor and capacitor in each circuit is the

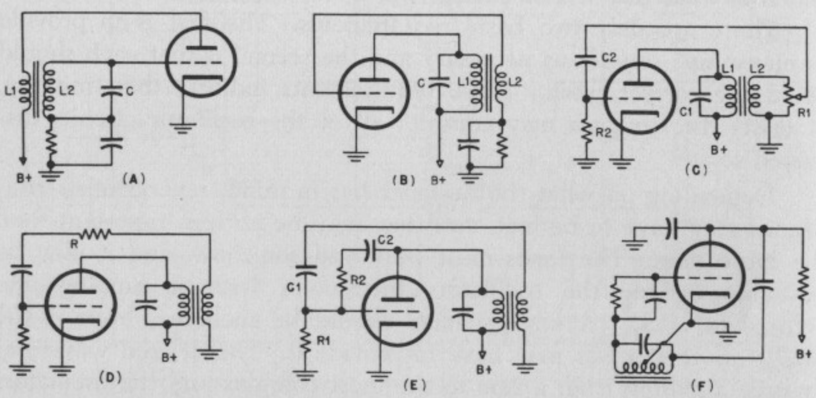


Fig. 20-21. Six feedback oscillators: (A) tuned-grid, (B) tuned-plate, (C) tuned-plate with feedback level adjustment, (D) another adjustment method, (E) adjustment by adding negative feedback, (F) Hartley.

grid leak. Values are not critical and experiments to find correct values may be begun around 50,000 ohms and .01 μf. If the time constant (product of *R* in megohms and *C* in microfarads) is too long, the oscillator may not start or may go on and off at intervals.

The amplitude of the positive feedback is a very important factor in determining output waveform and frequency stability. It is controlled within limits by the grid-leak arrangament because the increase in grid current which accompanies an increase in feedback amplitude automatically increases the negative bias and lowers the amplification of the tube. The grid leak controls amplitude only within a fairly narrow range, however.

In the two oscillators, (A) and (B) in Fig. 20-21, the principal factor controlling feedback is the turns ratio of the transformer. Ideally it should be wound so that the voltage reaching the grid due to the tube output is exactly the same as the voltage at the grid which produced that output. A feedback oscillator is almost a perpetual-

motion machine in that it supplies its own input. (The fact that external plate and filament power supplies are necessary, however, prevented the first oscillator inventor from rushing to Washington with a final solution to the perpetual-motion riddle.)

If the feedback is too great, the output waveform distorts and frequency stability suffers. Increased above a certain point, the feedback causes relaxation oscillations with a roughly sawtooth waveform having sharp, needlelike peaks at each apex. The frequency is then extremely sensitive to the slightest changes in supply voltage or heat movement of the tube elements and is not useful for electronic music unless synchronized because of the excessive frequency drift.

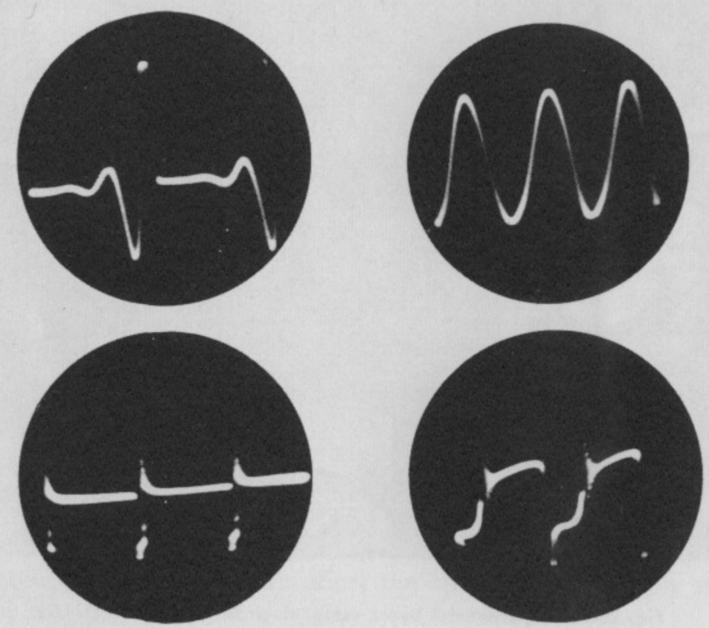


Fig. 20-22. Waveforms show results of excessive oscillator feedback.

The oscillograms of Fig. 20-22 show what happens when feedback in a conventional oscillator is too great and when the time constant of the grid leak is too long. The perfect sine wave results when feedback is reduced to the optimum value by any of the methods described here.

The usual noncommercial constructor shies away from winding his own coils for these oscillators and prefers to use available ones. That is made possible by the circuits of (C), (D), (E), and (F) of Fig. 20-21.

The circuit of (C) is a tuned-plate oscillator like that of (B). The transformer, however, is a standard interstage unit: L_1 is usually the higher-impedance winding, tuned by C_1. Normally the voltage across L_2, which usually has a stepdown ratio of only a very few times, is much too high for proper feedback. However, a potentiometer R_1, which may be anywhere between 10,000 and 100,000 ohms or even more, is shunted across L_2 and the arm is adjusted until just enough voltage is fed to the grid to sustain oscillations. At this point the waveform will be almost pure sine (if the grid-leak components R_2 and C_2 are chosen correctly) and the frequency will be most stable and nearest to the resonant frequency of L_1 and C_1.

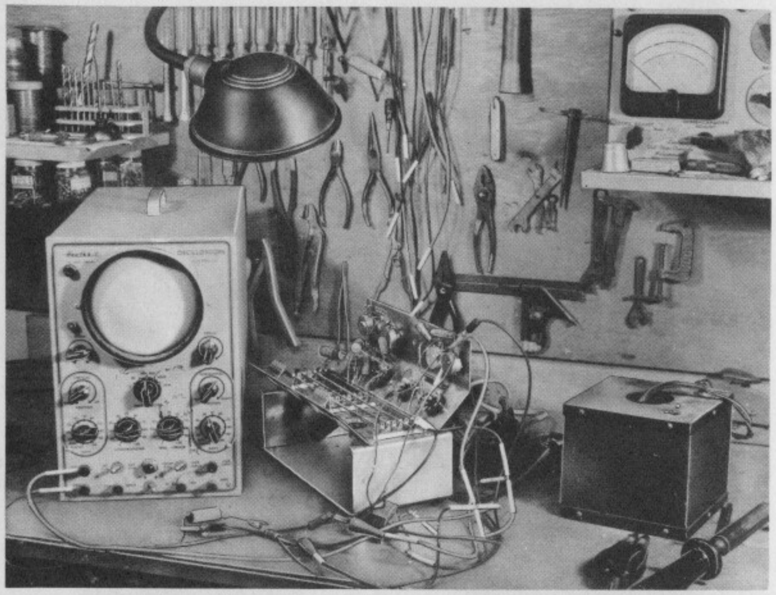


Fig. 20-23. Experimental bench setup of circuit of Fig. 20-21 (C).

Once the correct values for the upper and lower halves of R_1 have been found, the potentiometer may, of course, be replaced with a pair of ½-watt resistors.

Figure 20-23 shows an experimental setup of the oscillator of (C) in Fig. 20-21, which was used for making the oscillograms of Fig. 20-22.

The same trick may be done in a slightly different way by using the circuit of (D). Here resistor R is a series limiter. This, incidentally, is a well-known form of resistance stabilization ordinarily used to stabilize frequency. It is also useful here, of course, to compensate for the fact that the transformer turns ratio is not ideal.

The circuit of (E) in Fig. 20-21 illustrates a method of combined

stabilization and feedback correction original with the writer (although it was probably original with other workers at prior times). As usual with the standard interstage transformer used, the positive feedback is too great for stable sine-wave oscillation. The positive feedback is produced in the usual way.

The excess of positive feedback is remedied by adding negative feedback. An oscillator of the feedback type may be considered simply as a perfectly standard amplifier which produces its own input. The amplitude of oscillation depends on two things: the amount of coupling between plate and grid, and the amplification of the tube. The previous circuits have been stabilized by varying the coupling. In (E) of Fig. 20-21 the feedback voltage is lessened by reducing the amplification of the tube. R_2 is a simple negative feedback resistor from the plate to the grid. C_2 is a blocking capacitor which should have a low reactance compared to the resistance of R_2.

After the circuit is connected, R_2 is varied (use a 10-megohm potentiometer) until the circuit just oscillates. The positive feedback voltage is then correct. In addition, since the amplifier is operating with a fair degree of negative voltage feedback (the degree of permissible negative feedback depending on how much too large the transformer secondary voltage is), it is itself stabilized to a large degree against changes in amplification caused by fluctuations of supply voltages. Since, in an audio oscillator of this kind, changes in plate current and amplification, rather than changes in tube-element capacitances cause most of the undesirable frequency irregularities, stabilizing the amplifier helps matters considerably. An additional effect of the negative feedback is to lower the tube's effective output impedance, reducing the importance of any tube output capacitance that may affect the frequency, especially in the higher octaves.

(F) of Fig. 20-21 is a standard Hartley oscillator, which is usable for electronic music. Since, however, most tapped inductors ordinarily available are center-tapped (often primaries or secondaries of push-pull transformers), there will be too much feedback. This can be reduced by inserting a resistor at point X.

Output may be taken from the plates of any of the oscillators shown in Fig. 20-21 without a great effect on frequency, provided the impedance of the load is high and its shunt reactance low. For maximum freedom from loading effects, however, especially where the load is keyed or otherwise altered during operation, electron coupling is desirable. Almost any feedback oscillator may be used in an electron-coupled circuit. The method illustrated in Fig. 20-24 is to use a pentode tube and employ the screen as the oscillator anode.

The circuit of Fig. 20-24 is exactly the same as (A) in Fig. 20-21 with those exceptions. The screen does not, of course, draw much current, but the electron stream passing through it to the plate is modulated by the oscillations. The plate output, taken across a standard load resistor, is of the oscillation frequency and may be passed on to following stages. The output capacitor is uncritical, values between .01 and .05 μf being suitable.

