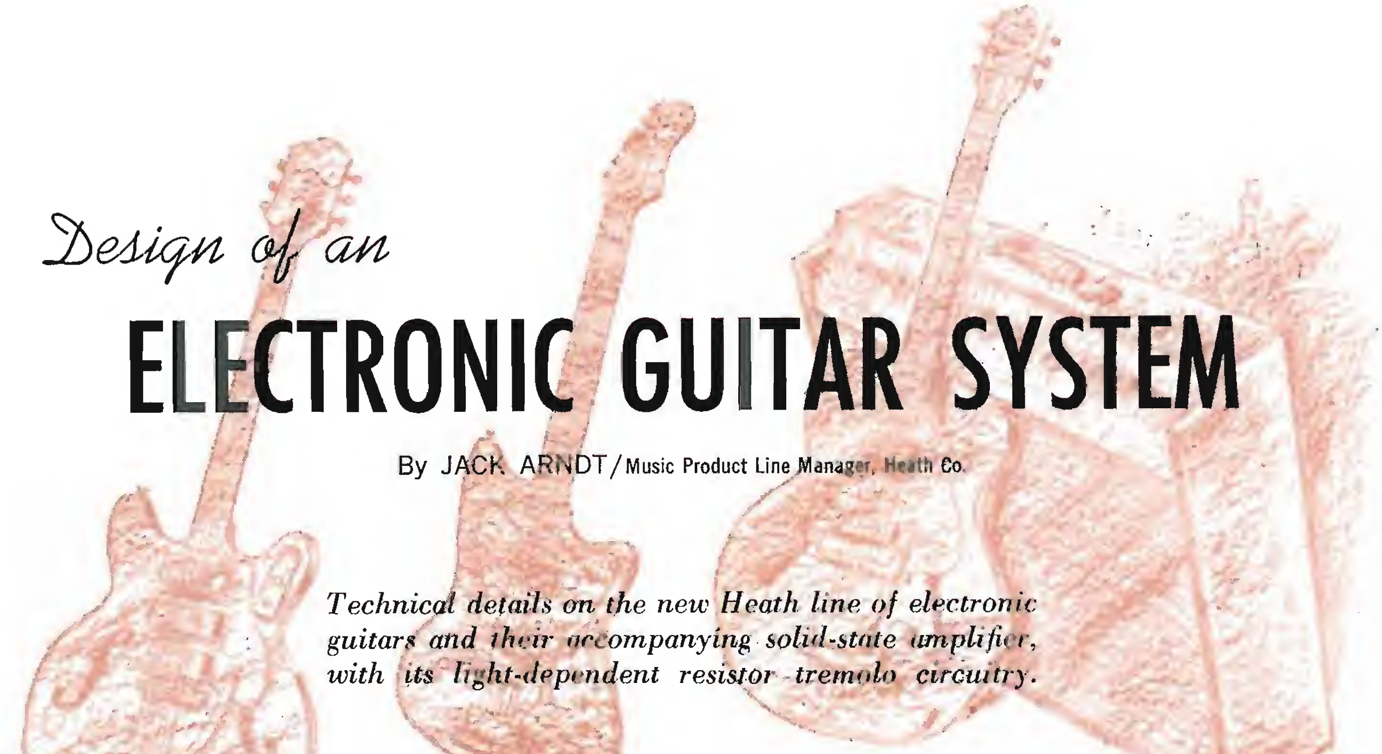


Design of an ELECTRONIC GUITAR SYSTEM

By JACK ARNDT/Music Product Line Manager, Heath Co.

Technical details on the new Heath line of electronic guitars and their accompanying solid-state amplifier, with its light-dependent resistor tremolo circuitry.

KIT-MAKERS from *Heath Co.* working with guitar-makers from *Harmony Co.* have recently introduced three styles of American-made electronic guitars, all in kit form. Following detailed instructions, the kit builder assembles and installs the electronic parts, mounts the vibrato assembly, tuning keys, bridge, strings, and other hardware, and then adjusts and tunes the entire guitar. There is no wood working or wood finishing involved. The guitar body and neck wood parts are supplied prefinished in a gleaming cherry red along with the carrying case which will later hold the assembled, tuned instrument. *(See cover photo for the Model TG-46 guitar and amplifier. —Editors)*

The three guitar kits are different in construction and therefore different in assembly. Average building time is from three to six hours, depending upon the unit. The two-pickup solid-body guitar is the easiest to build. On this unit, the two volume and tone controls, the pickup selector switch, and the output jack are mounted on and wired to a plastic guard plate before they are fastened to the guitar. On the two-pickup single-cutaway and the three pickup double-cutaway electro-acoustic guitars, the wires and components are formed into a completed harness before installation. The harness is fed into the guitar body through the F-shaped holes, and detailed instructions tell how to "fish" the components through the proper holes in the body. (Fig. 1 shows the circuit used in the three-pickup guitar.)

