Chapter 2

Special Signal Circuits

The basic function of the guitar amplifier is to increase the sound of the instrument so it can be heard under many different conditions—from small practice rooms to large concert halls. Yet electronics offers many possibilities for adding variety to the basic guitar sound. The location of the pickup influences the signal; some guitars use as many as three in different positions for various effects. Tone control circuits in the amplifier itself can make the sound brilliant or mellow. Other special circuits—echo (reverberation) and tremolo—can be added by separate units or can be built into the amplifier.

THE PICKUPS—TYPES AND CONSTRUCTION

At the input of the amplifier is the pickup itself—the device that converts the motion of the guitar strings into electrical signals. The first type used was a contact microphone constructed as shown in Fig. 2-1. The correct technical term for any of these things would be "electromechanical transducer," but they are called pickups for convenience.

The contact mike is just a microphone element of any kind—crystal, dynamic, etc. Instead of having a diaphragm like the voice-operated types, it has a coupling of some kind, so it can pick up only the vibration of the surface with which it is in contact (theoretically!). Actually, due to the high gain that is required, this kind of mike picks up many sounds very well—talking close by, the rubbing of clothing on the guitar, any jar that is given the instrument, and so on. Now the contact mike has been replaced almost entirely by the magnetic pickup. This responds only to a motion of the metal strings through the magnetic field of the pickup coil and has no microphonic effects that produce undesirable sounds.

Fig. 2-2 shows how this works. Only one coil is shown, although there is normally one for each string; they all work in the same way. All strings are made of metal; single strands are used for the higher pitch, and wrapped strings for the bass. The pickup consists of a small permanent magnet wound with a great many turns of a very fine wire. The metal string vibrating in the field of the magnet causes this field to move. Since the coil of wire (pickup coil) lies in the moving magnetic field, a small electric current is generated in the coil.

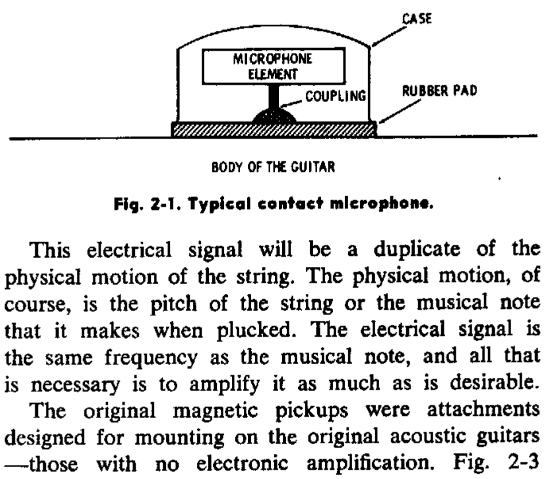


Fig. 2-1. Typical contact microphone.

This electrical signal will be a duplicate of the physical motion of the string. The physical motion, of course, is the pitch of the string or the musical note that it makes when plucked. The electrical signal is the same frequency as the musical note, and all that is necessary is to amplify it as much as is desirable.

The original magnetic pickups were attachments designed for mounting on the original acoustic guitars —those with no electronic amplification. Fig. 2-3 shows a typical mounting for one of these. A single, long, flat coil in a metal shielding case was used, and

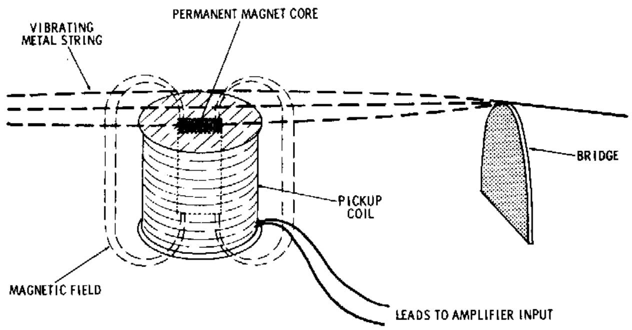


Fig. 2-2. Operation of a magnetic pickup.

a clamp was provided to hold it tightly in place under the strings. Some models had volume controls in the same assembly.

In specially built electric guitars the pickup coils, volume and tone controls, etc., are installed in cutouts in the body of the instrument where they are covered with chromed metal plates or plastic covers. There are other controls and special effects used on the custom models, which are discussed later.

All of the pickups use the same basic circuit shown in Fig. 2-4. If individual coils are used for each string, they are connected in series or parallel; the whole pickup unit is connected across a volume control of from 0.5 to 1.0 megohm or more. The simple high-cut tone control shown may be mounted on the guitar itself. A shielded coaxial cable is always used to connect the pickup to the amplifier; this eliminates hum, electrical noise, etc., from the signal. If the interconnecting cable is fairly long, say more than about 10 to 15 feet, a low-capacity cable should be used. Very small cable has a high shunt capacity and will cut down on the transmission of high frequencies. Most standard microphone cable is fairly low capacity, and up to 50 feet can be used without trouble.

Standard phone plugs and jacks are the type of connectors most commonly used. The connections on these must be kept clean and tight to get rid of any noise and hum. Full details on how to handle these plugs and make repairs to the mike cables will be given in the section on servicing.

TONE CONTROLS

All except the very smallest amplifiers have some sort of tone-control circuit. These do not change the fundamental tone of the instrument—that is, the pitch or frequency of the string. However, they can change the characteristics of the amplifier by increasing or de-

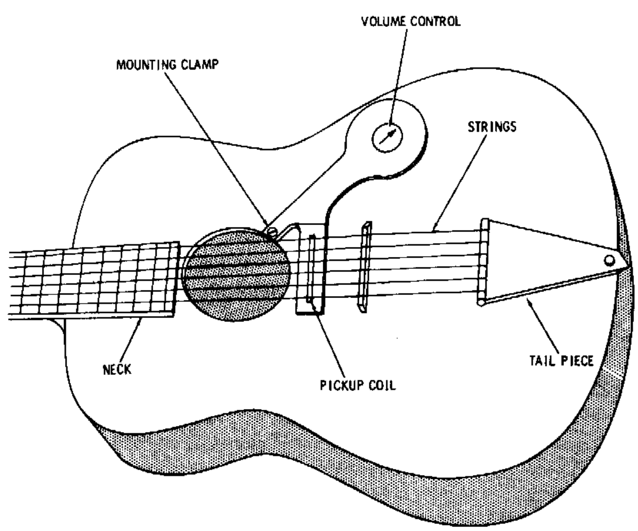


Fig. 2-3. Acoustic guitar with pickup-coil attachment.

(A) Single

(B) Double.

(C) Triple.

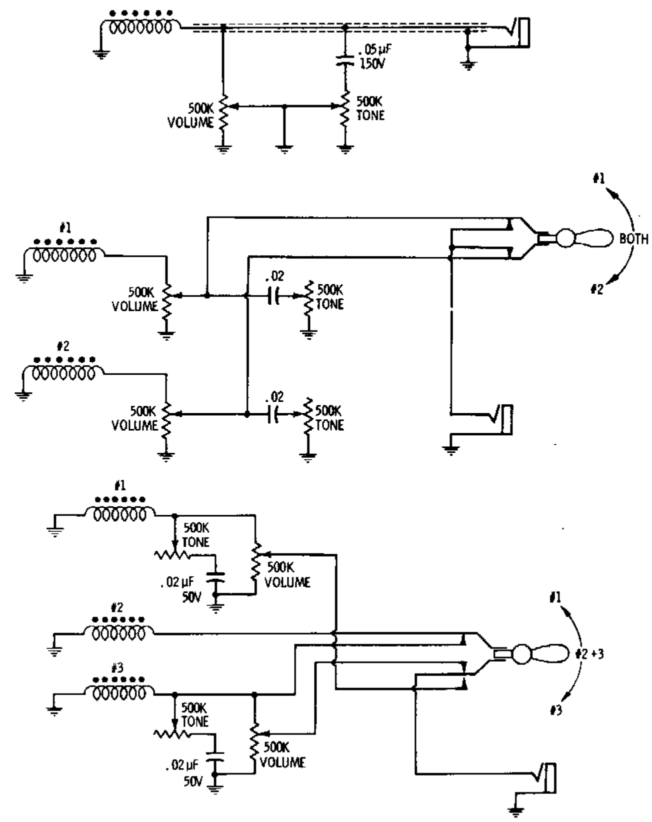


Fig. 2-4. Pickup circuits.

creasing the amplification of high or low notes, depending on the control setting.

The simplest tone control is what is known as a high-cut type, as used in Fig. 2-4. Basically, it looks like Fig. 2-5. A capacitor has one valuable characteristic: it shows a much lower impedance to high-frequency signals than to low frequencies. If it is placed across a signal circuit at any point in the voltage amplifier, the high-frequency response will be cut. The capacitor will short out the highs by giving them a low-impedance path to ground. This is called a high-cut circuit, because it cuts down on the high frequencies and gives the amplifier a lower tone. This type of tone control does not increase the bass tones; it simply takes out some of the highs and makes the bass sound bigger by comparison.

In order to vary the amount of high cut, a variable resistor is added in series with the capacitor (Fig. 2-5). The size of the resistor and the capacitor will determine how much of the highs are taken out or left in. If the control is set to its lowest resistance position, all of the capacitance is across the circuit, and the lowest tones will be prominent. If the variable resistor is set to the maximum resistance position, in effect the capacitor is taken out of the circuit, and the high tones are present, because the path to ground is now a high resistance. After the tone control, the signal goes on to the next stage of the amplifier.

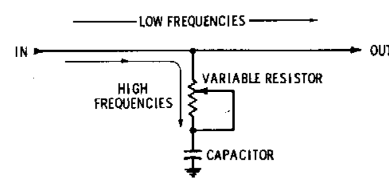


Fig. 2-5. High-cut tone-control circuit.

FEEDBACK TONE CONTROLS

In the more expensive instruments, a different type of tone control is used, involving what is called negative feedback. This either boosts or cuts the bass and treble frequencies. Since it is a pretty complicated circuit, the details of its design will not be discussed. However, it can be found in any good electronics textbook under bass boost and treble boost tone-control circuits.