Changes in the load which would affect the plate current have little effect on the oscillator circuit, and reactive loading of the

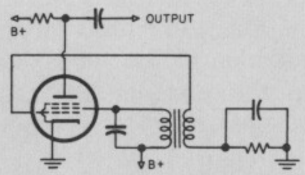


Fig. 20-24. Oscillator output taken by electron coupling.

plate has almost none at all. The screen supply voltage should be somewhat less than that supplied to the plate and should, of course, be under the maximum specified by the tube manual. It may be obtained from a tap on the power supply bleeder resistor. The tap should be bypassed to ground.

OTHER OSC'LLATORS

We turn next to negative-resistance and resistance-capacitance oscillators.

Negative resistance is easily visualized as a circuit property which exactly disobeys Ohm's law. Such disobedience is not quite the heresy it seems, for it takes place only in a figurative sense under special conditions.

The primary example of the negative-resistance oscillator is the dynatron, first developed by A. W. Hull many years ago. It employs a tetrode tube in the circuit of Fig. 20-25. The grid is grounded for a.c. and the screen-supply voltage is higher than the plate-supply voltage. The oscillator depends for its functioning on secondary emission.

The potentials of the grid and screen largely determine the number of electrons which leave the cathode, but the plate voltage determines the speed at which they travel, and of course how hard they hit the plate. In any tube whose plate voltage is above a certain minimum value, electrons strike the plate with such force that they

bounce off it. The electrons which bounce away from the plate are in a sense *emitted* by it, and the sum of the electrons emitted in this way constitutes secondary emission.

In a triode, the plate is the only positive electrode, so the secondary emission electrons all eventually return to it. As a result, secondary emission has no effect on total plate current. In the circuit of Fig. 20-25, however, the screen has a higher positive voltage then the plate, so the secondary emission electrons, seeking the most positive point as a haven – as they always do – are drawn to the screen and pass through it back to the power supply.

The amount of secondary emission depends on how hard electrons from the cathode strike the plate (assuming screen voltage is constant), which in turn is determined by the plate voltage. And when the plate voltage is above a certain minimum value, each electron from the cathode actually dislodges from the plate several others, the number depending on the plate voltage. Thus – and this is the crux of the matter – as the plate voltage is *raised*, the plate loses more electrons by secondary emission than the cathode supplies to it. Plate current *decreases* (and may actually reverse). This is certainly in contravention of Ohm's law, which says that as voltage is

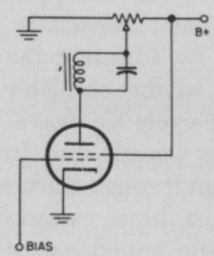


Fig. 20-25. The dynatron oscillator.

raised, current, too, ought to rise. (Of course, screen voltage must always be higher than plate voltage so that the secondary electrons will go to screen, and not return to the plate.) Since effective resistance can be defined as voltage divided by current and since plate current decreases when voltage rises instead of increasing as it ought to, the plate resistance of the tube is said to be negative.

Now let us see how this produces oscillation. Initially, plate and screen voltages are steady, as is tube current. Now a small random pulse of plate current passes through the tuned circuit to the supply. It causes a small instantaneous voltage drop across the tuned circuit, which makes the plate voltage a little less positive. Because of that, secondary plate emission drops and fewer secondary electrons are bounced from the plate and lost to the screen – so that velocity of the primary (cathode) electrons has momentarily decreased. According to the explanation above, plate current then increases. The increase in current passing through the impedance of the tuned circuit drops plate voltage even more and the action snowballs until plate voltage and secondary emission are at a minimum. This condition cannot remain in force, however, because of the tuned circuit, which has a high impedance only to a.c. As soon as conditions become steady, the impedance of the tuned circuit starts to decrease. There is no longer so much of a voltage drop across it and the plate starts to go positive again.

As it does so, secondary emission increases. This decreases plate current, making the plate even more positive, and again the action snowballs until the plate is at maximum positive potential, with minimum current. Because of the "flywheel" action of the tuned circuit, the a.c. plate voltage peak exceeds the quiescent positiveness and after the peak starts to recede in the negative direction, the entire cycle repeats itself.

This kind of oscillator has several excellent points. It is, first of all, a two-terminal oscillator, meaning that only the two ends of the *L-C* circuit require connections. That eliminates the requirements for a coil secondary or for taps with consequent difficulty of adjustment. Second, it is a stable oscillator, especially when oscillation amplitude is kept low (easily done by adjusting the grid-bias voltage). Third, vibrato is easily applied to the oscillator by superimposing on the grid bias voltage a 5- to 8-cycle a.c. wave. This makes for a pleasant combination of amplitude tremolo and frequency vibrato (the latter because it varies the current through the tank coil).

The tank circuit must have reasonable Q because the circuit will not oscillate unless the impedance of the tank circuit (approximately L/RC) is equal to or greater than the equivalent negative plate resistance. In practice this is not a difficult condition to meet.

Frequency and amplitude of the dynatron are not entirely stable over long periods, however, because they depend on secondary emission which varies among tubes of the same type and which within one tube changes with age. The transitron oscillator is an improvement on the dynatron which overcomes this disadvantage. Its circuit appears in Fig. 20-26.

TRANSITRON OSCILLATOR

The transitron does not depend on secondary emission and is more stable than the dynatron. As shown in Fig. 20-26, the suppressor is biased negative and the plate voltage is kept low. In the quiescent state, a large number of the electrons from the cathode are repelled

by the negative suppressor before they can reach the plate; they go to the screen. The first pulse of screen current passes through the tank, across which there is a drop, making the screen voltage go in the negative direction. This small negative pulse passes through the capacitor C to the suppressor, making it more negative. This stops more electrons that would otherwise have gone to the plate, passing them to the screen instead. The additional screen current causes additional drop across the tank and makes the screen and suppressor still more negative. The action snowballs until the screen approaches maximum negativeness, when the negative buildup begins to slow down. When the peak is reached, nothing is transmitted through capacitor C to the suppressor, and it begins to return to its quiescent voltage, which is more positive than the present state. The increasing positiveness causes fewer electrons to go to the screen; the lower

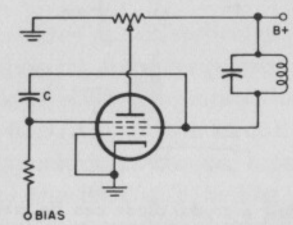


Fig. 20-26. The transitron oscillator.

screen current has a lower drop across the tank, so the screen voltage becomes more positive. The positiveness is transmitted to the suppressor through capacitor C, and again the action snowballs until the suppressor is at maximum positive, plate current is maximum, screen current is minimum, and screen voltage is most positive. Then the cycle begins again.

As in the dynatron, oscillation amplitude may be controlled by applying some negative bias to the control grid (not shown in the figure), and vibrato can be added in the same way. Stability of the transitron is excellent. Dr. Cledo Brunetti of the National Bureau of Standards reports that a well-designed transitron will show a frequency change of only .001% for a 33% change in B-supply voltage to the screen. Stability of that kind is well worth notice.

The waveshape generated by either of these oscillators is an excellent sine when oscillation amplitude is kept fairly low by adjusting the grid bias. At higher amplitudes it begins to distort, which is desirable for certain musical instrument designs, but frequency stability suffers.

CRYSTAL OSCILLATOR

Ordinary germanium crystal diodes such as the 1N34 can be used as oscillators without vacuum tubes because of a negative resistance characteristic. Such an oscillator is not especially suitable for music use, however, as oscillation depends on heat within the crystal. In addition, during this author's experiments, three crystals (at a little less than a dollar each) were ruined in quick succession because the voltage needed to make it oscillate was greater than the crystal would take without damage.

One crystal (1N34) oscillator is shown in Fig. 20-27. This is a relaxation oscillator. Since an excess of inverse voltage across the crystal changes its characteristic from positive to negative resistance, at one point it has zero resistance. Voltage is applied to the capacitor

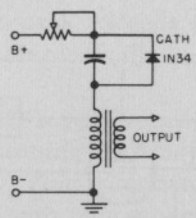


Fig. 20-27. How a crystal diode can be made to oscillate.

from the supply. Because of the resistor there is an *R-C* time-constant circuit and the capacitor takes time to charge. When it charges to the point where the crystal exhibits zero resistance, the crystal shorts the capacitor and discharges it, whereupon the cycle starts over again. Because of the fact that the resistance of the crystal changes rather gradually as it approaches the zero point instead of breaking down suddenly as does a neon lamp or gas tube (thyratron) the output voltage is not a perfect sawtooth.

R-C OSCILLATORS

There are two principal types of *R-C* oscillators – Wien-bridge and phase-shift – though there is at least one other employing a parallel-T network in the feedback loop. Both are very stable when correctly adjusted.

The Wien-bridge oscillator is the most common type and is used extensively in audio test generators. It requires two tubes (or a dual tube), and consists simply of a two-stage audio feedback amplifier of the usual resistance-coupled variety. Because two stages are used, the plate voltage of the second is in phase with the grid of the first. A Wien-bridge connects these two points. The resistors

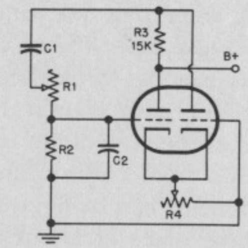


Fig. 20-28. A Wien-bridge oscillator with provision for varying the frequency and feedback slightly.

and capacitors of the bridge are chosen so that a.c. of only one frequency passes through it without phase shift, while all others are shifted. The frequency whose phase is not shifted is the only one which will be fed back completely and is the frequency at which oscillation takes place.

Figure 20-28 illustrates a Wien-bridge oscillator circuit used extensively by the author for fixed-frequency operation. The chassis in Fig. 20-29 is a bank of seven frequency dividers for a new organ. The eighth tube is a 6SN7-GT in the circuit of Fig. 20-28, used as a master oscillator to synchronize the six lower octaves.