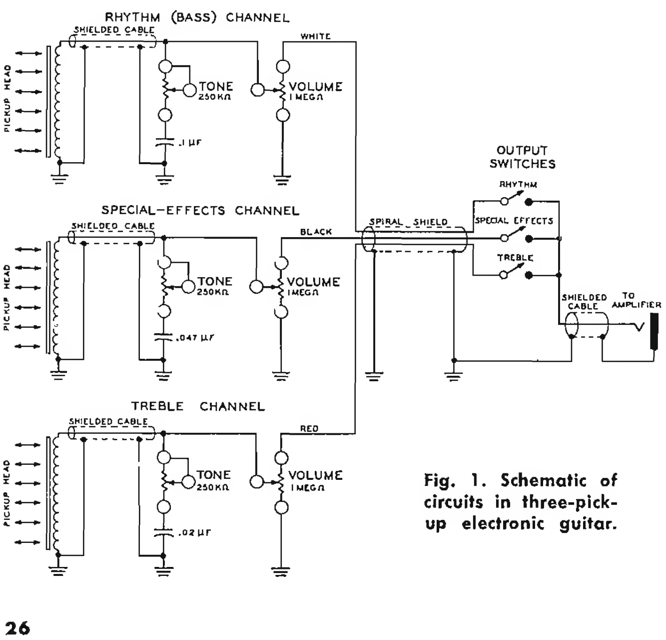


Fig. 1. Schematic of circuits in three-pickup electronic guitar.

After the electronic elements are mounted, the hardware parts are fastened to the guitar with screws through starter holes. Next, the bridge is precisely positioned and the strings loosely strung. Careful checks are then made of the adjustments for neck-bow and vibrato action.

Tuning

After the guitar is assembled, it is necessary to tune it. Several methods of tuning are described in the manuals and two means are furnished with each kit. An audible means is provided in the form of a small phonograph record which supplies actual guitar string tones. In this case, the tuning keys are adjusted until the guitar is in perfect tune with the record.

Another means, a visual one, is provided in the form of a new and ingenious little device called a "Vu-Tuner." This uses a resonant reed and indicator tuned to the lower E or sixth string tone of the guitar. The "Vu-Tuner" is clipped to a string adjacent to the bridge, and when the sixth string of the guitar is brought into proper tune, the reed of the device will vibrate in very noticeable resonance. Then the "Vu-Tuner" is removed and the fifth string is tuned in unison with the sixth string as the sixth string is held down behind the fifth fret. Next, the fourth and third strings are tuned in the same manner, each with the preceding string held down behind the fifth fret. Then the second string is tuned to the third string, while the latter is held down behind the fourth fret. And last, the first string is tuned to the second string, while the second string is held down, again behind the fifth fret. This provides quite excellent tuning, and because of the visual check of the initial step, it becomes quite an easy operation even for the inexperienced musician.

The Model TA-16 guitar amplifier has two channels of operation, each with bass, treble, and volume controls and two input jacks. One channel has tremolo and reverberation effects available (with foot-switch on-off action) while the other channel amplifies straight through.

The amplifier is rated at 60 watts of maximum peak power (peak ratings are commonly used in this industry), 25 watts of music power out- (*Continued on page* 79)