A feedback tone control can increase the amount of bass frequencies in a tone (bass boost) or cut them; the same thing can be done with the treble frequencies. Neither of these actions will affect the other. The simple high-cut tone control, of course, affects all frequencies somewhat. The basis of operation is the use of selective feedback—a network of resistors and capacitors that feeds back a part of the output signal into the input. By changing the amount, and in some cases the phase, the input signal can either be built up or lowered at selected frequencies. Inverse feedback always lowers the gain of an amplifier but improves its fidelity. If a large amount of bass frequencies are fed back, for example, the gain of the amplifier for bass notes is reduced. By cutting down on the amount of signal fed back, the gain is returned to normal (bass boost). By changing the frequency of the feedback circuits through the use of different resistor and capacitor values, the same action can be obtained for treble tones.

TREMOLO

Tone controls have no effect on the action of the amplifier; they simply change its frequency response or volume a little. The special effects result in a whole new character to the sounds. There are three of these in common use: tremolo (a variation in volume level), vibrato (a variation in frequency or pitch), and echo or reverberation.

Tremolo is basically pretty simple. If the bias of an amplifier stage is raised or lowered, the volume changes. Almost any subaudible frequency can be used to give a pleasing "vibration" effect to the musical tone. This causes the two effects to be confused; unless you have a very keen ear for musical notes, you can easily get tremolo and vibrato mixed up since they do sound a lot alike.

A tremolo effect can be created by varying the bias voltage on any amplifier stage at any desired frequency. The typical circuit will use frequencies from about 1 Hz up to 50 or 60 Hz. The average amplifier has a tremolo rate of about 10 to 15 Hz. Fig. 2-6 shows how this circuit works. Typical waveforms are included. Note that there are two controls shown. In most amplifiers these are marked STRENGTH and SPEED, meaning amplitude and frequency.

The heart of the tremolo circuit is a very low frequency oscillator, the output of which can be varied in frequency (by the SPEED control) to give the rate of tremolo wanted. The STRENGTH (amplitude) control varies the voltage of the output. At its low end there will be a barely perceptible quaver in the note; at the other extreme, the tremolo will consist of variations from fairly high to fairly low volume.

The output of the low-frequency oscillator might be called a slowly varying dc voltage for simplicity. This is fed, through isolating resistors, into the bottom of the grid circuit of the desired amplifier stage. There it affects the grid-bias voltage by adding and subtracting to the bias already present in normal operation. As a consequence, it changes the volume at the output of the stage. The tremolo effect, once added to the signal, goes on through all of the following amplifier stages.

Examine an actual tremolo circuit used in a typical commercial amplifier (Fig. 2-7). The 6SQ7 tube is a phase-shift oscillator; notice the network of capacitors and resistors connected between the plate and the grid of the tube. This takes the signal from the plate, delays it in phase as it passes through, and feeds it to the grid in just the right phase to cause oscillation. The frequency of oscillation in these circuits depends on the values of the resistors and capacitors; it can be controlled by varying either one of them.

Since variable capacitors are big, a variable resistor is used—in this case the 500k control shown con-

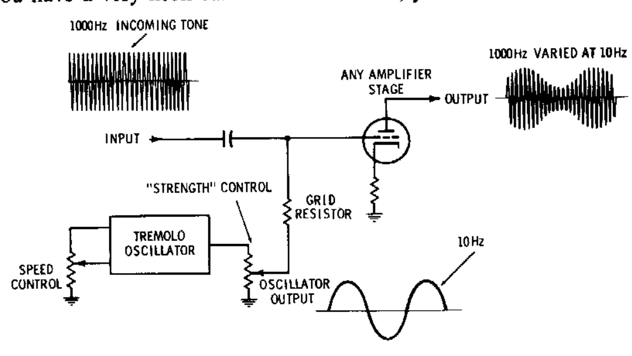


Fig. 2-6. Tremolo circuit.

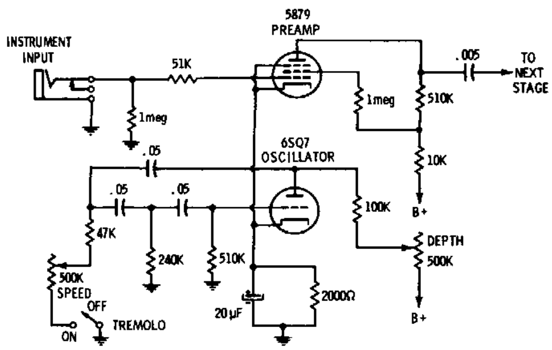


Fig. 2-7. Phase-shift oscillator tremolo circuit.

nected to the tremolo on-off switch. The output of this circuit will be very low frequency oscillation.

The bias variation is applied to the 5879 preamplifier stage. When the two cathodes are coupled together, the bias voltages on both stages vary simultaneously. This changes the gain and consequently the amplitude of the output of the preamp tube. Any instrument connected to the input of the 5879 preamp tube will have a tremolo in its output that can be varied by the setting of the tremolo controls.

There are two controls: frequency, the variable resistor in the oscillator circuit itself, and amplitude, the variable resistor in the B+ supply circuit to the oscillator plate. You will find these called STRENGTH (amplitude) and SPEED (frequency) in most amplifiers for simplicity. In one make they are labelled DEPTH (amplitude) and RATE (frequency). No matter what names are used, they do the same thing.

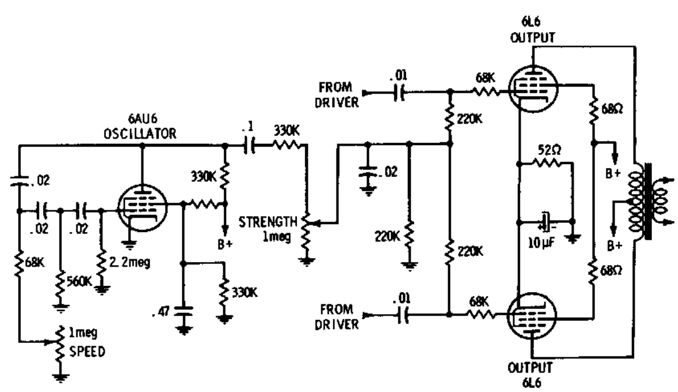
The amplitude control works by increasing or reducing the plate voltage of the oscillator. This does not have much effect on the frequency of a phase-shift oscillator, which is pretty stable, but it will affect the amplitude or strength of the oscillations. A high amplitude oscillation will cause a greater change in the cathode voltage and thus will give a much more noticeable tremolo in the preamp output. This is why it is called depth in one make: it varies the depth of the tremolo imposed on the original tone.

You will find several different versions of the amplitude control, but the effect is always the same. For instance, the amplifier shown in Fig. 2-8 is a high-powered unit with two 6L6s in the output, and feeds the tremolo bias directly to the output tubes. Here a 6AU6 tube is used as the tremolo oscillator (a phase-shift type), and the voltage variations are coupled into the grid return of the 6L6s through a 0.1-μF coupling capacitor from the 6AU6 plate and the strength control. In this case the strength control is a divider across the tremolo oscillator output.

The tremolo circuit in Fig. 2-9 is somewhat more involved than the previous ones. The phase-shift oscillator is like all the others although the actual controls are in a remote control assembly. However, the tremolo signal is applied to the amplifier in a very different manner. From the oscillator the tremolo signal goes to an inverter that puts out two tremolo signals—one at the plate and one at the cathode, each 180° out of phase with the other. The signal from the guitar is also split (by V4A) into two signals of opposite phase—one chiefly high frequencies and the other low frequencies. The corresponding tremolo signal is added to each guitar signal (in V5A and V5B), and then the outputs are combined. The purpose of all this circuitry is to eliminate any sound of the tremolo from the amplifier output when the amplifier is on but the guitar is not being played. This is accomplished by using the two out-of-phase tremolo signals; when they are combined without a guitar signal, they cancel out.

In the other circuits the tremolo voltage varies the bias; in Fig. 2-10 the ac signal from the tremolo is fed through a comparatively large capacitor to the screen grid of a pentode amplifier tube. The previous circuits

Fig. 2-8. Tremolo applied to grids of output tubes.



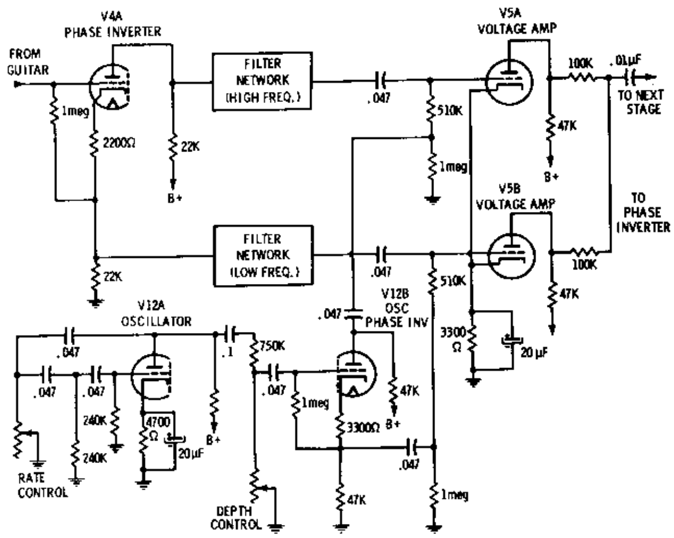


Fig. 2-9. Tremolo applied as two out-of-phase signals.