The left section of the 6SN7-GT in Fig. 20-28 is a cathode follower. The right section is a grounded-grid amplifier whose cathode is coupled to the cathode follower. For stable sine-wave operation, the signal fed back to the first grid through the Wien-bridge phase-

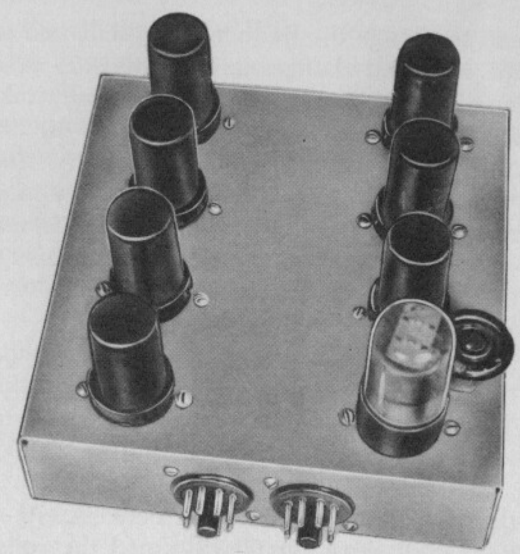


Fig. 20-29. A chassis with seven dividers.

shift network R_1-R_2-C_1-C_2 should be just equal to the original grid voltage, as in any feedback oscillator. Since the voltage loss through the phase-shift network is exactly three times, when $R_1 = R_2$ and $C_1 = C_2$, the net amplifier gain should be only three. This is adjusted with R_4, which can be a variable resistor for test purposes and may be replaced with a fixed resistor later.

The phase-shift components can be figured roughly by remembering that oscillation will take place at about the frequency at which the resistances and reactances are equal. One megohm and 150 $\mu\mu$f, for example, should cause oscillation at about 1,000 cycles.

The Wien-bridge oscillator is somewhat sensitive to changes in supply voltages, especially when the gain of the amplifier is a little more than optimum. Changes in supply voltages may change the gain,

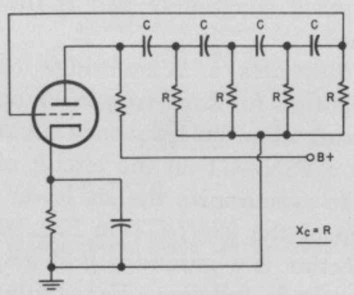


Fig. 20-30. Standard phase-shift oscillator. Use a high-mu triode or a pentode.

which will tend to change the frequency of oscillation to some extent. Changes of this kind and changes in characteristics caused by changing or varying the tuning components are usually taken care of in test generators with a small lamp in a cathode circuit. For single-frequency operation, however, that is not necessary. In the 6SN7-GT oscillator on the chassis of Fig. 20-29 a 20% change in supply voltage varies the frequency only about one-half tone. Under normal line-voltage variations, the change in pitch is insignificant. If used in a particularly bad district where line voltage is quite variable, voltage-regulator tubes may be used in the supply.

The second *R-C* oscillator is the phase-shift type. It is more stable than the Wien-bridge circuit and is more reliable at very low frequencies, for which reason it is often used for generating the 5- to 8-cycle vibrato signal. It is diagrammed in Figs. 20-30 and 20-31. In Fig. 20-30 a standard voltage amplifier has its output coupled to its input through a four-section phase-shift network. At one frequency the reactance of each capacitor *C* will equal the resistance of each resistor *R*. (All *R* and *C* values are identical.) The phase shift in each

section will then be 45 degrees and the total shift through four sections 180 degrees. The output at that frequency is then in phase with the input and the circuit oscillates.

Figure 20-31 shows the vibrato oscillator used in the Lowrey Organo. This circuit is identical to that of Fig. 20-30 except the phase-shift network has only three sections. Each section must then shift phase by 60 degrees. The formula for calculating the components (derived by the author from the more complex basic formulas) appears

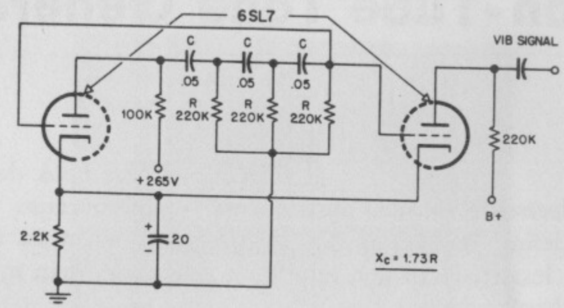


Fig. 20-31. Vibrato circuit used in the Organo.

in the figure. In this case the oscillation frequency is about 8 cycles. The second section of the 6SL7 is simply an amplifier to amplify the vibrato signal before sending it to the circuits it modulates.

There is much more loss in the phase-shift networks of Figs. 20-30 and 20-31 than in that of the Wien-bridge oscillator. A medium-mu triode is therefore not suitable. For a three-section network a high-mu triode such as the 6SL7 (one section), 6SC7 (one section), or 6SF5 is appropriate. Oscillators with four-section networks may use the same tubes or a pentode such as the 6SJ7 may be found more reliable. In each case amplification should be adjusted (by choice of cathode resistor) for minimum gain consistent with reliable oscillation.

While all components of the phase-shift network should theoretically have the same value, fine tuning can be provided for when the oscillator is used as a musical tone generator by making one of the resistors *R* variable over a small range.

Chapter 21

Non-Tube Tone Generators

BECAUSE this book deals primarily with *electronic* musical instruments – and because the writer prefers to define "electronic" as dealing with electron tubes – we shall devote less space to non-tube tone generators than to those discussed in Chapter 20.

Non-tube generators fall generally into two classes: electromechanical generators, and amplified acoustic generators. In the first class we find photoelectric, electrostatic, and electromagnetic generators, and instruments which employ the principle of playing back pre-recorded notes. In the second appear electric pianos, amplified reed organs, amplified guitars, most electronic chime and carillon systems, and the like. In most instruments which use acoustic generators, the design problems are largely limited to picking up and amplifying the tones. That kind of problem is very nearly standard in electronics (though, of course, special methods are sometimes necessary) and our treatment is limited in this book to a description of the Wurlitzer organ, which generates its tones with banks of air-driven reeds.

PHOTOELECTRIC GENERATORS

So far as the writer knows, only two photoelectric organs have been built for the commercial market, the German Welte organ and, very recently, one by an American manufacturer, Baldwin. Although the principles of photoelectric tone production are not complex, the practical problems of construction and production are.

In principle, photoelectric systems are similar to sound-on-film, in which a narrow strip at the edge of the film contains a continuously varying pattern of either variable density or variable area.

Figure 21-1 illustrates the variable area system. The strip at the left edge of the film varies in area as the film travels along. A steady light source in the projector shines a beam through this strip onto the cathode of a photoelectric tube. As the area of blackness of the strip varies so does the amount of light reaching the phototube. The tube's output varies accordingly. Fed into an amplifier, the output variations, which are at an audio rate, cause sound to be heard in the loudspeaker. In the variable-density system, the area remains constant while the opacity varies.

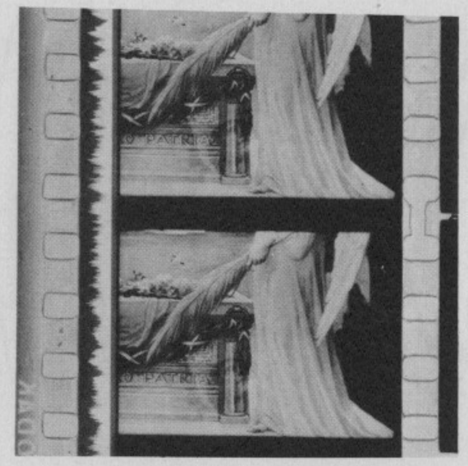


Fig. 21-1. Variable-area recording.

In electronic music, however, we are concerned not with a constantly changing recording but with a steady repetition of single tones. It is standard practice, therefore, to make "re-entrant" light recordings such as the one illustrated in Fig. 21-2.

Figure 21-2 shows a segment of a disc made of glass. The entire disc is clear except for the black waveform pictures in 18 concentric bands. If a slit of light is focused on one of these bands as the disc revolves and a photoelectric tube is placed on the other side of the disc so that whatever light passes through the disc hits it, the amount of light reaching the tube varies in accordance with the area of clear glass between the two. The tube output current varies in exact accord with the shape of the black waveform picture. If the disc is turned fast enough so that the variations are at an audio rate, the sound from the speaker will correspond to the waveform pictured and will have a frequency or pitch determined by the number of waveforms scanned each second.

This scheme is the electronic musical equivalent of variable-area film recording. A variable-density system may be used instead, by having each band consist of a strip of constant width but of varying degrees of opacity. This is much harder to make. Or, instead of a transparency on glass, the disc may be of metal, with holes in it. This is a chopper, which alternately interrupts and passes the light.

OBTAINING CORRECT PITCHES

The most obvious (and the most accurate) way of providing disc scanning patterns which give the correct musical pitches for an entire

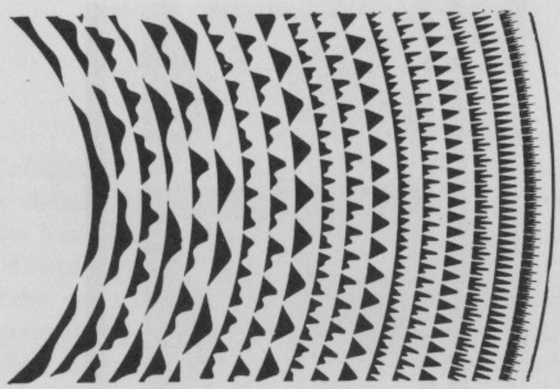


Fig. 21-2. Segment of a glass tone disc.

instrument is to have a separate disc for each of the twelve fundamental notes of the chromatic scale, and to drive each at the right speed.

For example, the C disc must generate a tone of 65.41 cycles for the lowest C or the organ manual. If it has a single pattern on the circular strip which is to generate this note, it must revolve 65.41 times per second or 3924.6 r.p.m. The speed necessary can be reduced by any desired factor, however, if the number of patterns around the strip for that note is multiplied by the same factor. If, for instance, there are ten patterns, the speed can be reduced to 392.46 r.p.m. and the pitch will still be 65.41 cycles.

The next band on the same disc would probably be designed to produce the next octave pitch 130.8 cycles, which means it must have twice as many patterns for the same speed. The octave above must have twice as many patterns as that, and so on, until the outer band has enough patterns to produce the highest pitch wanted. Obviously, twelve such discs, each revolving at a different speed, are necessary. The complexity of a gearing system needed to drive twelve discs at

different, yet exact, speeds is a deterrent to such designs, but the problem can be simplified to some extent by keeping in mind that pitch errors of about 0.25% or less can be neglected. This is equivalent to 1 part in 400, which means that a nominal 440-cycle note could be 441 or 439 cycles without too bad an effect. Even this error, however, can be detected by people with acute ears, and for the sake of safety the error should be limited to 0.1% or 1 part in 1,000. That means that actual frequency should correspond to nominal frequency to four significant figures, as given in the frequency chart on page 7. To be specific, F^{21}, for instance, listed as 87.31 cycles, may vary only 0.1 cycle, or from 87.21 to 87.41; C^{76}, listed as 2093, may vary 1 cycle either way, from 2092 to 2094.