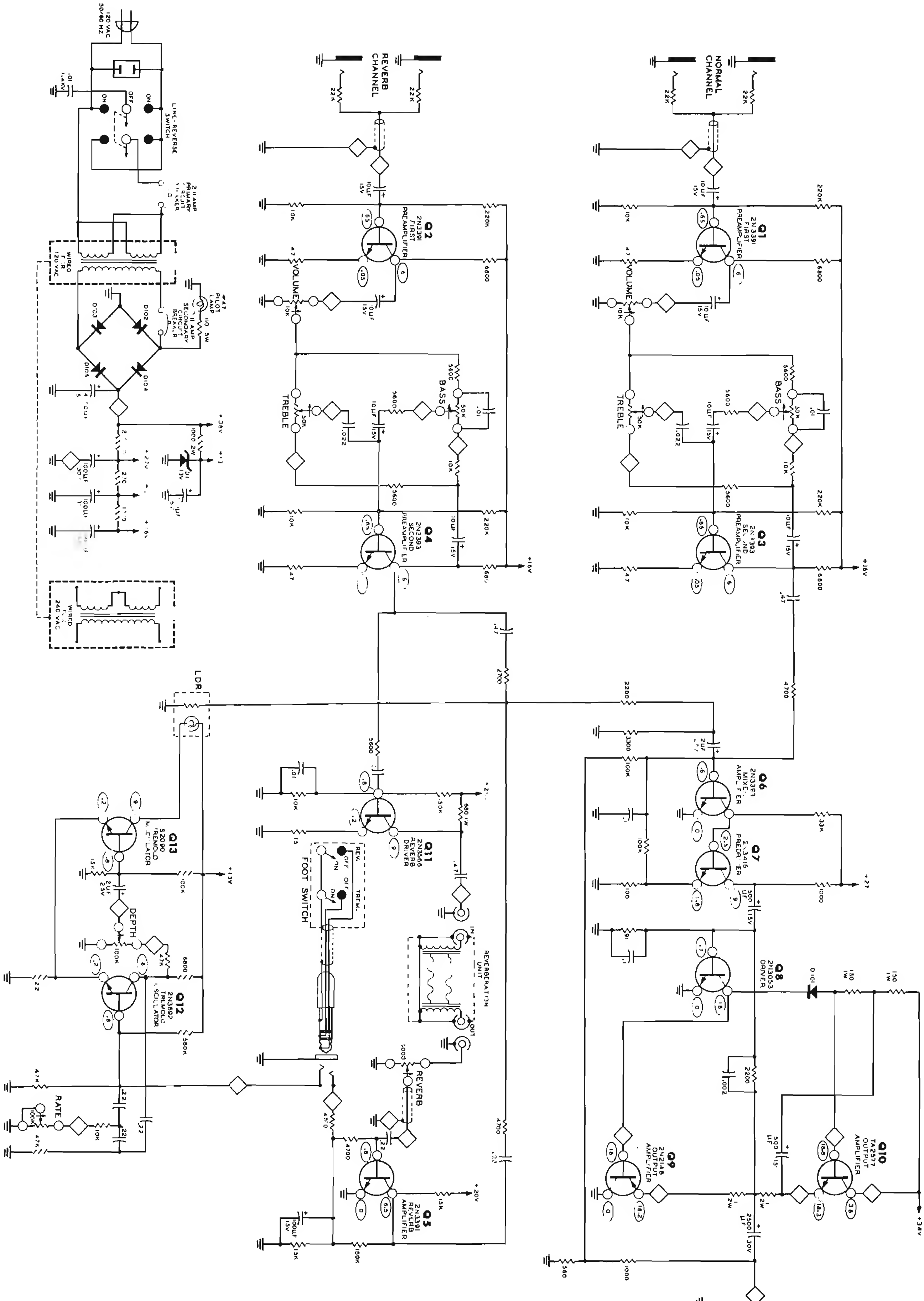


Fig. 2. Complete schematic of the guitar amplifier. Note especially the tremolo circuit at bottom right. The diamonds are circuit-board holes, while the numbers in the ellipses are positive d.c. voltages to ground, measured with a v.t.v.m.

Electronic Guitar System

(*Continued from page* 26)

put, or 20 watts of continuous power output. The latter two figures are at the EIA standard of 5% harmonic distortion.

The unit employs two special-response twelve-inch speakers. The power supply can be wired for either domestic 120-volt 50- or 60-Hz sources, or for export 240-volt 50- or 60-Hz sources. A line-bypass reversing switch (which also acts as an on-off switch) can be used to select the side of the power line which provides the least hum—an important feature for electronic guitars with their magnetic pickups.

Tube amplifiers are still commonly offered in the guitar industry, but this amplifier is an all-solid-state design with a fail-safe complementary transistor output circuit. The dependability, cool operation, and low microphonics of transistors make them most desirable for rugged portable service.

Tremolo Circuit

The tremolo section of the TA-16 amplifier employs an interesting new circuit (Fig. 2). It uses a light-dependent resistor (LDR) which can be found in the lower right-hand corner of the amplifier schematic. The LDR unit consists of a low-current lamp and a light-dependent resistance element. The value of the resistance varies with the brightness of the lamp—as the lamp glows brighter, the resistance decreases; as the lamp dims, the resistance increases.

The basic tremolo frequency is developed in transistor Q12 which is connected as a subsonic phase-shift oscillator. The frequency can be varied from approximately 4 to 14 Hz by the rate control. The amplitude of the oscillator frequency can be varied by the depth control. The signal is then coupled through a 2-μF capacitor to the base of tremolo-modulator transistor Q13. Transistors Q12 and Q13 are connected to a common emitter resistor to provide additional positive feedback for sustaining oscillation.