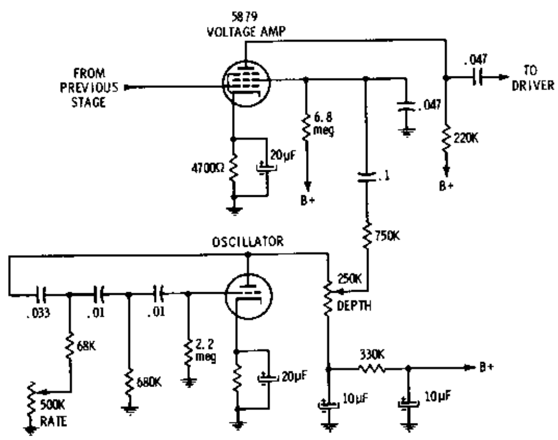


Fig. 2-10. Tremolo signal coupled to screen grid.

have been directly coupled—all resistors. This one is coupled through a capacitor. The action remains the same. There is a comparatively small screen-grid voltage applied to this tube. Look at the size of the screen-grid dropping resistor (6.8 megohms). As a result, the tube has a screen-grid voltage of about 20 volts. If an ac voltage varying about 10 volts peak to peak is fed through the big capacitor, the screen-grid voltage is actually changing from 20 to 30 and back to 10 volts. The actual voltage on the screen at any given instant will be the sum of the residual dc voltage through the B+ supply resistor plus the instantaneous value of the tremolo voltage coupled through the blocking capacitor. Since the amplification factor of a tube can also be changed by the screen-grid voltage, this adds the tremolo effect to the signal going through the tube at that time. (Very old radio sets used this circuit for volume controls; they changed the screen-grid voltage on the rf amplifier stages.)

All of the circuits shown in the preceding figures can be used in transistorized amplifiers. There is no difference in the basic operation. This also applies to any other special-effects circuitry.

The Light Dependent Resistor

A novel method of coupling the tremolo oscillation voltage into the voltage amplifier circuits involves the use of a light dependent resistor. Fig. 2-11 shows this circuit. A light dependent resistor (LDR for short)

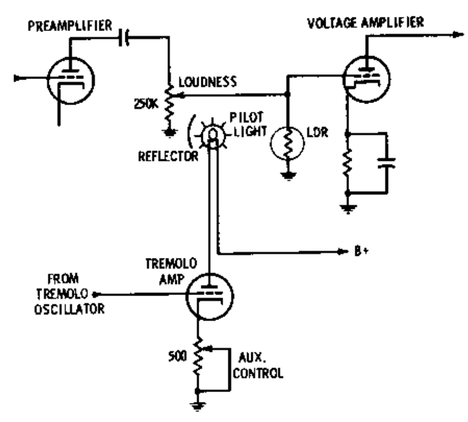


Fig. 2-11. Light dependent resistor (LDR) used to couple tremolo to signal channel.

is a photoelectric cell in which the resistance of the unit depends on how much light is falling on it. A variable resistance can be used as a volume or loudness control; there is one just ahead of this. By making the LDR vary in size, there is in effect another volume control in the same circuit that will raise and lower the volume just as the loudness control does. However, for this application the variations must be in step with the vibrations (slow voltage changes) of the tremolo circuit—from about 3 to 50 Hz. So, instead of coupling the tremolo voltage directly into the bias circuit, an LDR is used. The tremolo oscillator has no direct connection to the sound circuits; its plate circuit contains a small pilot light. As the signal changes, this light gets brighter and dimmer. The light is focused on the LDR, making its resistance change. This varies the volume in the signal channel, adding tremolo to the signal.

One reason for the use of the LDR is to isolate the tremolo oscillator from the signal circuits. Unless some precautions are taken, the low-frequency oscillation will go through the amplifier when the guitar is not being played and give an unpleasant sound in the output. With this system no oscillator signal is introduced into the guitar signal circuits; there is tremolo in the output only when there is a musical signal going through the amplifier.

Usually the LDR and the lamp are contained in a single unit which is drawn on a schematic as shown in Fig. 2-12. Either incandescent or neon lamps may be incorporated into the units, depending on the voltage used to excite them.

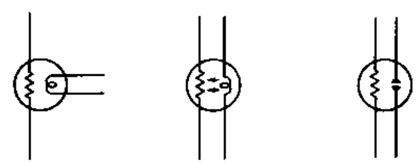


Fig. 2-12. Schematic representation of light dependent resistors and lamps.

VIBRATO

A true vibrato is very difficult to get, electronically. It would involve a very complicated circuit using differential phase-shifting that would be very hard to make adjustable. Most manufacturers use a mechanical lever action on the tailpiece of the guitar; the player strikes a chord, then moves a long handle back and forth. This changes the tension on all strings at the same time, alternately raising and lowering the pitch of the chord. Fig. 2-13 shows two vibrato units, one functioning as a tailpiece only and the other combining an adjustable bridge with the tailpiece.

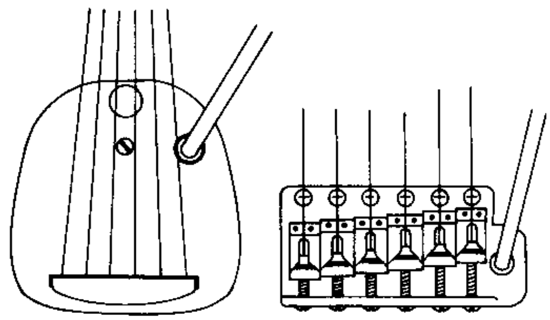


Fig. 2-13. Mechanical vibrato units.

NOTE: You will find the two terms—tremolo and vibrato—used interchangeably, even in some of the catalogs. By a strict musical definition, they are not interchangeable. If the circuit varies the volume of the tone, it will be a tremolo no matter what is on the control knob. If the pitch of the tone varies, then it is a vibrato. Frankly, both effects sound exactly alike to the untrained musical ear, so it probably doesn't make a lot of difference which term is used.

ECHO OR REVERBERATION

Echo or reverberation is a very popular effect of late. It is used in many popular recordings, musical arrangements, and so on. An echo effect is obtained by taking off a part of the signal, delaying it slightly, and then adding it to the original signal. The original note sounds, and a fraction of a second later there is an echo. This is done by changing phases in the signals.

Phase means a time relationship between any two similar electrical signals; it is generally used with reference to ac signals only. If one signal starts out and is followed a fraction of a second later by another signal just like it, the second signal is lagging in phase with reference to the first signal. This, of course, is the same effect you get if you yell into a canyon. First you hear your own voice come back from a nearby cliff, and then, a wee bit later, it comes from a more distant cliff. It is the same thing: echoes.

If the echoes are so close together that the listener can't separate them, the sound is called reverberation. The only difference between echoes and reverberation is the length of time or amount of delay between the signals.

To get an echo effect electronically, only a part (approximately half) of the signal is taken off. The original goes on through the amplifier; the part that is taken off goes through a special circuit which causes it to lose time—it is delayed. The amount of delay is regulated by controls in the echo circuitry; also the amplitude or the strength of the delayed signal can be varied. When the original and the delayed signals are combined in a second stage, there is an echo effect; the original signal is heard, and, a small fraction of a second later, the delayed or echo signal is heard. Fig. 2-14 shows a block diagram.

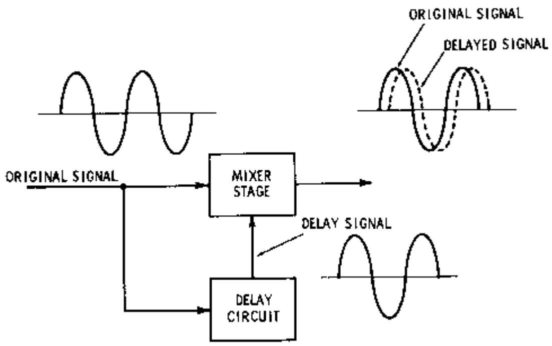


Fig. 2-14. Using signal delay to produce echoes.

Most circuits use a mechanical delay line for echo circuits. This looks like a pair of old-fashioned spiral screen-door springs in a little box. Actually, they are very carefully designed springs having very precise travel times, as will be seen. A speaker is mounted on one end of the two springs. This is not a true speaker, but a speaker-motor that converts the electrical signal into mechanical vibrations, just as a speaker does. This puts the sound signal onto the ends of the two springs. It travels the length of the two springs in very slightly different times due to the way the springs are wound. A typical time used by one major manufacturer is 29 milliseconds for one spring and 37 milliseconds for the other (1 millisecond equals .001 second). The delay effect is accomplished in the springs themselves. At the receiving end of the springs, there is a small microphone or its equivalent. It changes the mechanical vibrations that have traveled down the springs back into electrical vibrations which become the phase-shifted (delayed) signal that is combined with the undelayed, or original, signal.

The reverb circuit used in a commercial guitar amplifier is shown in Fig. 2-15. The signal comes in at the preamp grid; it is amplified, and fed to the "reverb-in" amplifier tube. From the plate circuit of this tube there are two paths for the signal to follow. One is directly to the input of the next stage through the resistor network shown above the reverb unit. The signal that follows this path is the undelayed signal.

The other path is through the reverberation unit. This consists of a speaker or reproducer, the delay springs, and the pickup unit. The signal is delayed as it passes through the springs, but it goes through the alternate path at normal speed. There are two signals coming from the same source—one is normal and the other lags behind by 29 or 56 milliseconds. They are combined in the plate circuit of the reverb output amplifier tube and sent along to the next amplifier stage.

The reverb control determines the proportion of delayed signal used. You can use just a little or a lot, depending on how deep you want the reverb effect. This, in effect, is a reverb volume control.

Transistor Units

Fig. 2-16 shows a reverberation unit with transistors. This unit is designed as an attachment instead of a part of the original amplifier. The basic action, of course, is the same. The signal comes in at the input jack to the base of Q1, which is an emitter-follower circuit in order to match the high impedance output of the amplifier. This is fed to the base of the second transistor, and coupled to a third transistor which drives the reverb springs. The pickup at the output end of the springs feeds its signal to the base of the output transistor (a common emitter circuit), which in turn feeds the output jack.

The reverb control is a 10,000-ohm variable resistor across the pickup. A foot switch can be used to cut the reverb effect in or out as desired. It does this by grounding the junction of two .005-μF capacitors in

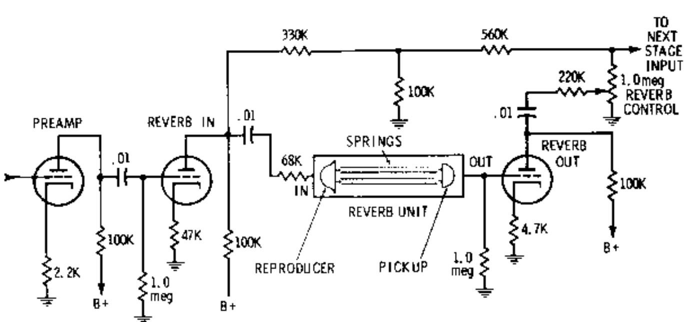


Fig. 2-15. Reverberation circuit.