Mathematically, of course, there is no reason why a single speed would not be adequate for all the notes. The maximum permissible speed would be that which, in cycles per second, is a submultiple of all the desired audio frequencies. Other speeds are possible, too, if some of the discs contain a single pattern for the lowest note and others have more than one.

In a practical commercially designed organ, that speed has worked out to approximately one revolution every two minutes; the revolving element is a drum rather than a series of discs. The extremely slow speed, however, poses other problems, such as regulation, for the slower the speed, the harder it is to keep it constant.

KEYING AND TONE SHAPING

Photoelectric organs can be keyed electrically by simply closing switches in series with separate lamps which provide light sources for the pattern bands. A more common method, however, though more complex for the individual constructor, is mechanical – the keys operate shutters which cover the patterns when they are not desired.

Tone coloration may be dependent on the patterns themselves, as in Fig. 21-2, which is the scheme of a European organ. A separate disc is provided for each note; the eighteen bands on the disc shape the waveform of the light reaching the phototube to produce eighteen different tone colors, corresponding to the same number of organ stops. A system of shutters, operated magnetically by the playing keys and stop switches, allows light to pass through an appropriate part of the disc. The discs are glass, with photographic emulsion, and the patterns are printed photographically from a master negative.

In any photoelectric organ, it is important that the patterns on the discs or drum be re-entrant. That is, it is not sufficient merely to have the right number of patterns or holes in a given circular band; they must be evenly spaced. To illustrate, Fig. 21-3 (A) shows a disc with eight holes in the first band, sixteen holes in the second. If the disc is rotated at 367.5 r.p.m., the inner band will produce G^{11} at 49 cycles and the outer G^{23} at 98 cycles.

Figure 21-3 (B) shows a similar disc with the same numbers of holes, but with the outer ring unevenly spaced. When the light is passing through the sixteen holes in turn, a higher frequency than 98 cycles will be produced because the close spacing makes the cycles occur too quickly. When the one wide-spaced pair of holes comes around, there will be a sudden low-frequency element.

The total result, however, strange as it may seem, will be to produce the same frequency as with evenly spaced holes (though the sound will probably be unpleasant. This has been demonstrated by

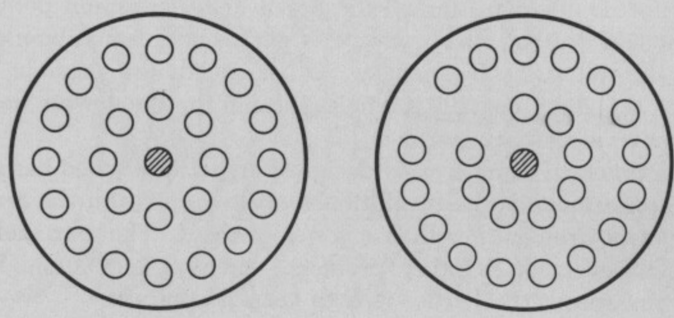


Fig. 21-3. Patterns on rotating generator must be re-entrant.

the writer and is due to the fact that there is a sudden phase shift each revolution. This constitutes a form of phase modulation of the frequency caused by the hole spacing and is just enough to cause an apparent change of net frequency back to the frequency which would be produced by even spacing. Whether the frequency goes down to that which would have been produced by the same number of evenly spaced holes or up to that which would be produced by the next larger number of evenly spaced holes depends on whether or not the uneven gap is greater or larger than one-half the hole spacing.

This illustrates, among other things, the fact that the number of holes in each circle must be a whole number. If the designer starts with a certain speed in mind, then figures the number of holes to be placed in each band, he will probably come out with a fractional number – say, 4.5 holes. It is impossible, naturally, to make half a hole, and the only other alternative is to space the holes as if it were possible – to lay out the design by dividing 360 degrees by the number

of holes, 4.5, then drill what holes are possible and leave an uneven space in one spot.

The correct procedure is to figure on a certain number of holes, then calculate the necessary speed. A formula can easily be worked out: Frequency divided by the number of holes or patterns gives the speed in revolutions per second. This is multiplied by 60 for r.p.m. Thus:

$$\text{r.p.m.} = \frac{60f}{n}$$

where f is the audio frequency desired and n is the number of holes or patterns. Note that the spacing of the holes from the center of the disc makes no difference in the pitch.

Another method of tone coloring in a photo organ is disclosed by Alexander Roth in Patent No. 2,513,109. His instrument uses a rotating-disc system to produce sine waves, one for each note. Each note is keyed by lighting a lamp, which shines through a band of slits in a disc. With switches, the player can make more than one lamp light when a key is pressed – he can light lamps of the notes corresponding to harmonics of the key he presses. Potentiometers regulate the current through the individual lamps; the amount of light determines the loudness of each harmonic, and the player can thus make up from its harmonics whatever tone color he wishes. The system is fundamentally equivalent to the Hammond Organ's harmonic synthesis system, which will be described in detail in the next chapter.

Photoelectric organs are very difficult to build and operate satisfactorily. It can be done, however, by constructors with sufficient mechanical skill. The author would be most interested in hearing from readers who have done so.

ELECTROSTATIC GENERATORS

The electrostatic system of tone generation is used in the Compton Electrone, a British instrument which we have described in Chapter 15. It is also the basis of most amplified reed systems, including the Wurlitzer.

Electrostatic pickups are based on the simplest principles of capacitance. A capacitor is a pair of conducting surfaces in proximity to each other, separated by a nonconductor or dielectric. When a voltage is applied between the plates, there is a sudden rush of electrons from the positively charged plate to the negative one through the voltage source. 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.

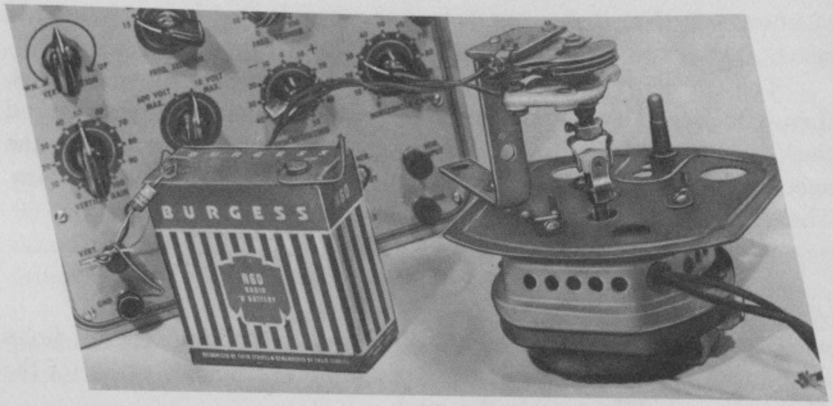


Fig. 21-4. Simulating a capacitive (electrostatic) generator on the experimental bench.

The crucial point here is that the number of electrons that make the rush is determined by the magnitude of the applied voltage, the character (good or bad insulating properties) of the insulation between plates, the distance between the plates, and the area of the plates themselves. To distill things further, if the d.c. voltage and the dielectric remain the same, the area of the plates and the distance between them determine the magnitude of the initial current flow.

Now let us set up a laboratory experiment such as we have pictured in Fig. 21-4 and diagrammed in Fig. 21-5. A variable capacitor with its plates fully meshed is connected to a resistor and battery in series. The moment we connect them there is an initial rush of current from the positive plate through the battery and the resistor to the negative plate. In a small fraction of a second the capacitor charges fully and the current flow comes to a gradual stop.

Now we give the capacitor shaft a half turn or so. The area of

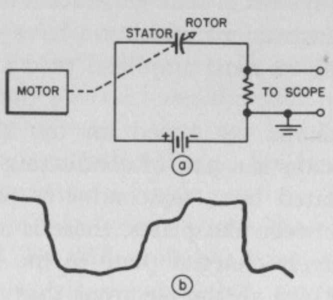


Fig. 21-5. Diagram of the electrostatic experiment, and resulting waveform.

the rotor which is between the stator plates is now smaller – there is less capacitance, just as if we had lopped off a piece of the rotor plates. Some of the electrons on the effectively smaller rotor plates are now crowded off and return through the resistor and battery to the stators, creating a little rush of current through resistor and battery in the opposite direction to the original charging current.

We will find, in fact, that every time we rotate the rotors and change the capacitance, there will be this rush of current, its direction depending on whether we decrease the capacitance (discharge) or increase it (charge).

Now we haul an old phonograph motor out of the stockpile and connect the shaft to the shaft of the capacitor. When we start the motor the capacitor's rotors begin whirling at the same speed as the motor. And with each revolution the capacitance reaches a minimum (plates unmeshed) and a maximum (plates fully meshed). As a result, current through the resistor and battery reaches a maximum, first in one direction, then in the other. In fact, the current takes the form of a.c. In the actual experiment shown in the photo the resistor was 15 megohms, the d.c. supply 90 volts, and the capacitor a 10-30-$\mu\mu$f unit. The current through the resistor created a very small a.c. voltage, too small to measure. Since the motor shaft drove the capacitor at about 900 r.p.m., the frequency of the a.c. was equal to the (r.p.m. divided by 60), revolutions per second, in this case 15 cycles. It was fed to the vertical input of an oscilloscope and the waveform appears in Fig. 21-5(B)

It is obvious that the waveform in Fig. 21-5(B) is not a sine, but since the a.c. was at an audio frequency it could easily be heard when a pair of headphones were placed across the output of the oscilloscope amplifier. The waveform depends only on the shape of the capacitor plates.

With that in mind we can design electrostatic generators to order, in at least two ways. The first involves varying the effective plate area. The specific method is taken from L. E. A. Bourn's Patent No. 2,522,-923, though Bourn's method is presented here only in bare principle.

Figure 21-6 shows the two necessary parts. A disc of insulating material at *(b)* has five annular rings of metal mounted on it. The rings are shown as the shaded push-pull sawtooth-shaped parts. This is the stator disc. Notice that the outer ring of the stator has thirty-two waveforms on it and each inner ring half the number of the next outer one. Each stator ring is electrically connected through a playing key to the grid of the first amplifier tube, as in *(c)* of Fig. 21-6.