Transistor Q13 draws collector current through the lamp element in the LDR unit, causing the lamp to glow. Since transistor Q13 is amplifying the tremolo oscillator signal, the collector current will follow this signal, causing the lamp in the LDR unit to glow correspondingly brighter or dimmer. The resistance of the LDR will vary in the same way, from a very low to a very high resistance. Since the LDR is connected between the reverb-channel signal path and ground, its resistance variations will modulate the signal with the low-frequency tremolo signal. ▲

ELECTRONIC GUITARS and AMPLIFIERS

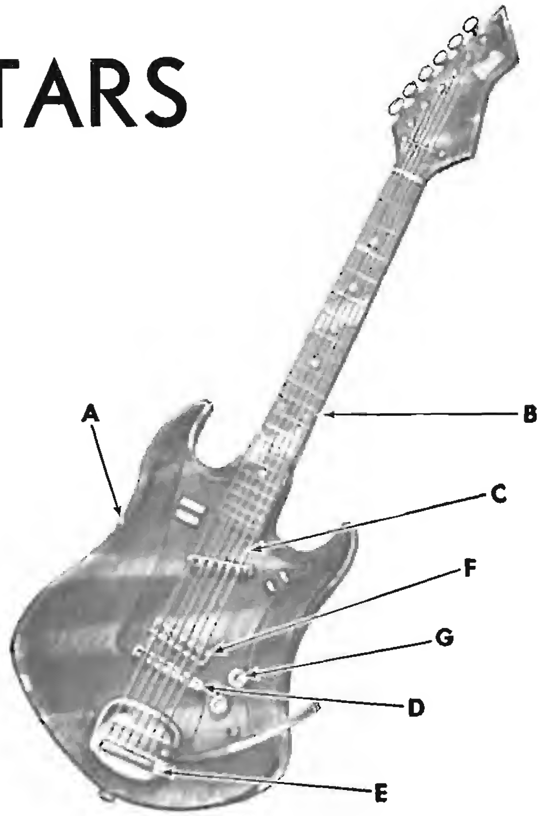


Fig. 1. Typical solid-body electronic guitar showing (A) body, (B) neck and frets, (C) strings, (D) bridge, (E) tailpiece and vibrato lever to change string tension, (F) pickup, (G) control.

By DANIEL QUEEN / Chief Engineer, Audio Products
Perma-Power Co.

Starting from the vibrating strings and progressing through the pickup, amplifier, and loudspeaker, this article covers the various design parameters of this most popular of all electronic instruments.

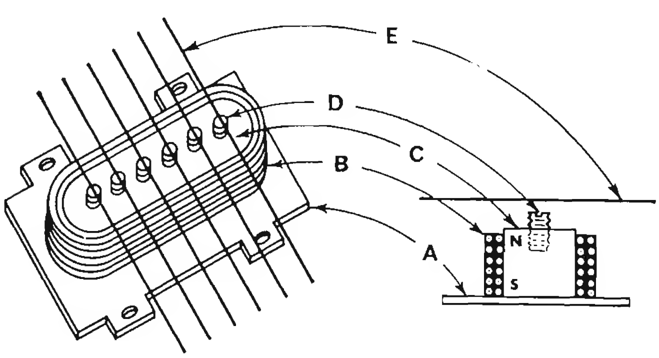
THE electronic guitar has become a new instrument, not merely the old amplified. True, there are guitars with acoustic output that have amplifiers attached, and historically such combinations were the prototype of the modern instrument. Initially, the electronic guitar consisted of a contact microphone attached to the sound chamber of an ordinary mechanical guitar. The microphone was connected to a public-address amplifier which drove a jukebox speaker. In time, because the contact microphone picked up room sounds and was subject to acoustic feedback, the magnetic pickup was developed.

At first the pickups were attached to an ordinary guitar. Then, in some cases, the sound chamber was reduced in size and eventually eliminated, with the body serving only to support the instrument during use.

The Mechanical Guitar

The mechanical instrument has mass. It consists of lumps of matter which when at rest tend to stay at rest and which when in motion tend to stay in motion; that is, they have inertia. This mass is analogous to inductance in an electrical circuit, which also impedes a change in motion—electron motion. All solid matter also has elasticity. When a force deflects the material, the force is stored and will be returned later as a result of the property of elasticity. This is analogous to capacitance in an electrical circuit. These properties are combined into resonant networks to provide mechanical musical instruments with their characteristic tone or timbre.

Fig. 2. A guitar pickup with its non-magnetic metal housing removed showing (A) lower pole piece, (B) output voltage coil, (C) permanent magnet, (D) individually adjustable upper pole pieces, which are located under each of the separate strings (E).