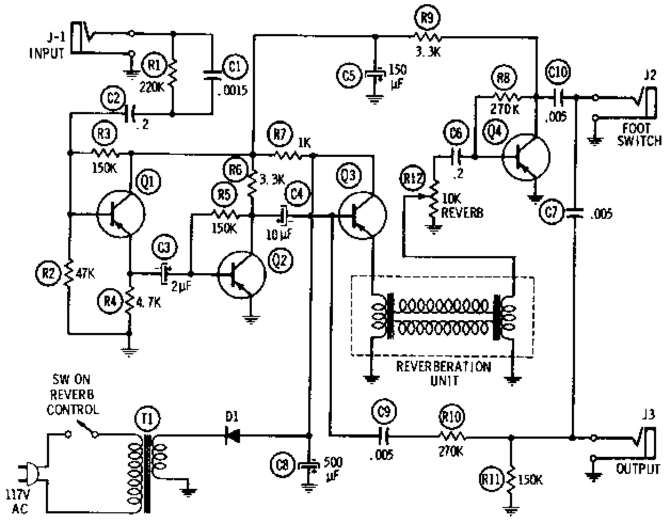


Fig. 2-16. Transistor reverberation unit.

series in the output circuit, eliminating the reverb signal.

MULTIPLE INPUT CONNECTIONS

One feature found in all but the smallest guitar amplifiers is multiple input connections. These are provided so that several instruments can be connected to the same amplifier at the same time. If two guitars (a lead and a rhythm), a bass, and a violin are plugged in together, for example, the whole band can use a single amplifier.

Some of the amplifiers have microphone inputs as well. These have slightly higher gain than the instrument inputs to compensate for the low output of microphones and to give ample volume for vocal choruses.

The main problem, of course, is not gain, but of mixing the various inputs. All of these must be controllable so that the volume of one instrument can be raised to take a solo passage, for example, while the rest stay below him. One volume control must not affect any of the rest. Instead of connecting them all together, it is necessary to isolate them by means of mixer stages. Correctly built, these give some gain too.

Fig. 2-17 shows a diagram of the most common mixer circuit, with four instrument inputs and a mike input. Each one has its own volume control; a master volume control to adjust the gain of all inputs at the same time is used later in the amplifier circuit.

Here (Fig. 2-17) the output of each instrument goes to the grid (input) of a triode tube. For economy, one of the popular twin-triodes (12AX7, 12AT7, etc.) is used. All of the grids are separate, but notice that all of the plates are tied together. They are connected in parallel, but with isolating resistors (R1, R3, R6, R7, and R9) in series with each plate circuit. These will not affect the gain, but they will help to keep one circuit from interfering with another. You may find isolating series resistors used in the grid circuits, too.

Individual volume controls are used, ordinarily in the input of each stage, although they will work the same in the plate circuits.

The microphone input uses a pentode tube instead of the triode for higher gain. Because of the increased gain, noise becomes a problem again. So, the tube type shown (5879) is a nonmicrophonic pentode especially designed for use in such a stage as this. The output goes to the same common line; all of the signals are mixed here and are fed through coupling capacitor C1 into the next stage of the amplifier.

Twin-triodes are shown in Fig. 2-17. You will probably find triple triodes (Compactrons) in some amplifiers.

TRANSISTORIZED MULTIPLE-CHANNEL INPUTS

You will find similar mixer circuits used in transistor amplifiers. In the circuit of Fig. 2-17 by replacing the tubes with transistors we have a transistor four-channel mixer. All ordinary bipolar transistors can be regarded as "triodes," so they can replace a triode tube in the mixer with the greatest of ease. The outputs will be combined just as before. You may find an extra amplifier stage between the input and the

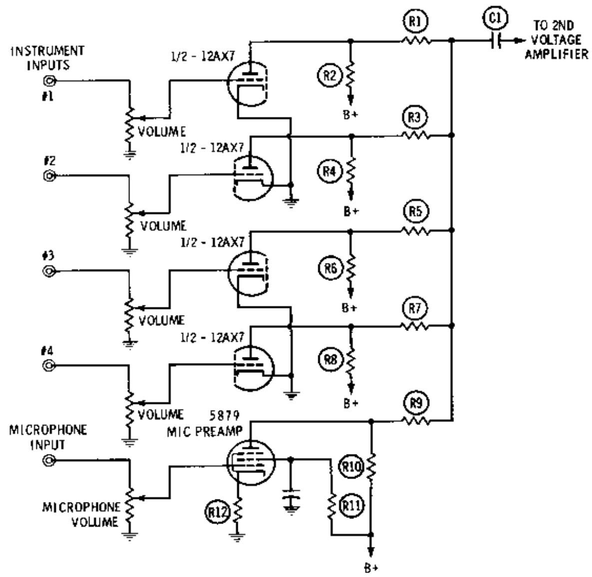


Fig. 2-17. Typical circuit for a 4-channel mixer.

mixing circuit, but basically the circuit will be just the same.

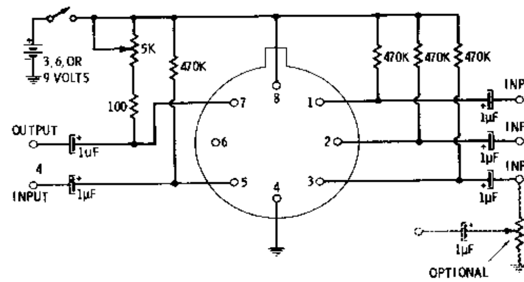
INTEGRATED-CIRCUIT MIXERS

Although, to the best of my knowledge, no integrated circuits are being used for mixers in commercial guitar amplifiers as this book is being written, I have no doubt that they will show up in the very near future. Fig. 2-18A shows a typical circuit using a Motorola HEP-581 "4-input Gate" IC. This is a small, round 8-pin IC, not much bigger than a TO-5 cased transistor. These ICs may be found soldered permanently (sic) into the circuit, or made to plug in for quick replacement.

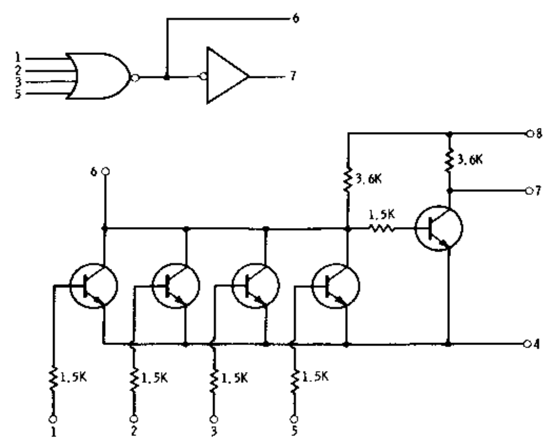
This unit is one of the simplest of the integrated circuits. It contains only five transistors and seven resistors. The internal circuitry is shown in Fig. 2-18B. This unit is not difficult to test. First, be sure that the normal input voltage is present on pin 8, then check each output with the scope, feeding a test signal into each of the four inputs. If there is no output signal and all of the external circuitry is alright, that is, no shorted or open coupling capacitors etc., then the IC is probably bad. Try a new one.

THE INTEGRATED-CIRCUIT AMPLIFIER

Certain ICs can be used as preamplifiers, also. Fig. 2-19 shows one of these, with the circuit, and the internal connections of the IC. Here again, the same



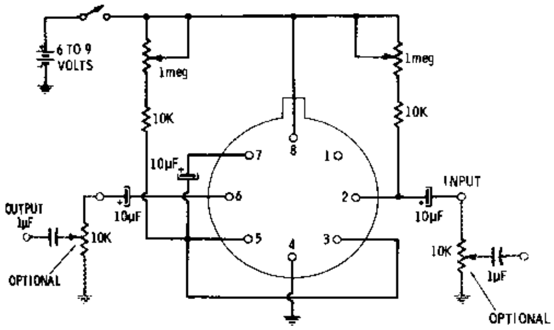
(A) Used as a 4-input mixer.



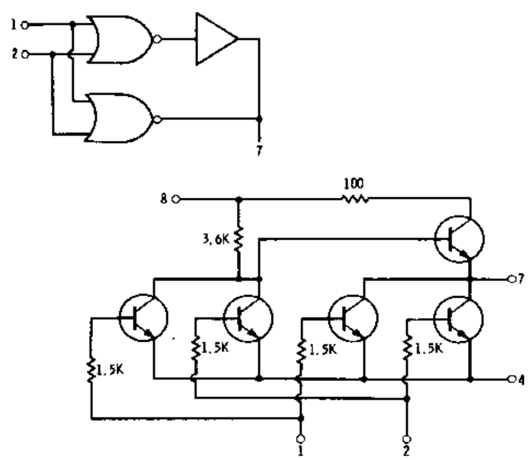
(B) Schematic diagram.

Fig. 2-18. Motorola HEP-581 4-input gate.

test methods can be used. First make sure that the IC has the correct voltage supply, and then check the output for signal level. In this one, you should find a good-sized voltage gain. The actual gain depends on the circuit, size of load resistor used, etc.



(*A*) *Used in high-gain amplifier.*



(*B*) *Schematic diagram of half the unit.*

Fig. 2-19. Motorola HEP-582 dual buffer.

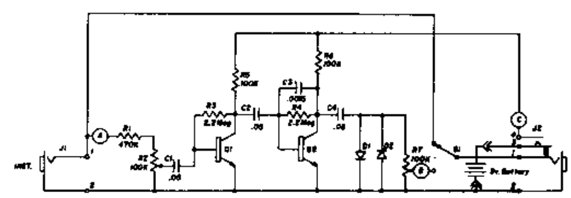
In the small amplifier section of this book, you'll find amplifiers which are "all IC." The whole amplifier circuit, including the output stage, is built in an integrated-circuit package. At this time, these are limited to about 5 watts rms maximum power, but you may find them later on with greater power output.