The rotor appears in *(a)*. It is composed of a network of rigid wires mounted on a rim. The solid radial lines in the drawing represent the wires. All the wires are connected together and to the positive end of the polarizing potential, as shown in *(c)*. The rotor is placed parallel to the stator and very close to it.

Notice next that in the rotor there are radial wires of five different lengths. The result is that the outer group numbers thirty-two – the same number as the waveforms on the outer ring of the stator – and that their spacing is exactly the same as the stator's waveforms. The same holds true for each of the four inner rings.

Now let us set the mechanism in motion. When it is standing still, there is a constant capacitance between each wire of the rotor and

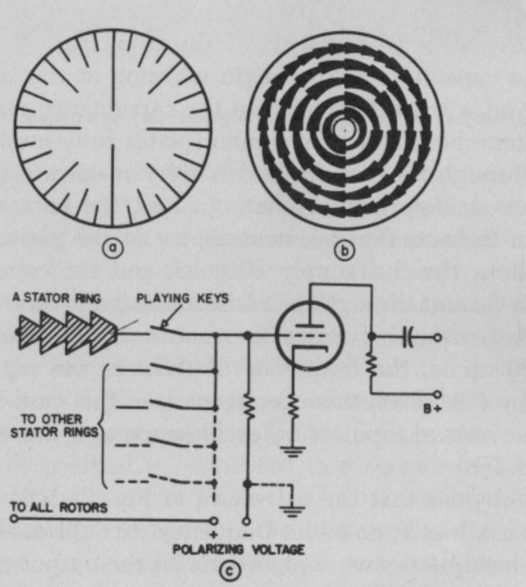


Fig. 21-6. Components of the Bourn generator used in slightly altered form in the Compton.

the portion of an annular stator ring which faces it. As the rotor turns, each stator wire scans its wave pattern opposite. When a wire faces the wide portion of the waveform the capacitance is great; as it goes toward the narrow part the capacitance decreases. We thus have five continuously varying capacitors. Because of the 2-to-1 relationship in the number of waveform patterns and rotor wires between each ring and its neighbor, the outer ring's capacitance varies fastest, the next half as fast, and so on.

We thus have a mechanism and a circuit for refining the results obtained in our laboratory experiment. Because of the waveshapes on the stator, we can, by closing the appropriate playing switches, get from the output of the tube a note of the scale in sawtooth form over

a range of five octaves. The speed of the disc determines what note it shall be. By providing twelve such discs, each rotating at an appropriate speed, we can make available five full octaves of notes. By placing six or seven annular rings on each rotor we can, with twelve discs, produce six or seven octaves.

The same trick can be attacked from another angle. In Fig. 21-7 we have a disc with a scalloped edge. Close to its edge and facing it edgewise is a small electrode. Connections are as shown in the figure. Now, as the disc rotates, its metal edge alternately approaches and draws away from the pickup electrode. The resulting capacitance between the two varies, and again an audio voltage appears across the resistor.

The electrostatic method has also been used extensively to make "electric pianos" and amplified reed organs. The method for a piano

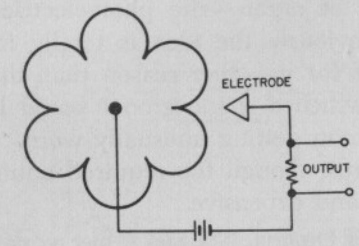


Fig. 21-7. Scalloped-edge disc produces tone electrostatically.

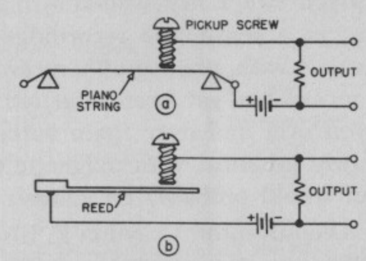


Fig. 21-8. Electrostatic pick-up from strings or reeds.

is illustrated in Fig. 21-8(A). A pickup screw is mounted close to the string and the polarizing voltage is connected. As the string vibrates, the capacitance between it and the string varies at an audio rate and there is output across the resistor. In an amplified reed organ the pickup screw is placed close to a selected portion of the vibrating reed as in Fig. 21-8(B).

Electrostatic systems have the disadvantage that high voltages are used on more or less exposed parts of the mechanism, with consequent danger of shock. The high voltage is desirable even though a lower voltage will work because of a second disadvantage, the systems susceptibility to hum pickup due to the high impedances involved. Higher voltage means higher signal-to-noise ratio.

The fact that electrostatic pickup depends on variations in capacitance immediately suggests another approach which eliminates the high voltage and can greatly reduce hum – using the variable capacitance to vary the frequency of an oscillator and produce frequency modulation.

Practical circuits for the purpose will quickly suggest themselves, especially for electric pianos and reed organs which need no electrical keying circuits.

RECORD REPRODUCTIONS

One of the grandest ideas in scope ever kept in mind by the author – and undoubtedly by many readers – is to make an instrument with a series of sound recordings as tone sources. If we could make a record of several ordinary acoustic insturments, each record containing a single pitch played on a single instrument, then have all the records playing continuously, we could select the output of any record with playing keys and produce a precise replica of the sounds of each instrument.

Fascinating as the idea is, it is very difficult to make practical, although one fairly well-known kind of organ – the photoelectric – does, in a sense, use recordings. Obviously the idea is totally impractical with phonograph records, if for no other reason than that no record has yet been invented in which a single groove could be played over and over again without soon getting unusually worn. It is fairly practical with magnetic records, though the required equipment would probably be voluminous and expensive.

One inventor, Graydon F. Illsey, of Omaha, Nebraska, has worked on the idea in patent No. 2,533,961, using magnetic wire. In bare outline, he has a series of discs, each with a length of recorded wire mounted in a shallow groove around its edge. If the organ has five octaves, there are 61 discs for each instrument to be reproduced, and each disc has a recording of one tone. If there are ten instruments to be reproduced, 610 discs are necessary – each with its own magnetic playback head. While there is no reason why the idea would not work, it would probably not be cheap to produce.

The same problem rears its head here as with other rotating-disc ideas – this fact that the waveforms must be reentrant. That is, the joined ends of the circular recording must have waveforms in phase. To put it another way, each record must contain an integral number of waveforms. The problem is not difficult with photoelectric and electrostatic discs, for they are usually provided with the necessary patterns by an artist or draftsman when the original models and the production dies are made. In any case the waveforms are visible.

In magnetic records the waveforms cannot be seen, so another method is necessary. John Hays Hammond, Jr., has given an excellent answer in his Patent No. 2,475,742. Figure 21-9 shows how it works.

The note to be recorded is played into the microphone, amplified, mixed with supersonic bias, and sent to the recording head. The disc on which the endless recording is to be made is driven by a synchronous motor (thorugh a mechanical filter to remove transient speed variations) which may be powered either by the 60-cycle line or by a variable 59- to 61-cycle source.

At the other edge of the disc is a monitor head which picks up the recording just made and sends it through the necessary equalizer to one pair of plates of an oscilloscope. The other pair of plates is fed a sweep signal triggered by the same amplifier.

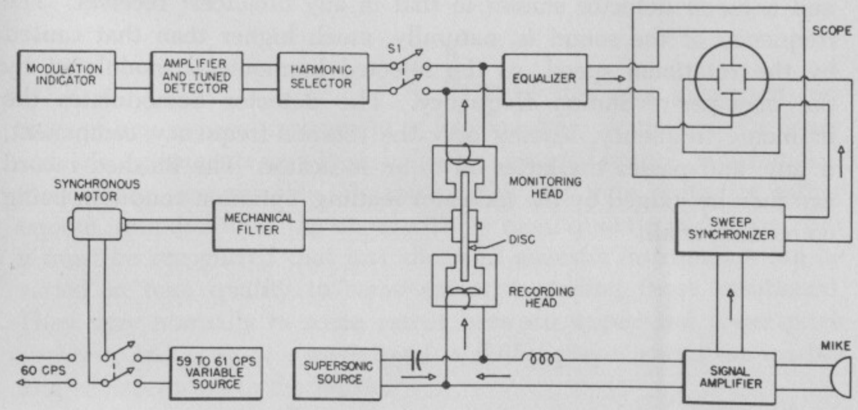


Fig. 21-9. A method of making re-entrant magnetic recordings.

The waveform on the scope quickly shows whether the track is re-entrant. If not, the operator simply adjusts the speed of the motor by switching the motor to the variable-frequency power source until it is. (Although Hammond does not mention it, an erase head would probably be placed so that the signals could be erased after unsuccessful tries. Naturally, the speed of the disc and the frequency of the signal have been calculated in advance.

Any incorrectness manifests itself as a single point of the disc where the phase of the monitored signal suddenly jumps – that is, the reproduced wave, if sine, might suddenly for one cycle take on an appearance like that shown in Fig. 21-10, and a portion of a cycle is skipped. In that case the signal contains a large component of a frequency equal to the rotational speed of the disc.

The invention has a way to check that. When the switch S_1 is closed, the monitored signal is sent through a filter which selects one harmonic of the recorded sound (actually, of course, it is not a

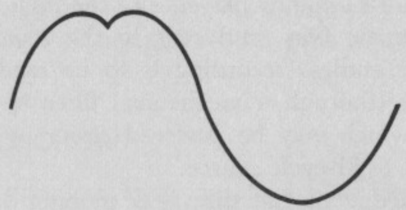


Fig. 21-10. Waveform appearance indicating nonre-entrancy.

sine wave) and suppresses all others. This is followed by an amplifier and a tuned detector similar to that in any broadcast receiver. The frequency of the sound is, naturally, much higher than that caused by the rotational speed, so the selected harmonic is modulated by the once-per-revolution frequency. The detector demodulates the harmonic frequency, leaving only the rotation-frequency component, if any, and passes the latter on to an indicator. The finished record can then be judged by the indicator reading, optimum condition being no reading at all.

Chapter 22

Tone Coloring, Amplification, and Control

ONE of the very important features of any musical instrument is its tone color or timbre. Ordinary acoustic instruments may have only one principal timbre, which makes them easily recognizable. Examples are the flute, with its rather smooth, rounded tone; the oboe, with its nasal quality; and so on. But it must be recognized that just about all acoustic instruments can be varied in tone quality to some extent, including those mentioned. They vary normally to some extent between upper and lower pitch registers, and even at a given pitch a skilful player varies the quality to give expression to the music.