Transient characteristics, that is, the sharpness of the attack and decay of tones, are determined similarly to the manner in which transient characteristics of electrical circuits are determined. A high-"Q" mechanical structure, having large mass and elasticity but with low frictional or viscous losses (analogous to resistance in an electrical circuit), can be expected to have poor transient response, just as a high-"Q" electrical circuit does.

In a musical instrument, such "poor transient response" may be the very characteristic which is desired. After all, what rings more than a bell, with its large mass and low viscosity?

All musical instruments produce sound from vibrating members, such as strings, reeds, columns of air, or the lips of the performer. To couple this vibration to the outer air, a mechano-acoustic system such as a horn, sounding board, or sound chamber is used. In a grand piano, the sounding board is very large compared with most of the wavelengths it must couple; hence, it is very efficient.

In a small instrument such as the guitar, the sounding board cannot efficiently couple the low-frequency tones. It must have additional help provided by structural and acoustic resonances built into the instrument. Small stringed instruments are in effect a series of closely coupled resonant circuits, spread across the tonal range of the instrument.

Pickups & String Motion

The advent of the electronic guitar made possible the removal of this restriction on the small plucked-string instrument. Fig. 1 shows the structure and important parts of a solid-body electronic guitar, one with no sound chamber.

The pickup (Fig. 2) consists of a magnet, with one pole toward and one pole away from the strings. Around this magnet a coil is wound. Some guitars utilize a single magnet and coil for all strings while others use separate magnets or pole pieces at each string or group of strings.

The vertical component of the string vibration varies the length of the air path for the flux and therefore varies the flux intensity. This varying magnetic flux, passing through a coil of wire, produces an output voltage representative of the string motion. Note that steel strings must be used for electronic guitars since the strings must influence the magnetic flux.

The string motion varies along its length. At the ends there is no motion, since the string is clamped. The greatest motion is possible at the center. The motion may be split into multiples on fractional sections of the string; such motions are harmonics of the string fundamental. The degree of their occurrence will depend upon the point of initial plucking and the width of the plectrum. In a mechanical stringed instrument, these harmonics will cause the bridge of the instrument to twist, coupling the vibrations to the sound chamber. In this path, the masses act as series inductances, attenuating the upper harmonics and integrating the attack and decay transients.

In an electronic guitar, only that part of the string which is vibrating above the pickup will have its motion reproduced as sound. All harmonics and transients present at the pickup can be converted to electrical output, if desired. Referring to Fig. 3, we see the vibration of a string at its fundamental and at its second and third harmonics. From this it may be seen that the tone of the guitar can be varied by positioning the pickup.

If the pickup is near the center of the string, it will produce more of the fundamental and lower harmonics, and this will result in a mellow sound. If it is placed near the end of the string, it will pick up more of the upper harmonics, and this will result in a sharper sound. Most guitars now in use have at least two pickups which are used either alone or in combination. Some guitars are available with five or more pickups. Others have separate selectable pickups for the low and high strings. Proper pickup location is important in producing the proper tone color (Fig. 4).

The Power Amplifier and Loudspeaker

The pickups are designed to work into a high-impedance input of an amplifier. This amplifier must be capable of producing the dynamic and frequency range of the instrument and must have very good transient overload recovery.

Generally, an inexpensive amplifier that uses negative feedback will be on the verge of oscillation at a certain low and high frequency. A transient containing one of these frequency components will cause a damped oscillation, the amplitude of which may overload the amplifier, blocking it and causing severe unpleasant (non-harmonic) distortion. A guitar amplifier must be designed with these conditionally stable frequencies well outside the range of the guitar. Since the guitar tones come in bursts at much lower frequencies than the tones themselves, the amplifier must not oscillate at the burst frequencies.

Because the guitar amplifier is frequently being overloaded, it must have good transient recovery. The frequent occurrence of high-level, low-frequency tones together with high-level, high-frequency tones requires the amplifier to have low intermodulation distortion.

Only in frequency response is the guitar amplifier not required to match the usual high-fidelity amplifier. It is, in fact, desirable to design the amplifier with a narrower frequency response to improve transient stability. The lowest string fundamental on the electronic guitar is about 80 Hz. On the electronic bass guitar, the lowest string is 40 Hz. The highest string when fretted all the way down can produce a fundamental frequency of about 1300 Hz; it is doubtful that anything over the sixth harmonic of this string is worth reproducing, especially since the seventh harmonic is discordant. Thus, if one took the sixth harmonic of the upper string as the high limit, the required frequency response would be 40 Hz to 8 kHz.

There are two approaches to the loudspeaker system. It can either faithfully reproduce the vibration of the string above the pickup or it can be used to modify or color the tone.

First, a loudspeaker system can modify the tone by introducing its own resonances, much in the way the sound chamber introduces resonances in the mechanical instrument. Second, it can introduce non-linearities; that is, it can produce harmonic frequencies.