SPECIAL EFFECTS UNITS

There are several "special-effects" units which are used with electronic amplifiers to give the "new sound." You may find these in separate cases, plug-in, or built into the circuit of some of the more elaborate amplifiers. Separate or built-in, they all work the same, and can be checked the same. Most of these are fairly simple, and the standard tests can be used to troubleshoot them.

The Fuzz Box

The simplest of the special-effects units is called a "fuzz box", "scrambler" and so on. It does just what its name implies. Fig. 2-20 shows the schematic of a typical unit. The signal from the guitar is fed through the input jack to a standard two-stage transistor amplifier. Note the reverse-paralleled diodes D1 and D2 shunted across the output of the second stage. When the signal reaches a certain level, these diodes will go into conduction. This will clip the peaks from the signal waveforms; one clips the positive peaks, the other clips the negative peaks. Fig. 2-21 shows the input signal and the output signal after clipping.



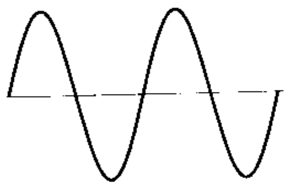
Courtesy Allied Radio Shack

Fig. 2-20. Schematic diagram of a "fuzz-box" circuit.

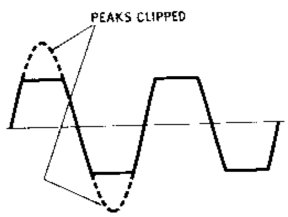
This clipping action, and the resulting almost "square-wave" shape, gives the tone its characteristic "fuzz" sound. The higher the signal level, the greater the clipping action. The volume control R2, at the input, determines the level of signal at the output and the amount (percentage) of clipping. So, this control is called the "fuzz-tone" control. At the output, another control, R7, sets the output signal level. From the output jack, the signal is fed directly into the input of the guitar amplifier. Switch S1 allows the fuzz tone to be switched in or out, at will. As you can see, in the "out" position, the signal goes straight through the unit to the output jack. In the "in" position, it goes through the fuzz circuitry.

With only two small transistors, and a very small power output, these units are powered by a 9-volt transistor radio battery. Some of them have clips inside the case, for holding a spare battery, in case it goes out during a performance.

Testing these is easy; just turn the switch on, set the fuzz control to maximum, and play a few chords. You'll be able to hear the fuzz effect if it's working. If no signal will go through the unit, check the battery voltage first; if this is up to normal, then check the input and output jacks, wiring, switch, etc., to make sure that the "signal-path" is not open. If these



(*A*) *Input signal waveform.*



(*B*) *Output signal waveform.*

Fig. 2-21. Input and output signals of a "fuzz-box."

are all in good shape, one of the transistors may be bad. Checking these can be done with the same tests and equipment used for any other amplifier, for that's all it is—a two-stage transistor amplifier.

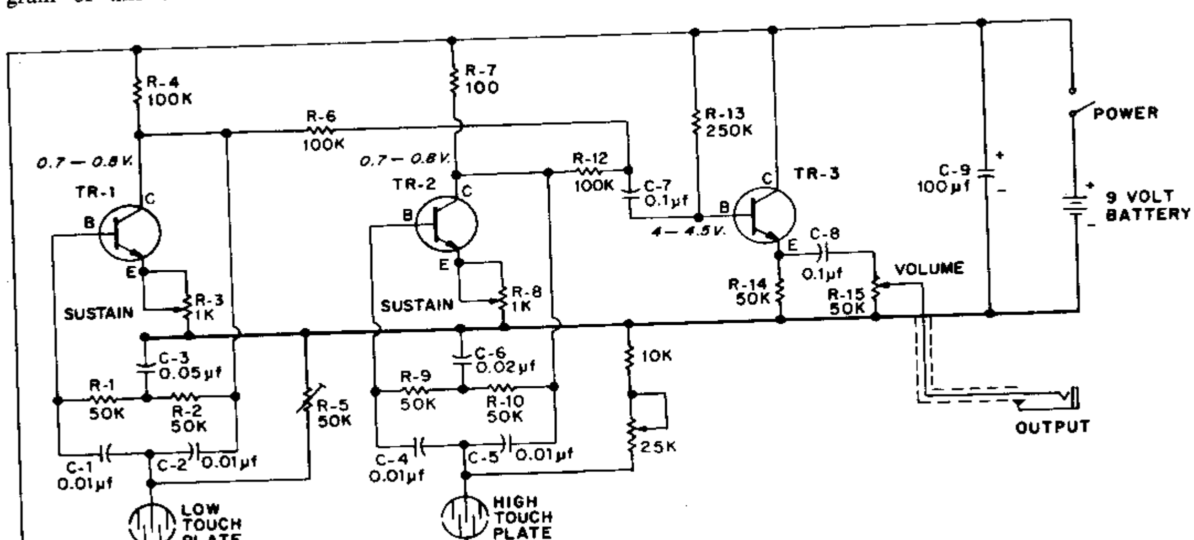
The Electronic Bongos

Another interesting special-effects device is actually an "electronic drum"; in fact, it is called the "electronic bongos." Fig. 2-22 shows the schematic diagram of this unit. Electronically, it's fairly simple. There are two pulse generators controlled by capacitive "touch-plate" switches. These are special capacitors used to trigger electronic circuitry; you've probably seen them used for "push buttons" (which didn't push) on automatic elevators.

Each unit has two of these, like the original bongo drums, which are played in pairs, one tuned to a high pitch and the other to a lower. Note that one is marked "low" and the other "high." Each of these is actually a triggered *oscillator*. They are set up so that the circuit is *not* oscillating, but is ready to, at even the slightest disturbance. This disturbance is provided by touching the touch-plate switches. This starts the oscillator to working, and it gradually dies away and stops.

This generates a very sharp, short "pulse-train" signal, which, fed to an amplifier, creates a sound exactly like the hollow ringing sound of a real bongo drum. The length of this pulse-train can be controlled by the "Sustain" control in each circuit; R-5 in the low and R-11 in the high. The output of both bongo circuits is fed to an output amplifier, TR-3, and then to the amplifier, through the output jack and cable. This can be fed into one of the low-gain inputs, since the output is at a fairly high level, much greater than the output of a microphone or guitar pickup.

Testing these units is not difficult. If the unit doesn't work at all, check the battery voltage. If this is up to normal, check the output amplifier stage by feeding a signal into its base. A bad transistor, open



Courtesy Allied Radio Shack

Fig. 2-22. Schematic diagram of an electronic bongo.

coupling capacitors C-7 or C-9, or open wiring, can kill the output.

The bongo oscillators can be very easily checked by running the "setup" tests to make sure that they can oscillate. Turn the volume control full on, then move the setup control, R-5, for the low, or R-11 for the high. Move this just a very little bit. If the oscillator is working, you'll hear the amplifier begin to squeal. This should start at a very low pitch, and gradually go higher as the control is turned. If you hear this reaction, this oscillator is alright. Check the touch-plate wiring; touch the contact with the tip of a screwdriver, test prod, etc. If the oscillator circuitry is alright, this should make the oscillator work, and you'll hear the hollow "bong" in the speaker.

If the control works, reset it to the correct place. Turn it so that the squeal goes *down* in pitch and then stops. Check by touching the touch-plate. Do not set this control so that the squeal goes up in pitch and then stops; this is wrong, and the bongo won't work.

If one bongo works but the other does not, you can cross-check between the dead one and the working circuit for dc voltages, resistance, etc. Parts values are very close to the same, aside from the frequency-determining capacitors. Look for things like leaky capacitors, leaky or shorted transistors, or open wiring. Fig. 2-23 shows how the touch-plate switch unit is assembled to make the "bongo" unit on the cabinet.

Electronic Drummer

Now, we come to a class of units which I must admit makes me just a bit resentful; they're of the type where "machines replace people." These are special-effects devices which can be plugged into the main amplifier, and will generate any kind of rhythm beats—they replace the drummer.

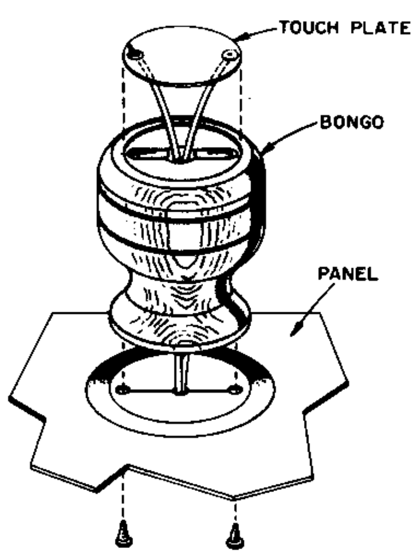


Fig. 2-23. Touch-plate capacitive switch used in the bongo.

These amazing devices are actually signal generators. By the use of pulse-generating circuitry of the same type as that used in computers, they can give very realistic imitations of the sounds of a bass drum, snare drum, and a "top-hat" type cymbal. The speed (tempo) of the beats can be set wherever necessary. In addition, they can be switched to provide the typical background rhythms of any kind of music. Anything from what we used to call a simple "boom-trot" (march or 2/4) rhythm to a syncopated rock beat can be obtained.

Fig. 2-24 shows a complete schematic diagram of one of these units, a Knight-Kit KG-392 "Combo Sideman." This unit works by generating pulses. For example, the hollow boom of the bass drum is generated by a very low-frequency (phase-shift) oscillator shown at the upper left corner of the schematic. This oscillator has a pitch adjustment and a sustain control to allow the musician to select either the full-toned boom or the shorter "thud" sound of a muffled bass drum. The characteristic sound of a snare drum is made by keying in very short bursts of "white noise," which is basically a blow or rushing sound. Another circuit generates the bell-like tone of the cymbal, once again keyed in bursts on the beat.

By adjustment of the Rhythm Control, the bass drum can be made to sound on the down-beat, with the snare drum and cymbal on the up-beat, or on the after-beat, depending on the type of rhythm desired. For syncopated beats, any of the "instruments" can be "accented" (made louder). For solos by the lead or melody instrument, where the drummer usually lays out (stops), pressing the "Solo" button will make the device play a straight or unaccented series of beats, such as bass drum alone as background.