Some instruments have more variation than others. One of the most expressive from this standpoint is the violin, which is an excellent reason for the fact that it stands head and shoulders above almost all others as a solo instrument.

An electronic musical instrument must have versatility to justify its existence; otherwise it becomes just a collection of pitch sources and loses its musicality. The Theremin did not have this versatility, which explains why it died. Organs especially must have a wide variety of tone colors at the player's command, whether or not the organ is supposed to be imitative of pipe organs or is designed to present new and different effects.

The principles of tone color were discussed in Chapter 4 and the explanations of today's instruments in succeeding chapters have covered most of the possible ways of varying tone, leaving little to say on the subject in this final chapter. While the mechanics (or electronics) involved can vary according to the designer's ingenuity, there are only a few basic methods of tone coloring:

1. Harmonic synthesis. Tone generators provide fundamentals and harmonics separately and controls allow mixing in any desired

proportions. The Hammond Organ is the outstanding example. The Wurlitzer 44 is another.

2. Generation of tone containing abundant harmonic content and subsequent elimination of harmonics not desired. The outstanding example is the Baldwin Organ. Most others also use the same system, with less fine control than the Baldwin obtains with the sawtooth waveforms.

3. Generation of two types of tones – with and without harmonics – followed by mixing and filtering in various proportions and manners. The example here is the Connsonata.

4. Direct generation of tones with the desired waveforms, by photoelectric and electrostatic means, and with magnetic recordings. The prime example of this system with a photoelectric approach was the Baldwin photo organ. The Compton Electrone is a well established electrostatic instrument which generates preformed tones to some extent, relying on mixing them in various proportions for further varieties. Such devices as electronic pianos and amplified guitars fall under this heading, too, since the preformed tones have the qualities of the piano and guitar, but they are not really electronic instruments and are not covered in this book. Electronic carillons are subject to the same comment.

There may be other methods yet to come, for electronic music is gaining rapidly in the number of people devoting time and interest to it, but for the present these four choices beckon.

One aid can be given to designers interested in method No. 1. It is the Harmonic Table shown here. The information for compiling it is taken from the scheme of the Hammond Organ and it shows the frequencies which must be keyed and controlled to give the correct harmonics for each fundamental.

The first column gives the 61 keys of a 5-octave manual. The nine following columns show what frequencies are used for each of the harmonics and subharmonics in the Hammond drawbar system, with the ordinal number of the drawbar (DB) controlling it. The frequencies are denoted by numbers from 4 to 94. Those up to 88 denote the numbered frequencies in the Tempered-Scale Frequency chart of page 7. Above that the frequencies can be arrived at by multiplying the next lower-octave frequency by 2. The 7th harmonic is not used. All harmonics are taken from generators also used for fundamental tones (except the very highest ones) and there is no fundamental close enough to the genuine 7th harmonic to be useful. This scheme will be used for any harmonic-synthesis instrument.

In Chapter 4 we showed a few oscillograms of tones played on a Kilgen organ and made by Prof. Hugh Lineback of Oklahoma A & M

HARMONIC TABLE

Fund. Key	DB I Sub fund.	DB II Sub 3rd	DB III Fund.	DB IV 2nd H.	DB V 3rd H.	DB VI 4th H.	DB VII 5th H.	DB VIII 6th H.	DB IX 8th H.
C 16	4	23	16	28	35	40	44	47	52
C# 17	5	24	17	29	36	41	45	48	53
D 18	6	25	18	30	37	42	46	49	54
D# 19	7	26	19	31	38	43	47	50	55
E 20	8	27	20	32	39	44	48	51	56
F 21	9	28	21	33	40	45	49	52	57
F# 22	10	29	22	34	41	46	50	53	58
G 23	11	30	23	35	42	47	51	54	59
G# 24	12	31	24	36	43	48	52	55	60
A 25	13	32	25	37	44	49	53	56	61
A# 26	14	33	26	38	45	50	54	57	62
B 27	15	34	27	39	46	51	55	58	63
C 28	16	35	28	40	47	52	56	59	64
C# 29	17	36	29	41	48	53	57	60	65
D 30	18	37	30	42	49	54	58	61	66
D# 31	19	38	31	43	50	55	59	62	67
E 32	20	39	32	44	51	56	60	63	68
F 33	21	40	33	45	52	57	61	64	69
F# 34	22	41	34	46	53	58	62	65	70
G 35	23	42	35	47	54	59	63	66	71
G# 36	24	43	36	48	55	60	64	67	72
A 37	25	44	37	49	56	61	65	68	73
A# 38	26	45	38	50	57	62	66	69	74
B 39	27	46	39	51	58	63	67	70	75
C 40	28	47	40	52	59	64	68	71	76
C# 41	29	48	41	53	60	65	69	72	77
D 42	30	49	42	54	61	66	70	73	78
D# 43	31	50	43	55	62	67	71	74	79
E 44	32	51	44	56	63	68	72	75	80
F 45	33	52	45	57	64	69	73	76	81
F# 46	34	53	46	58	65	70	74	77	82
G 47	35	54	47	59	66	71	75	78	83
G# 48	36	55	48	60	67	72	76	79	84
A 49	37	56	49	61	68	73	77	80	85
A# 50	38	57	50	62	69	74	78	81	86
B 51	39	58	51	63	70	75	79	82	87
C 52	40	59	52	64	71	76	80	83	88
C# 53	41	60	53	65	72	77	81	84	89
D 54	42	61	54	66	73	78	82	85	90
D# 55	43	62	55	67	74	79	83	86	91
E 56	44	63	56	68	75	80	84	87	92
F 57	45	64	57	69	76	81	85	88	93
F# 58	46	65	58	70	77	82	86	89	94
G 59	47	66	59	71	78	83	87	90	71
G# 60	48	67	60	72	79	84	88	91	72
A 61	49	68	61	73	80	85	89	92	73
A# 62	50	69	62	74	81	86	90	93	74
B 63	51	70	63	75	82	87	91	94	75
C 64	52	71	64	76	83	88	92	83	76
C# 65	53	72	65	77	84	89	93	84	77
D 66	54	73	66	78	85	90	94	85	78
D# 67	55	74	67	79	86	91	83	86	79
E 68	56	75	68	80	87	92	84	87	80
F 69	57	76	69	81	88	93	85	88	81
F# 70	58	77	70	82	89	94	86	89	82
G 71	59	78	71	83	90	83	87	90	83
G# 72	60	79	72	84	91	84	88	91	84
A 73	61	80	73	85	92	85	89	92	85
A# 74	62	81	74	86	93	86	90	93	86
B 75	63	82	75	87	94	87	91	94	87
C 76	64	83	76	88	83	88	92	83	88

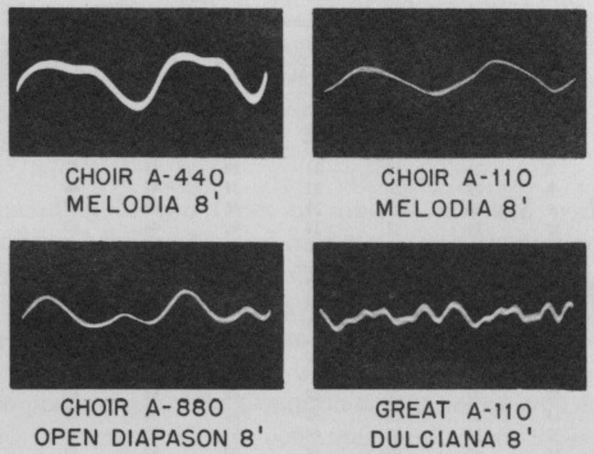


Fig. 22-1. Four pipe-organ tones obtained from a Kilgen organ.

Figure 22-1 shows four more such photos. Figure 22-2 shows six oscillograms, made by the author, of tones played on a small but well voiced organ in the Assembly Hall of Temple Emanu-El, New York City. Five are open diapason tones at different frequencies, showing how the waveform changes. The last is an oboe pattern at a low frequency. All these patterns may be used for reference, but it will be unlikely that an electronic organ will produce identical ones. Too many factors can enter, one of the most baffling being phase shift of harmonics. This makes no difference to the ear, according to best-informed workers, but two tones with the same harmonic structure

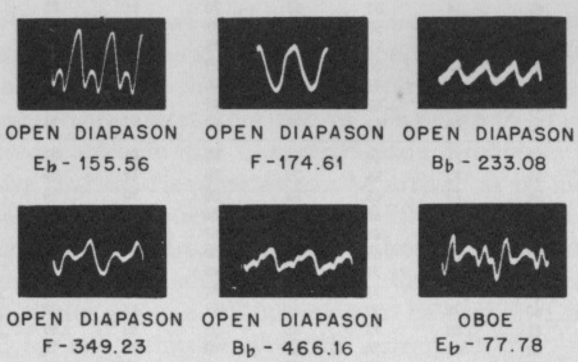


Fig. 22-2. Six oscillograms made from a small pipe organ.

and different harmonic phase relationships will produce absolutely different oscillograms.

AMPLIFICATION

There are very few words to say on amplifiers for electronic musical instruments, though the writer is often asked to advise constructors what kind of amplifier to use. The answer is that any good amplifier is suitable. The definition of "good" means, of course, an amplifier designed for high-quality music application and not a standard public-address amplifier.

The amplifier should satisfy the following conditions:

1. Its power output capability should be at least twice what would be required for a "high-fidelity" home music system if it is to be used in a living room. A 20-watt amplifier is about the minimum. If used in a church or auditorium power will have to be greater, but it is not normally economical to buy amplifiers of higher power rating than about 25 watts. An initial purchase of this character can be supplemented as necessary by additional amplifiers.

2. The simplest possible amplifier is desirable – without phono or microphone inputs. A single input for the organ is sufficient.

3. The power rating must apply at the lowest frequencies, not usually true of commercial jobs. This is an important reason for using higher power than considered necessary. Pedal tones are much louder and lower than most amplifiers of moderate power can handle without very bad distortion.

4. The fidelity of the amplifier should be excellent, especially respecting harmonics and intermodulation. Nothing giving greater intermodulation than about 1 or 2% should be accepted, and less than 1% is easily obtained. High-frequency response above about 8,000 cycles is not very important, but bass response down to 30 cycles is.