However, the loudspeaker is generally a single device, whereas a mechanical sound chamber is a multiplicity of structures, each responding differently. If a low tone and a high tone are struck on the mechanical instrument, they will excite different parts of the structure which will not interact harshly. In a loudspeaker, both tones move the same cone structure so that there will be interaction and possibly intermodulation distortion. For this reason, a guitar amplifier using a single loudspeaker designed with many resonances and non-linearities must be played carefully. Low-rhythm passages must not occur simultaneously with the high melody line. The use of well-spaced chords or bass figures while playing melody thus becomes nearly impossible. This

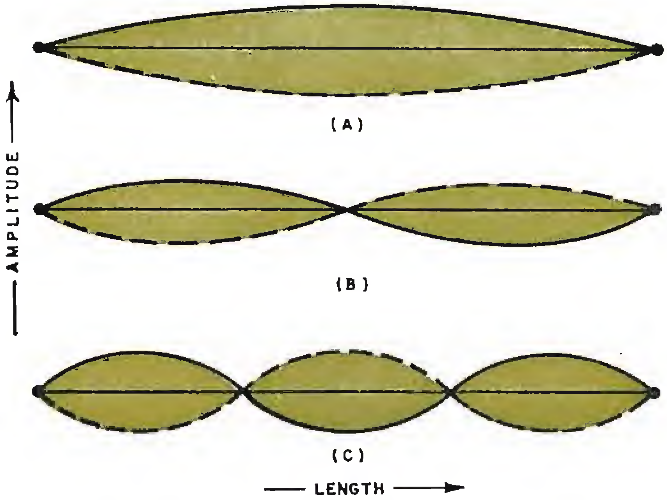
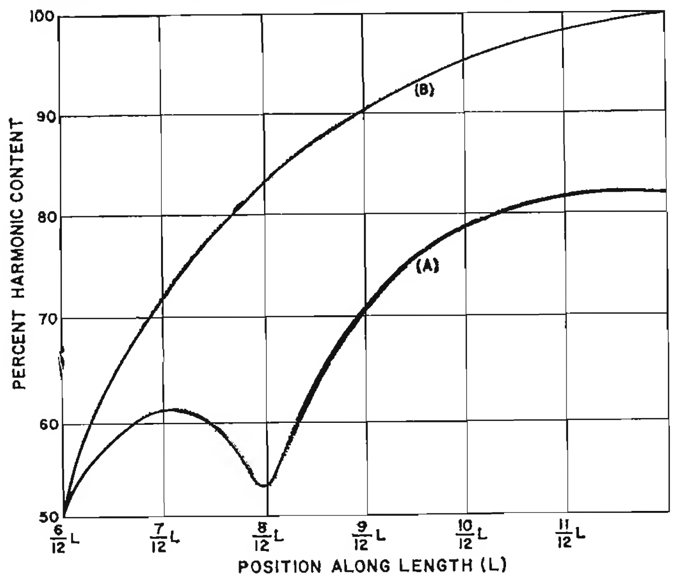


Fig. 3. The first three vibration modes of a stretched string are (A) the fundamental half-wave, (B) second, and (C) third harmonics.

Fig. 4. As the pickup is moved closer to the bridge, the harmonic content is increased. Curve (A) shows the harmonic content when the fundamental and second and third harmonics are excited equally. Curve (B) is for fundamental and infinite harmonics.



tends to further proscribe the playing style of the performer.

Furthermore, any system containing high-"Q" resonances must also have poor transient response. The sharp attack which is possible from the string and pickup system is no longer attainable, even if it is desired. It would be desirable, therefore, to use a loudspeaker having low distortion and few resonances. To obtain this, a thick, light cone is called for, and the speaker system must be able to handle the extremely high values of peak power that are produced.

How, then, can timbre and tonal effects be generated? Let us look at the non-linearity in the speaker (Fig. 5) which is graphed in Fig. 6. Low-amplitude signals are produced with little distortion, while high-amplitude signals are distorted. The low frequencies, which produce greater amplitudes of cone motion, will be distorted, while higher frequencies will not. The low tones, therefore, will be pleasantly rich in harmonics. The problem comes when the performer tries to play the high and low tones simultaneously. The high frequency is modulated by the low, producing non-harmonic raucous tones as shown.

To eliminate this, a separate loudspeaker for each range of tones could be used, but one would end up with at least one loudspeaker and crossover for each string. A better solution would utilize that which *can* be separate for each string—the pickup and its position.

From Figs. 4 and 6, one can see that the harmonics produced due to non-linearity and those produced due to off-center pickup position are very similar and will generate the same sound.