Testing and Repairing

The "electronics" in this unit are all on a single PC board, which is wired to the controls through cables. The basis of the variable-tempo and rhythm circuitry is a group of flip-flops, seen at the lower left of the schematic (Fig. 2-24). Although integrated circuits may be used in future models, the flip-flops in this unit are "discrete components," and so can be tested and repaired individually if need be.

Once again, the fastest test method is the process of elimination. Check the unit; if none of the sections is working, check the battery or power supply first. Nothing can work without the proper voltages. After making sure that voltages are within limits, check the

NOTE:Pages32-33containedschematicforKnight/RadioShack“ComboSideman”Rhythm machine.Schematicwastoodetailedforscanner,andoriginalofpoorquality.

unit to find out just which sound effect isn't there. Always check interconnecting cables, switches, and the rest of the simple things before suspecting any serious trouble in the electronics. Because of the hard wear that cables get, they can cause a lot of the troubles.

If, for example, the bass-drum sound is missing, but the snare and cymbal come through, go straight to the bass-drum oscillator circuit, shown in the upper left of the schematic. This circuit is basically a low-frequency phase-shift oscillator, of the same type used for vibrato in other amplifiers. Any of the capacitors or resistors being off-value, either open or shorted, can kill the oscillator or make it run far off the correct frequency. Check the oscillator transistor for open or shorts, and check all resistors and capacitors for proper value, leakage, or drift. Note that there are quite a few diodes used as *couplers* in this unit. If any one function fails but the basic oscillator seems to be working, check the coupling diodes for either open or shorts. (Diodes also work as "shapers" in some of the wave-forming networks.)

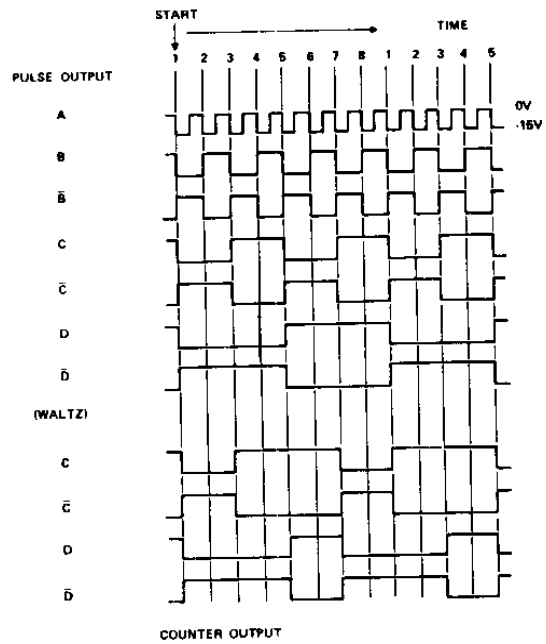
In this type of circuitry, the scope will be by far the most helpful test instrument. The output waveforms of each stage are shown on the schematic. The scope will tell you instantly and definitely whether a given stage is working or not. You can also use it to follow the different signals through the coupling and shaping circuits to the output.

Rhythm Console

Fig. 2-25 shows the schematic of a much more elaborate unit of this type. This is the Knight-Kit "Rhythm Console," and is basically like the unit we just looked at. However, this unit has a total of *ten different* rhythms—waltz, tango, bossa nova, cha-cha, beguine, mambo, rumba, slow rock, twist and swing. Any of these can be selected by a push-button switch on the front panel. It can produce the sounds of *eight* different percussion instruments: bass and snare drums, two bongos (high and low), claves, maracas, cymbal, and even a cow bell.

Computer circuitry is used even more extensively here than in the previous unit. The basis of the sounds is once again a group of phase-shift oscillators, set so that each one is right on the verge of going into oscillation, like in the electronic bongos. These oscillators are triggered on by a keying pulse; this pulse comes from a unit which has four astable multivibrators, plus a countdown circuit for forming the desired rhythm. The pulses are fed through buffer amplifiers, differentiating circuits, and gating circuits to control both the output and frequency of the oscillators. The combined output of the oscillators is then mixed and fed through an output amplifier. The output of this amplifier goes to a low-level input jack on the main amplifier.

Servicing and testing methods are just the same as those used for the preceding unit. Counters, oscillators, gates, etc., can be easily checked for proper output with the oscilloscope. Fig. 2-26 shows the normal counter output waveforms. Fig. 2-27 shows the waveforms at the AND gate, Fig. 2-28 shows the differentiator output, and Fig. 2-29 shows the output of the OR gate.



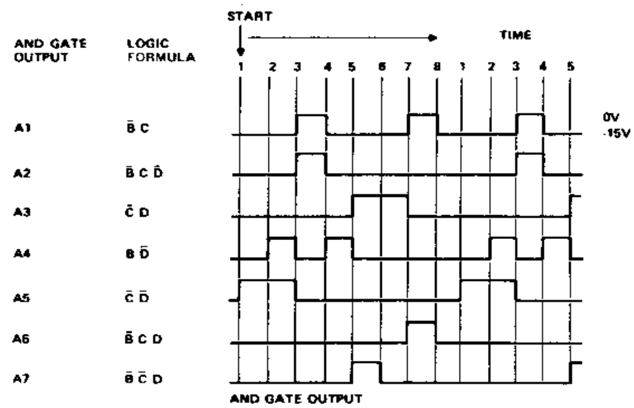
Courtesy Allied Radio Shack

Fig. 2-26. Normal counter output waveforms.

Once again, the counters, etc., are discrete component types, so that they can be repaired if a part fails. This, too, will undoubtedly be "out" in an IC version, possibly before this book is printed. However, servicing will still be the same. If any stage does not have the normal output, all external parts are in good shape, and correct voltages being supplied, etc., then you replace the IC.

Maestro Rhythm 'N Sound for Guitar, Model G-2

Another interesting special effects device is the Maestro Rhythm 'N Sound for Guitars. The schematic for this unit is shown in Fig. 2-30. This unit is a completely transistorized sound modification device de-



Courtesy Allied Radio Shack

Fig. 2-27. AND gate output waveforms.

signed to be used with a guitar and amplifier. The circuit operation follows.

Input Preamp No. 1—Transistor Q1 amplifies all input signals from the guitar except those sent to Fuzz Preamp 1, Q18. The No. 1 Input Preamp's output is directly connected to the Input Preamp No. 2, Q2, and through the Natural Amp tabswitch to Percussion Modulator Q22. In addition, when the Echo Repeat, Wow Wow, and Color Tones 1 or 2 tabswitches are off, the output signal from Preamp 1 connects to Output Preamp Q29.

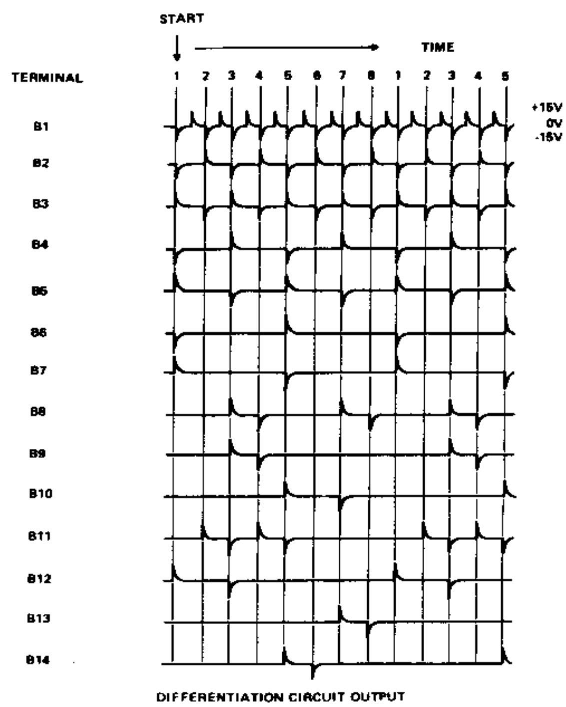
Input Preamp No. 2—Transistor Q2 provides a second stage of amplification for the previously amplified guitar signal from Preamp No. 1. This output signal is applied to Pick Detector Preamp Q3, provided one or more of the following tabswitches are in the on position: Wow Wow, Echo Repeat, Brush, Clave, Tambourine, Bongo and String Bass. When the String Bass tabswitch is in the on position, the signal from Input Preamp No. 2 Q2 is also connected to Squaring Preamp Q8.

Pick Detector Preamp—Transistor Q3 provides further amplification of the previously amplified guitar signal from Input Preamp No. 2 Q2 through the Sensitivity Control VR1. By properly adjusting the Sensitivity Control, this circuit amplifies only the strong guitar signal which is produced when a string is initially "picked." The output signal from this preamp is connected to Pick Detector Q4.

Pick Detector—The signal from Pick Detector Preamp Q3 is converted (rectified) by Q4 to a positive voltage pulse suitable for triggering the One-Shot Multivibrator Q5-Q6. Since the positive voltage pulse is capacitively coupled, only rapid voltage changes will be sensed by the One-Shot Multivibrator.