5. Transient response is of moderate importance only, as is damping factor, which depends on how much negative feedback exists. Too clean an organ tone is not desirable unless there is an organ chamber and the church or auditorium is big enough and has a long enough reverberation time to muddy up the tone somewhat.

6. Long-period reliability is very important in most cases. An electronic instrument suffers from one drawback not possessed by a pipe organ – it can quit entirely if a single part breaks down in the amplifier. Parts should be overrated at least 100% and oil-filled filter capacitors should replace electrolytics wherever possible.

Quite a number of amplifiers fill this bill, a situation which is true largely because of the big boom in high-quality audio reproduction in effect today. It is just as feasible to build an amplifier, a task which can certainly be performed by anyone capable of building the musical instrument in the first place. Circuits abound in such magazines as *Audio* and its earlier form, *Audio Engineering*, and in *Radio & Television News* and *Radio-Electronics*. The bibliography at the end of the book lists a few books on the subject. Some of the "Williamson" and other-type amplifier kits are also excellent.

CONTROL CIRCUITS

The controls and control circuits for instruments which are not organs depend entirely on the instrument and nothing general can be said about them except that they should be easily operable, conveniently located, and well marked. Control marking on either metal or finished wood can be done with decals such as the Tekni-Cals or Tekni-Labels obtainable in radio parts stores.

Organ controls should as much as possible resemble organ controls rather than electronic devices. The principal ones are:

1. Key and pedal-clavier switches
2. Stop switches
3. Coupler switches
4. Expression controls (swell pedals)
5. Crescendo pedal
6. Combination pistons
7. Brightness (on some electronics)
8. Vibrato or tremulant
9. Main-echo control.

Key and pedal-clavier switching is the most important problem. It is most easily solved when d.c. can be keyed rather than audio, since any clicks due to imperfect contacts can be filtered out without affecting audio. Few designs allow for this, however. There are not many choices for the contacts when it comes to keying, and these are discussed in Chapter 17.

Stop switching (or whatever switching is necessary for tone-color control) should preferably be handled with tablets of either the center-pivoted or end-pivoted kind or with draw knobs. Tablets, especially end-pivoted ones, are more convenient for the organist in quick registration changes, though at least one organist has said he feels draw knobs make on organ seem richer and more elaborate, a purely psychological effect that probably is not widespread. Either type of stop control can be found on pipe organs and if a second-hand console is used there is usually little choice. If the console is being built from scratch ordinary or special switches are usable, but the tab and knob ideas have resulted from long experience and probably few if any other types of actuators would be as easy to use.

The actual switching contacts for knobs or tabs are very easy to make. Probably the quickest solution is spring wire of phosphor bronze or nichrome, both obtainable from organ manufacturers and repairmen, as are the tabs themselves in many cases. Coupler switches and vibrato or trumulant controls fall under the same heading and all the above is applicable to them.

Expression controls or swell pedals are another matter. No matter what the schematic, this is one point where the constructor

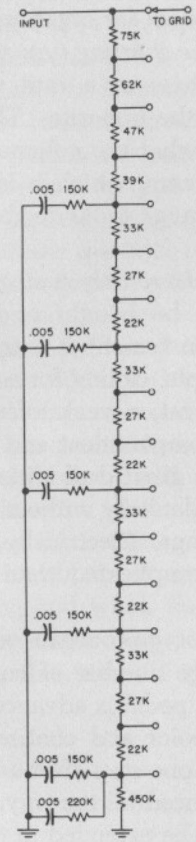


Fig. 22-3. Few organs use compensation as elaborate as this standard loudness control, but it does no harm if the builder has the necessary ambition.

must display some mechanical ingenuity. One solution involves potentiometers, racks, and pinions, as explained in Chapter 19. Other solutions might involve constructions like that of the Minshall in Chapter 7. In any case, the control and system of actuating must be super-reliable, for this kind of volume control is used thousands of times for every once the control on a radio or amplifier is twirled.

Some form of compensated volume control – what we nowadays call "loudness control" – is highly desirable rather than a straight volume control. The human ear becomes less sensitive to bass tones as the general volume level decreases. The result is that for a given stop combination, lowering the volume *via* the swell pedal causes the pedal foundation tones to disappear. To correct for this there should be a circuit which causes bass accentuation roughly proportional to volume decrease. The author did not provide this on his Electronorgan and regrets it.

Figure 22-3 is the circuit of the conventional loudness control used on some very fine amplifiers. It is ideal for organ use. The pedal is made to actuate a 17-point switch *of the shorting type* which reduces volume and simultaneously increases bass at a rate which almost exactly offsets the ear's tendency to do the opposite. This effectively keeps the ensemble constant no matter what the volume. The lowest switch step does not cut the volume to zero, which is ideal; no swell pedal should go all the way to zero. A range of 30 decibels or so is as much as there should be.

Most commercial electronic organs have only a single swell shoe for the entire organ. If a console can be bought or made with a separate shoe for each manual, the improvement is extremely useful. It means that in a registration, for example, calling for a solo voice on the swell with accompaniment on the great, a weak solo voice can be made to stand out against a strong accompaniment and the range of possible accompaniment registration is a great deal wider since registration may be used to determine tone color only without much regard to the characteristic volume of certain stops. Electrically, the addition of the extra swell shoes is no more complicated than an ordinary audio mixer circuit.

A crescendo pedal is found on most pipe organ consoles. It is usually a progressive switch with a large number of contacts. With crescendo out, no contacts make. As the pedal is advanced, each contact makes in turn with the common point and continues to make. This may be set up as desired to actuate one stop after another, building up to full organ with the pedal advanced all the way. If the leads going to the normal stop switches cannot be extended to the crescendo pedal in parallel because of interference and hum pickup, very likely small relays can be inserted with the coils operated by crescendo pedal and stop switches in parallel.

Combination actions are found on most organ consoles but in few electronic instruments. One of the difficulties is that electronic organ manufacturers dislike to put in the necessary pneumatic pistons with the required air compressor, and no one can blame them. For most constructors, either an air supply must be furnished and the pneumatic

pistons in the console used, or an elaborate mechano-electric system involving a motor and cams must be used. It was at first thought that simple solenoid actuators could replace the pistons, but it soon was evident that this was a lost hope. The solenoids act far too fast, with a clatter than can be heard a mile away. Any attempt to slow up their action – electrically or by airchecks, etc., – produces a horrible electromagnetic buzz which can be heard at about the same distance. Baldwin has solved the problem electromechanically in its Model 10 organ, but few builders will be able to do the same. Better to leave the combination action out of commission.

A brightness control is very useful and can be added to any electronic organ. It is nothing more nor less than a simple, old-fashioned tone control placed across some plate or grid in the amplifier system (within the console) with a knob on the panel operating a potentiometer or step-type control. It adds to the expressive capabilities of the instrument by making it possible to tone down or point up the sharp treble qualities of reeds and brasses. Make it by placing a variable resistance in series with a capacitor across the grid or plate. Experiment to find the best values.

Most organs are heard through a single group of speakers in a single location. Additional versatility without great complication is possible by placing a second power amplifier-speaker setup in a remote part of the auditorium and adding a switch on the console to select either or both at will. The remote sound source serves something of the function of the Echo Organ in a standard installation, a group of pipes distant from the main organ. Especially in a large, reverberant church, the tones coming from a distance have a lovely, ethereal effect; and when they mix with the main tones they are in various phase relationships, tending to add a chorus or ensemble effect not possible in most electronic instruments.

Appendix

Electronic Music Patents

Not the least among the values of the United States patent system is the knowledge it disseminates. It is one thing to read in a book the classic ways of doing things – designing electronic musical instruments in this case – and to know how the commercial versions work; it is quite something else again to be able to read through hundreds of patents on the subject and realize how many brains have worked over methods of doing things in a thousand different ways.

Many an aspiring inventor feels his goal has been reached if he can obtain a patent on his brainchild. Alas, the majority of patented inventions (including electronic music circuits) never see the commercial light of day. But the inventor has at least the satisfaction that he has contributed to the fund of human knowledge. His organ might have worked well but for a bug or two; there may have been no bugs but those of uneconomical manufacturing processes; or his ideas, though new, may have been so bizarre and unrealistic that they were, on the whole, impractical. Or he may simply have lacked the contacts and the salesmanship to sell his first-class invention to a manufacturer or the public.

Whatever the cause of his final economic failure, the inventor has given us, *via* the Patent Office, a permanent record of his thinking, his brainstorms, and his experiments. If he succeeded in commercializing his invention or if he was a paid employe whose inventions automatically belonged to his employer, his patent specification still has something to say that we cannot find in the actual marketed product it has fostered. The specification records his thinking, and the derivation of his ideas.

Patents serve the researcher-designer with ideas, schemes, germs of solutions, angles of approach. He may pick the brains of a hundred inventors for the kindling which his own match may set afire with a

new and brilliant design. Or if he is not an original designer by skill and temperament, he may use the same hundred inventors' ideas to piece together his own musical instrument with individual circuits lifted bodily from the patents.

Because of the tremendous value of the patent literature the writer has compiled a large file of patents on electronic music. To make the most important and useful of these patents available to readers, this appendix lists them.

The patents are arranged in groups, covering the important aspects of electronic music. Almost every patent, however, treats of more than the subject covered by the main heading under which it is listed. The patents are listed by year, to give the reader an idea of their "freshness." In each category there are listed first what the writer considers the most interesting and useful, with a brief descriptive comment on each; following that, others – less important – are listed by number only. To conserve space we have not followed standard bibliographic procedure, giving only the information on each patent likely to convey a message to the reader.

A copy of any patent may be obtained for 25¢ from The Commissioner of Patents, Washington 25, D. C. There is no discount for quantity. The large public libraries also contain excellent patent files, as well as copies of the Patent Office Gazette, issued weekly and containing a summary of all patents issued.

TONE GENERATORS

1931

1,817,704. Phonograph records contain tones, with pickups keyed to reproduce them.

1,823,716. Charles J. Young, assigned to General Electric. Polyphonic instrument using heterodyne principle.

1,823,724. Wendell L. Carlson, assigned to GE. Heterodyne instrument with r.f. harmonic filters to select tone structure.

1,837,144. French inventor. Heterodyne instrument capable of plucked-string effects.