As the string is fretted to a higher frequency, the harmonic content will reduce, as it does in the non-linear loudspeaker. This is because the pickup is proportionally at a greater distance from the guitar bridge with respect to the length of the vibrating string.

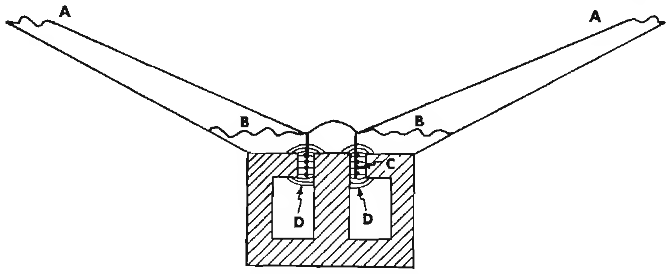


Fig. 5. A direct-radiator loudspeaker showing possible sources of non-linearity. The annular surround (A) or the spider (B) may be stretched beyond the point where they act as simple springs. Also part of the voice coil can move out of the magnetic gap at (C) and into the less dense leakage flux at (D).

Fig. 6. Loudspeaker non-linearity effects. A high-frequency signal (A), which is handled on the linear portion of curve, produces an undistorted output (D). A low-frequency large amplitude signal (B) may shift the operating point and produce the non-linear waveform at (E) which now contains harmonics. When both signals occur at the same time (C), the resultant at (F) contains harsh intermodulation distortion, shown alone (G).

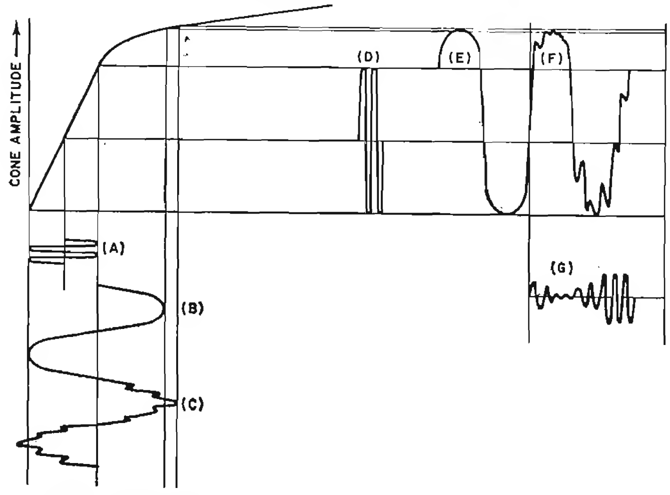


Fig. 7. The "Ampli-vox Baronet" guitar amplifier shown here has an output power of 5 watts and stands only 10-in high.

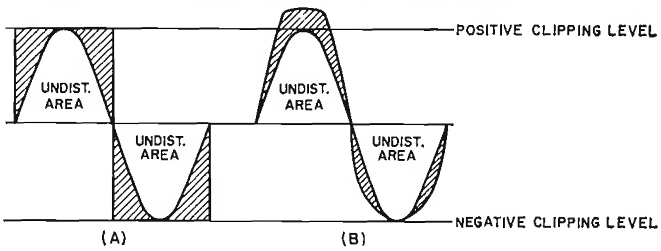


Fig. 8. (A) Amplifier that overloads in this manner with sharp clipping results in output containing many dissonant harmonics. (B) If clipping can be "softened" resulting in smoother waveform, as shown here, then the amount of dissonance is reduced.

Hence, judicious placement of the pickups, together with switching of the correct pickups by the musician, will produce the same timbre as harmonics intentionally added in the speaker system. Yet, because the sound originates at each individual string, the intermodulation distortion will be even less than that which can be obtained by having individual speakers for each tonal range.

Power Output and Overload

A large 100-watt system with 300 square inches of speaker cone area, while needed in a night club or concert hall, could hardly be used in a small practice room. On the other hand, a five-watt system (Fig. 7) will fill the practice room with as high a sound pressure level as the larger unit will produce in the concert hall.

Moreover, the two amplifiers, in their respective settings, can be designed to overload at almost the same sound level. The nature of this overload is of prime importance to the musician. Because the instruments are usually played in continuous overload for popular music, the overload characteristic must be carefully designed. It must result in the sound of harmony rather than distortion.

In Fig. 8, we see a sine wave that just reaches the clipping level of an amplifier. The sound intensity is the unshaded portion of the waveform. When the amplifier is fully overloaded, the shaded portion is added to the waveform, resulting in additional sound intensity. An amplifier with large amounts of negative feedback will produce a square wave at maximum overload (Fig. 8A), with 33% third harmonic, 20% fifth harmonic, 14% seventh harmonic, 11% ninth harmonic, 9% eleventh harmonic, etc.