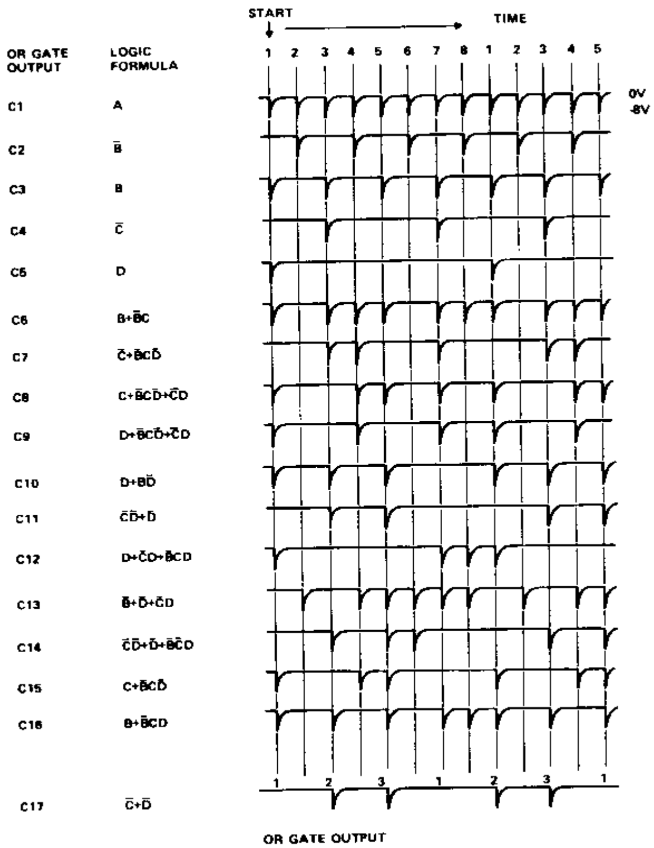
One-Shot Multivibrator—When Q5-Q6 is triggered by a positive pulse from Pick Detector Q4, this circuit momentarily grounds the capacitor connected to the base of Pulse Former Q7 and, when the Wow Wow tabswitch is on, the base of the Wow Wow Shaper Amp Q25 through diode D8.

Pulse Former—A strong positive voltage pulse is produced at the collector of Q7 when its base is grounded through a capacitor by the One-Shot Multivibrator Q5-Q6. The voltage pulse from the Pulse



Courtesy Allied Radio Shack

Fig. 2-28. Differentiation circuit output waveforms.



Courtesy Allied Radio Shack

Fig. 2-29. OR gate output waveforms.

Former is used to key the circuits listed below providing their respective tabswitches are in the on position:

Noise Amp Q 15 (Brush)
Noise Gate Q16 (Brush)
Tambourine and Clave Generator
Bongo Generator Q17 (Bongo)

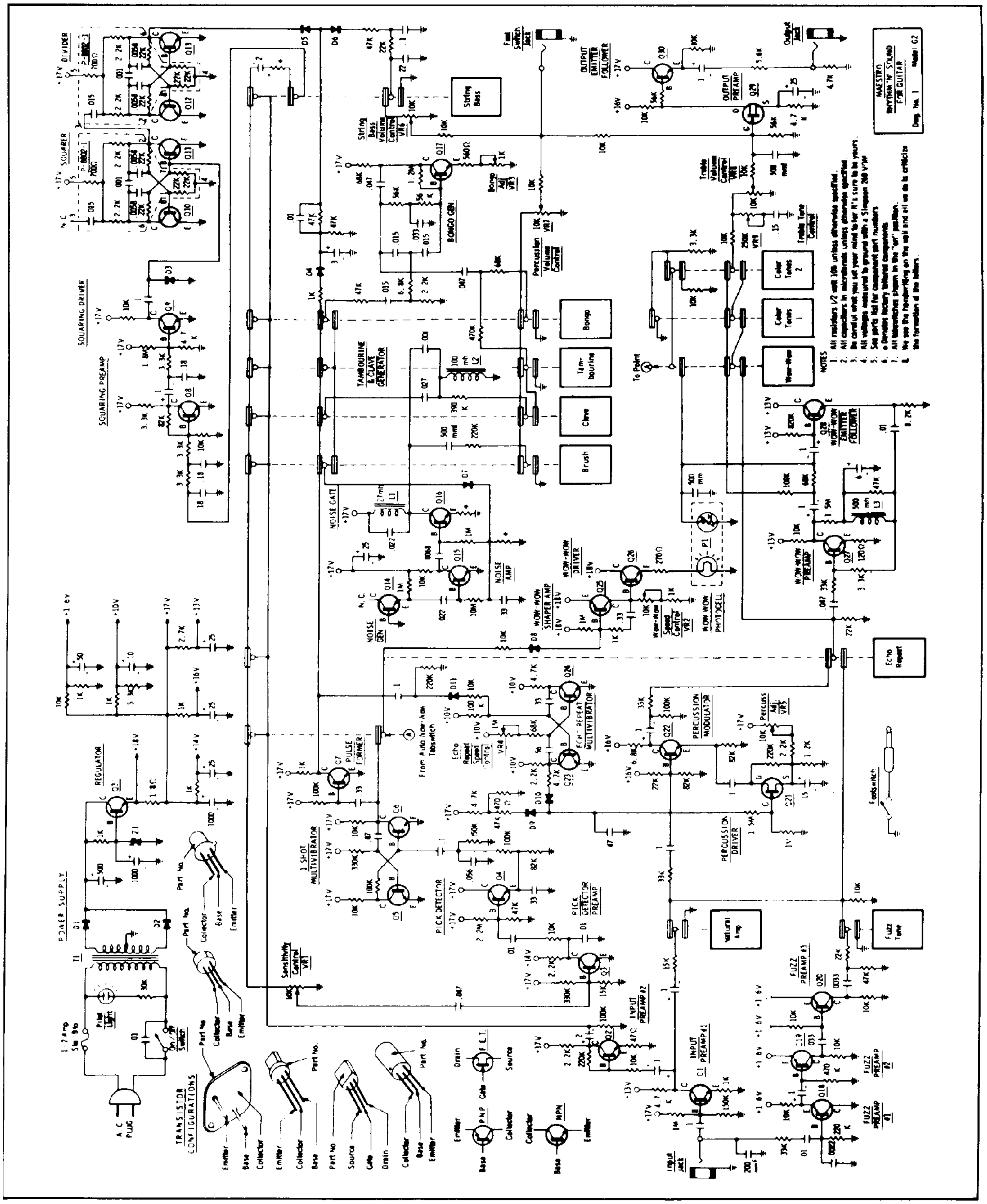
In addition to these circuits, the pulse from the Pulse Former is used to key through diode D4, diodes D5 and D6 which are in series with the output from Divider Q12-Q13. Also, the Pulse Former output pulse is used to trigger (through diode D11) the Echo Repeat Multivibrator Q23-Q24, causing it to restart with the pick of the guitar string.

Squaring Preamp—The amplified guitar signal from Input Preamp No. 2 Q2 is RC filtered to produce a square waveform signal, and then amplified by Q8. The output of Squaring Preamp Q8 is connected to Squaring Driver Q9.

Squaring Driver—The filtered and amplified guitar signal from Squaring Preamp Q8 is further amplified and clipped (diode D3) to provide a square waveform drive signal for Squarer Q10-Q11.

Squarer—The amplified and clipped guitar signal from Squaring Driver Q9 is converted to a square-edged waveform signal. This square-edged signal is used to drive Divider Q12-Q13.

Divider—The square-edged signal from Squarer Q10-Q11 is divided down by Q12-Q13 to a square waveform signal of half the input frequency. Example: A 440-hertz squared signal becomes a 220-hertz square waveform signal. The output signal from the Divider is connected to a diode keying circuit (diodes D5 and D6) which is keyed by the positive voltage pulse from Pulse Former Q7 through diode D4. The output of the Divider circuit then connects to the String Bass tabswitch, String Bass Volume Control VR6, and on to Output Preamp Q29.



Courtesy Gibson, Inc.

Fig. 2-30. Schematic diagram of the Maestro "Rhythm 'N Sound for Guitar," Model G-2.

Noise Generator—A constant B+ voltage applied to the emitter of transistor Q14 causes the internal base-emitter junction to zener breakdown, thus producing a constant random noise signal. The output of the Noise Generator is connected to Noise Amp Q15.

Noise Amp—When not keyed, Q15 acts as a blocking circuit between Noise Generator Q14 and Noise Gate circuit Q16. Two things happen to the Noise Amp circuit when it is keyed by a positive voltage pulse from the Pulse Former Q7 through diode D7:

1. It amplifies the noise signal from Noise Generator Q14. (A positive voltage applied to the base of the Noise Amp biases it "on.")
2. The amplified noise signal is allowed to pass to Noise Gate Q16.

Noise Gate—The noise signal from Noise Amp Q15 is amplified and voiced when Noise Gate Q16 is momentarily biased on by a positive voltage pulse from Pulse Former Q7. Output signal from the Noise Gate is connected to the Brush tabswitch and Tambourine circuit L2.

Bongo Generation—Q17 is a low-frequency oscillator that produces a short duration audio signal of diminishing amplitude when excited by a positive voltage pulse from the Pulse Former Q7. The output of the Bongo Generator is connected through the Bongo tabswitch and Percussion Volume Control VR7 to Output Preamp Q29.

L2 Clave and Tambourine Generator—A positive voltage pulse from Pulse Former Q7 excites the Clave Generator (mainly coil L2 and capacitors) into momentary oscillation. The Clave Generator output signal is connected to the Clave tabswitch. When the Tambourine tabswitch is on, the Clave circuit is combined with Brush signal to produce the Tambourine signal.

Fuzz Preamps—These three preamps, Q18, Q19, and Q20, amplify and clip the input signal from the guitar. The output signal is obtained from the third Fuzz Preamp. (The output is like the original input waveform except the waveform peaks are clipped.) The output from Fuzz Preamp No. 3 connects through the Fuzz Tone tabswitch to Percussion Modulator Q22, or through the Echo Repeat tabswitch to Wow Wow Preamp Q27 and Output Preamp Q29 when the Wow Wow and Color Tone tabswitches are off.

Percussion Driver—Positive voltage pulses from the Echo Repeat Multivibrator Q23-Q24 are converted into highly linear momentary drain-to-source resistance changes by Q21. These resistance changes effectively ground the emitter element of Percussion Modulator Q22.

Percussion Modulator—The audio signal from Input Preamp No. 1 Q1 and/or Fuzz Preamp No. 3 Q20 is applied to the base of Q22. When Q22 emitter element is momentarily grounded by Percussion Driver Q21, a short pulse of audio signal is allowed to pass. This audio pulse is applied through the Echo Repeat tabswitch to Wow Wow Preamp Q27 and to Output Preamp Q29 when the Wow Wow and Color Tone tabswitches are off.

Echo Repeat Multivibrator—This multivibrator, Q23-Q24, runs continuously except when restarted by a pulse from Pulse Former Q7 through diode D11. The multivibrator runs at the speed set by Repeat Speed Control VR4. As this circuit runs it produces strong positive voltage output pulses that are connected through the diode D10 to the Percussion Driver Q21.