1932

1,865,428. Same inventor. Damped oscillations started by capacitor discharge give piano-harp effect.

1933

1,937,021. John Hays Hammond, a prolific inventor, not connected with Hammond Organ. Photoelectric instrument.

1,937,389. Nicholas Langer, well known primary worker with relaxation-oscillator organs. A gas-tube design.

1,940,093. Bell Labs. Heterodyne instrument with pitch controlled by a trolley-car motorman's form of handle.

*Grateful thanks go to the Editors of *Radio-Electronics* for allowing the writer to filch freely from their own impressive file of early patents on this subject.

1935

1,993,890. Langer. Monophonic neon-lamp instrument.

2,014,741. Photoelectric organ, timbres generated directly.

2,017,542. Langer. Synchronizing octave strings of neon oscillators from a master oscillator.

1936

2,035,238. Langer. Plucked-string effect with neon lamps.

2,039,651. Langer. Improvement on 1,993,890, allows fine tuning of individual notes.

2,044,360. Langer. Another method of syncing neons.

2,046,463. Dr. Winston Kock, assigned to Baldwin, whose organ he mainly designed. Various neon oscillators including inductors.

1937

2,070,344. Harry F. Waters. Neon instrument with continuous tuning device mounted on replica of a violin neck.

1938

2,128,367. Kock. Thyratron octave strings.

1939

2,148,166. Tones generated by cathode-ray tube.

2,171,936. More of the same by the same inventor.

1940

2,185,635. Kock. Improvement on 2,128,367.

2,221,097. James A. Koehl, v.p. of Central Commercial Industries, maker of Lowrey Organo. Photo organ using film strips which start moving on keying, then rewind when key is released.

1942

2,276,389. Laurens Hammond. This *is* the organ man. Octave divider strings using hard (vacuum) tubes as relaxation oscillators.

2,293,499. Western Electric. Instrument using just rather than tempered scale but allowing playing in any key. Interesting discussion of temperament.

1949

2,472,595. J. T. Kunz, Schulmerich (important electronic chime manufacturer). Capacitive pickups for bar chimes.

2,473,897. British inventor. Photo organ.

2,474,847. E. M. Jones, Baldwin. Photo organ with light-interrupter disc.

2,476,607. E. L. Kent, Conn. Connsonata oscillator.

2,480,131. Laurens Hammond. Struck coil-springs with magnetic pickups.

2,481,608. Spencer McKellip. Oscillators with two tone qualities.

2,484,914. Photo organ.

2,485,829. Dutch inventors. Photo organ with stationary short film wave tracks scanned by light from cathode-ray tube.

2,486,039. Langer. Aperiodic flip-flop dividers. See Chap. 20.

2,489,857. Singers' voices recorded optically and pitches played back selectively.

2,492,919. Cornell-Dubilier. Vibrating reeds, magnetic pickups.

2,494,943. British inventor. Photo organ.

1950

2,498,367. Laurens Hammond, and John M. Hanert, Hammond's Research Chief. Chorus generator for Hammond Organ.

2,500,820. Hanert. Tube generators with chorus effect.

2,509,923. Hanert. Chorus and vibrato effect for generators initially incapable of varying frequency.

2,513,109. Photo organ.

2,522,923. L. E. A. Bourn of British Compton. Electrostatic generator.

2,533,461. Magnetic tone recordings.

1951

2,539,130. Magnetic tone recordings.

2,539,826. Frequency-doubler strings.

2,542,611. V. I. Zuck, Wurlitzer. Capacitive reed pickups.

2,559,276. Photo organ.

2,561,349. Tones from acoustic bass instrument picked up and brought down an octave lower.

2,568,862. Constant Martin, France. Struck strings with magnetic pickups.

2,573,975. Rotating magnetic tone generators.

2,574,577. RCA. Electronic generation of bell tones.

2,576,760. Baldwin. Means for stabilizing speed of rotating photoelectric generator.

1952

2,588,680. Photo organ with single high-speed rotating disc.

2,601,265. Tones generated with cathode-ray tube.

Miscellaneous

1,782,542	2,569,521	2,540,285
1,847,119	2,576,759	2,544,722
2,340,001	1,832,402	2,570,178
2,500,947	2,036,691	2,581,653
2,535,323	2,480,945	2,636,989
2,543,629	2,508,514	

KEYING ENVELOPE CONTROL

1936

2,043,828. Coupleux, France. Delay by momentarily reducing heater voltage in filament-type tube.

1939

2,161,706. Laurens Hammond. Delay suitable for but not used on Hammond Organ.

2,173,888. Gilbert Smiley, Hammond. Grid-bias keying.

1940

2,215,709. Benjamin F. Meissner. Key contacts capacitive so tone transmission varies with key position.

2,216,513. Laurens Hammond. Slow decay with capacitors.

2,224,729. Laurens Hammond. Improved grid-bias keying.

1941

2,266,030. Laurens Hammond. Envelopes affected by rate at which key is pressed.

1942

2,287,105. Walter F. Kannenberg, Bell Labs. Envelope control by d.c. applied to rectifier bridge. See 2,025,158 for more complete theory of rectifier bridge.

2,296,125. Pressure-sensitive keys.

1948

2,452,307. Koehl, Central Commercial. Key contacts unshort several resistors for step-type tone rise.

1949

2,482,548. Dutch inventor. Struck-string envelope keying.

2,483,823. Thomas J. George. Delay circuit with diodes; produces string and flute tones from sine generator.

1950

2,497,331. Several busses, swept by each key contact, give step-type tone onset and decay.

2,500,821. Hanert, Hammond. Pedal generators with slow decay.

1951

2,543,628. Hanert, Hammond. Envelope control for monophonic instrument; legato playing still retains separate envelope for each note.

2,575,230. Baldwin. Resistive key contacts (not used in present Baldwins).

2,576,758. E. M. Jones, Baldwin. Envelope control for photoelectric instrument; shutters.

2,577,753. Hanert, Hammond. Sustaining means for pedal tones.

SPECIAL PURPOSE KEYING SCHEMES

1931

1,824,402. Maurice Martenot, France. Pitch determined by amount of wire unwound from spool.

1936

2,045,172. Baldwin. Thirteen oscillators for entire organ.

2,048,610. Kock, Baldwin. Highest note of any chord struck sounds loudest.

1940

2,203,432. Thomas J. George, Hammond. Use of six oscillator strings on premise adjacent notes never used together. See Chapter 12.

1941

2,241,363. Laurens Hammond. Single pedal provided, sounds note determined by lowest note being played on manual.

1943

2,323,242. Bell Labs. All notes of instrument produced by harmonics filtered from six generators tuned below lowest octave.

1949

2,484,930. Black keys eliminated. Selector set at key of composition and white keys automatically provide correct scale.

2,492,320. Bendix Aviation. Practice instrument, with many keyboards connected to single generator system; headphones.

1951

2,549,697. Bendix Aviation. Key switches operated by piano-like hammer action, remain closed for time depending on player's stroke.

2,577,493. Dutch inventor. Fewer generators than keys, on premise that only limited number of keys can be struck at once.

1952

2,581,680. Maurice Martenot, France. Capacitor varied by movable tape, as used in the Ondes Martenot. See Chapter 15.

TONE COLOR

1925

1,530,498. Harmonic outputs of oscillators controlled.

1932

1,877,317. Westinghouse. Each note has fundamental oscillator and also harmonic oscillators as required.

1933

1,933,299. German inventor. Modifying tone qualities of standard instruments.

1934

1,947,020. Richard H. Ranger. Generators provide all harmonics. Filters for each note subtract those not wanted.

1936

2,035,836. Richard H. Ranger. Tones distorted by circuit to provide complex structure.

2,039,201. Friedrich Trautwein, Germany. Generation of tone accompanied by shock-excitation formants.

1938

2,139,023. Kock, Baldwin. Shock-excitation formants furnished by neons just short of oscillation.

1940

2,227,100. Central Commercial. Harmonic synthesis.

1941

2,251,051. Laurens Hammond. Several generators for each note, each with different tone quality.

1949

2,474,380. Tone-shaping with cathode-ray tube.

2,486,208. Addition of transients – breath effects, etc.

1950

2,506,723. Merwin J. Larson, Stromberg-Carlson. Sine waves distorted for tone variations.

2,533,821. Langer, Central Commercial. Conversion of square to sawtooth waves.

TREMOLO, VIBRATO, ETC.

1932

1,853,630. Maurice Martenot, France. Whole keyboard is slightly movable; vibration of hand on key produces vibrato.

1936

2,040,439. Langer. Neon-lamp vibrato circuit.

1949

2,466,306. Tremolo by varying speaker field current.

2,485,538. Paul H. Rowe, Mass-Rowe. Tremolo by oscillator varying grid bias of push-pull amplifier output stage.

2,489,653. Donald J. Leslie. Rotating vanes in front of speaker.

1950

2,503,352. Electronic variation of tube gain.

2,534,342. Same idea as 2,485,538.

1951

2,542,065. Baldwin. Separate tremolo for each manual.

2,565,033. Played back film track has vibrato added by cyclically moving scanning light or varying speed of film.

2,580,217. French inventor. Adding "movement" to tones to simulate random frequency variations of organ pipes.

1952

2,600,870. RCA. Synthetic reverberation with stretched string and magnetic transducers.

COMPLETE INSTRUMENTS

Photoelectric

1,980,292	2,169,842

Acoustic with Pickups

1,705,395		1,901,985
2,089,204		2,300,609
	2,486,545	

Electromagnetic

380,035 (1897)	1,956,350

Electrostatic

Re. 21,554	Re. 22,321

Magnetic Recordings

2,549,145

Electron-tube Designs

1,190,332	2,276,390	2,497,661
1,661,058	2,294,178	2,540,727
1,911,309	2,295,524	2,544,466
Re. 20,831	2,301,871	2,545,665
2,211,540	2,310,429	2,562,908
2,233,258	2,478,867	2,563,477
2,245,337	2,489,497	2,568,644
2,254,284		

MISCELLANEOUS PATENTS

2,475,742. John Hays Hammond. Production of reentrant magnetic recordings. See Chapter 21.

2,491,189 and 2,491,190. Thomas H. Long, Conn. Harmonic wave analyzer.

2,538,184. Wurlitzer. Electronic pickup for piano tuning; eliminates need for rubber wedges.

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Index