Wow Wow Shaper Amp—When the base element of Q25 is momentarily grounded through diode D8 by One-Shot Multivibrator Q5-Q6, a positive voltage pulse is developed at the collector. This output pulse is connected directly to Wow Wow Driver Q26.

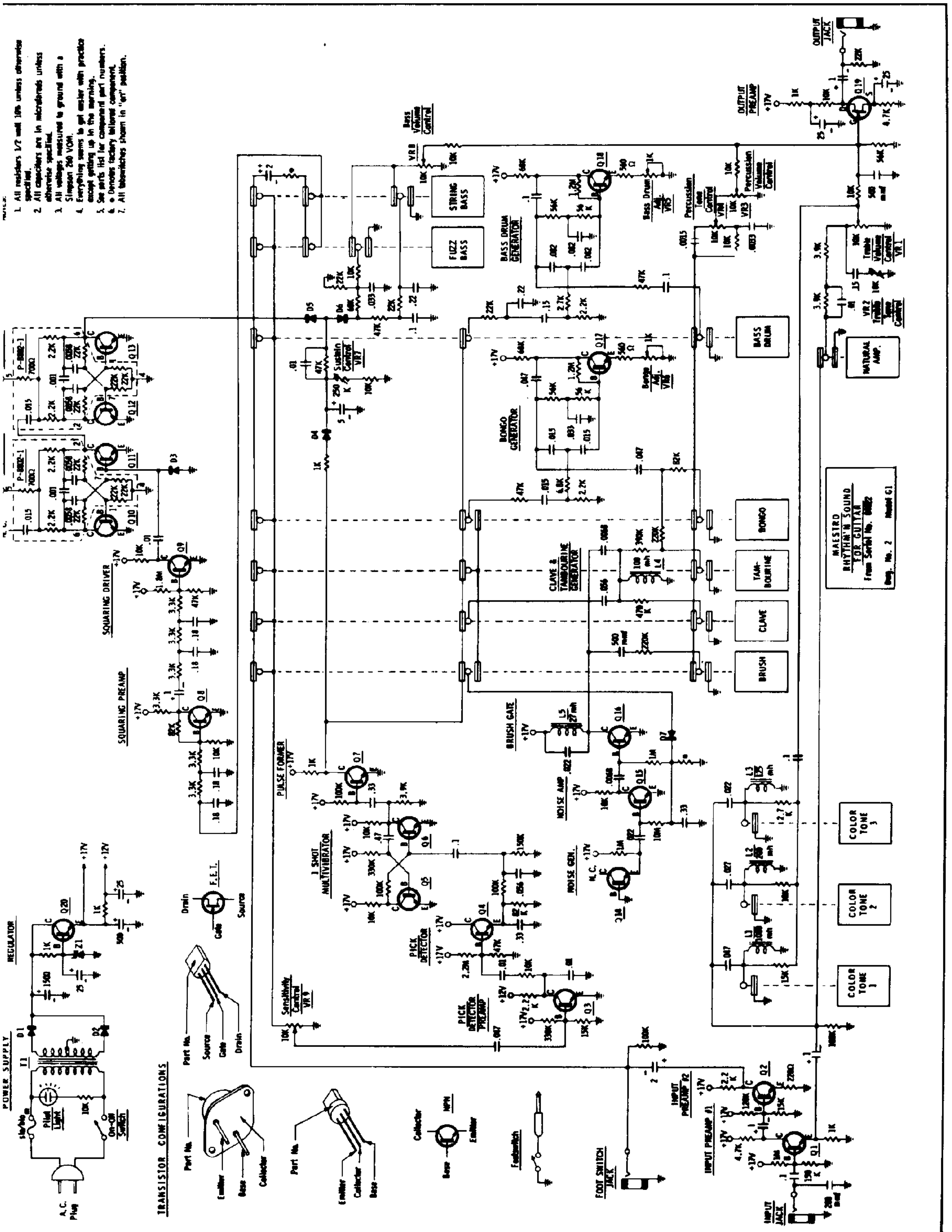
Wow Wow Preamp and Emitter Follower—The audio signal from Input Preamp Q1, Fuzz Preamp Q20, or Percussion Modulator Q22 is applied to this preamp. Wow Wow Preamp Q27 is a variable tuned circuit that amplifies only the audio signal near the frequency to which it is tuned. This frequency range is approximately 300 to 1700 Hz. The Wow Wow Photocell P1, together with the Emitter Follower Q28, determines the frequency to which the Wow Wow Preamp is tuned by electrically changing the effective value of the .01-μF capacitor attached to the emitter of this transistor.

Output Preamp and Emitter Follower—All Percussion, Bass, Natural Amp, Fuzztone, Wow Wow and Color Tone signals are combined and amplified by these two circuits, Q29 and Q30. The output signal from Emitter Follower Q30 connects to the Output Jack and on to a suitable power amplifier.

Regulator—Transistor Q31 works in conjunction with zener diode Z1 to regulate and filter the dc voltage produced by power transformer T1, diodes D1 and D2, plus several resistors and filter capacitors.

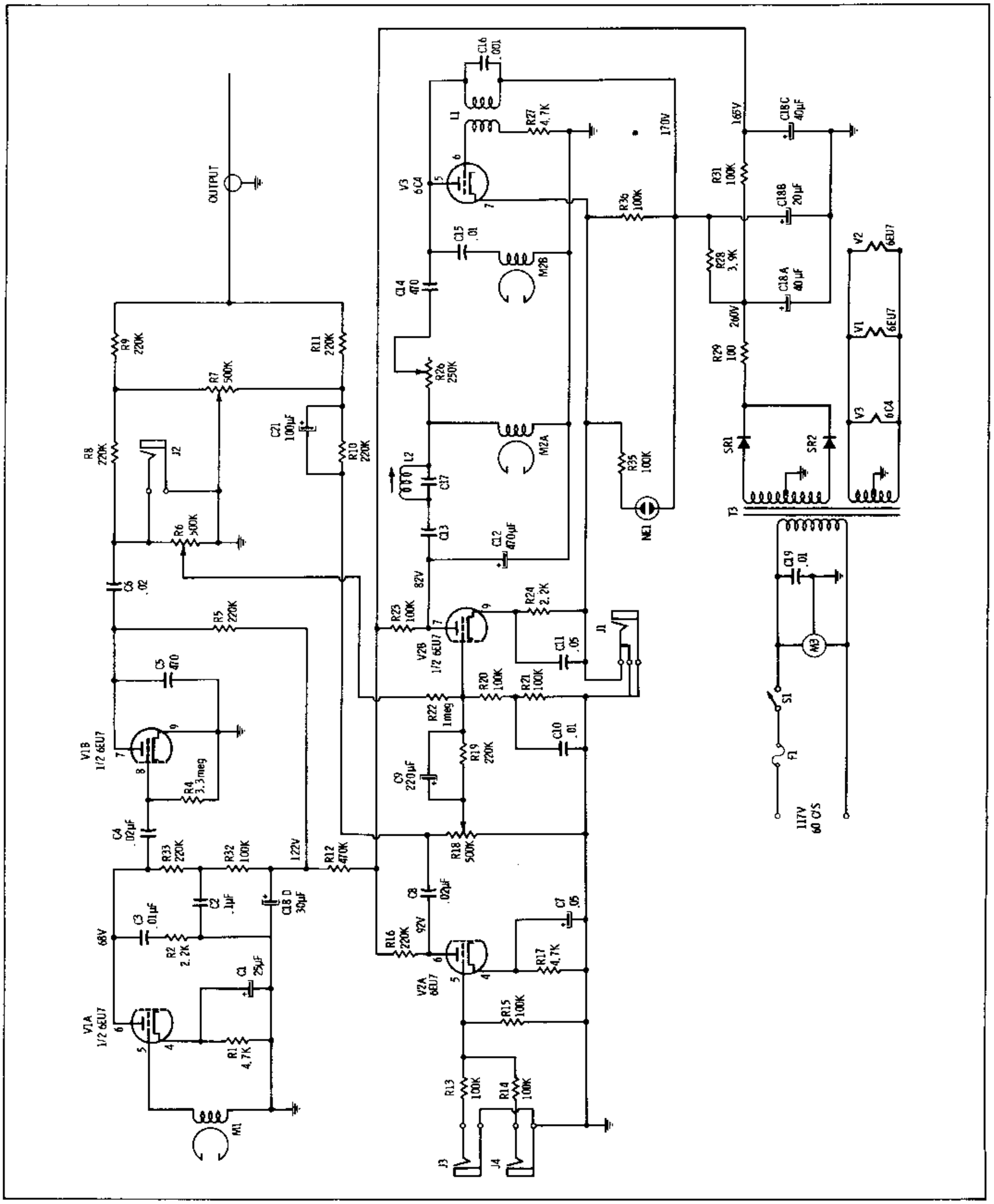
Maestro Rhythm 'N Sound for Guitar, Model G-1

This is another special effects unit designed to be used with an amplifier and guitar. Special effects sounds that can be created with this unit are bass drum, bongo, brush tambourine, clave, string bass, and fuzz bass. Circuit operation is very similar to the Model G-2 previously discussed. The schematic diagram is given in Fig. 2-31.



Courtesy Gibson, Inc.

Fig. 2-31. Schematic diagram of the Maestro "Rhythm 'N Sound for Guitar," Model G-1.



Courtesy Market Electronics Co.

Fig. 2-32. Schematic of Echoplex tape unit.

Echoplex Tape Unit

Another very interesting special effects unit, the Echoplex can be used with a guitar or other musical instrument. Its purpose is to create an "echo" sound along with a straight through guitar sound. It is an interesting unit, since it uses no mechanical springs to effect the delay.

The unit is provided with a movable magnetic head that enables it to match beat and rhythm. It is designed to provide simultaneous recording and playback, which introduces the necessary delayed signal for the echo. The desired number of echo repeats is set with the Echo Repeats Control. The Sliding Head Pointer is used to obtain the desired echo delay. A schematic of the unit is shown in Fig. 2-32.