In musical terms, the fundamental, third, and fifth harmonics are a non-dissonant spaced-triad chord consisting of the root, a doubled fifth, and a twice-doubled third, but the seventh, ninth, and eleventh harmonics are dissonant.

Table 1 shows the musical relationships of the harmonics. Tabulating the dissonance ratings of the odd harmonics in a square wave as shown yields a total rating of 12. On the other hand, the even harmonics of a rectified sine wave over this same approximate range of harmonics total only 4. Therefore, the best manner of overloading would *not* be one producing the familiar sharp clipping; instead, an overload waveform as in Fig. 8B containing a low dissonance rating is to be preferred.

The problem of overload characteristics is most acute for low notes, since their upper, dissonant harmonics are in a more sensitive region of the hearing curve. On higher notes, the upper harmonics are barely heard due to the lack of high-frequency response in the system.

The one component that will overload differently at low frequencies is the loudspeaker, as we have seen. For this reason, some amplifiers use speaker non-linearity to produce a favorable overload waveform. Often this is done by providing more undistorted amplifier power than the speaker can handle, consequently shortening the life of the speaker.

Another, more flexible method of softening the clipping at low frequencies involves the audio transformers. Fig. 9 shows the magnetization curves of a transformer core at middle and low frequencies. Note that the curve (A) is similar to the curve in Fig. 4B. When overload occurs at low frequencies, then, the waveform is rounded off instead of being sharply clipped. In class-B transistor amplifiers, non-linear *beta* will also serve this same purpose. However, this method may also result in high distortion at lower levels. Whatever method of overload characteristic modification is used, it is clear that the amplifier design certainly cannot stop at the clipping point.

Rating the Power Output

Because of the above considerations, the meaningful power rating of a guitar amplifier is not the steady-state sine-wave output, but rather the maximum output—distorted. Since the amplifier does not overload symmetrically, this is not simply a peak square-wave power output—or twice the sine-wave power—but is, instead, a function which will produce greatly differing readings depending upon the method of measurement. Hence manufacturers, with some justification, often base power ratings on the highest voltage obtainable with a peak-reading voltmeter.

The power from the output stage thus obtained is not yet a true measure of the performance of the system. Differences in speaker efficiencies can easily produce loudness differences equivalent to a factor of three in output stage power. A true comparative measurement would have to use the sound pressure level (SPL) produced by the loudspeaker.

For this reason, new "Ampli-vox" equipment is rated in maximum sound pressure level obtainable with pink noise (40 Hz to 8 kHz) input to the amplifier. The amplifier is placed in a reverberant room with a sound-level meter set on the "C" weighting curve. The inputs and controls are adjusted to give the highest reading with the noise input. To convert the reading of the meter to SPL in dB's at the loudspeaker, the reverberation time of the room must be measured using the amplifier for a sound source. From it, the absorption (a) can be calculated. The sound pressure level in dB above 0.0006 dyne/square centimeter is then expressed as follows: $\text{SPL} = \text{meter reading} + 10 \log_{10} a$.

The sound chamber of the mechanical instrument provides a musical effect in addition to coupling to the air. It allows the tone to sustain itself after the string is plucked. Having no sound chamber, the solid-body electronic guitar requires a method for obtaining *legato* effects. If we are avoiding multiple speaker resonances, a wide-range reverberation system provides a ready method.

Harmonic	Frequency (Hz)	Note	Degree	Dissonance Rating
1st	261.5	C	Root	0
2nd	523.3	C^1	Octave or doubled root	0
3rd	784.0	G^1	Doubled fifth	1
4th	1047	C^2	Octave or twice-doubled root	0
5th	1319	E^2	Twice-doubled third	2
6th	1568	G^2	Twice-doubled fifth	1
7th	1830	$A^{\#2}$	Twice-doubled augmented sixth	3
8th	2093	C^3	Octave or thrice-doubled root	0
9th	2350	D^3	Thrice-doubled second	3
10th	2630	E^3	Thrice-doubled third	2
11th	2878	$G^{\flat 3}$	Thrice-doubled diminished fifth	3
12th	3136	G^3	Thrice-doubled fifth	1
13th	3400	$A^{\flat 3}$	Thrice-doubled diminished sixth	3
14th	3663	$A^{\#3}$	Thrice-doubled augmented sixth	3

Table 1. Musical relationships of harmonics in key of C-major. Dissonance rating is based on classic theory of harmony. 0 indicates perfect consonant octave; 1 indicates perfect consonant fifth; 2 indicates imperfect consonance; 3 shows dissonance.

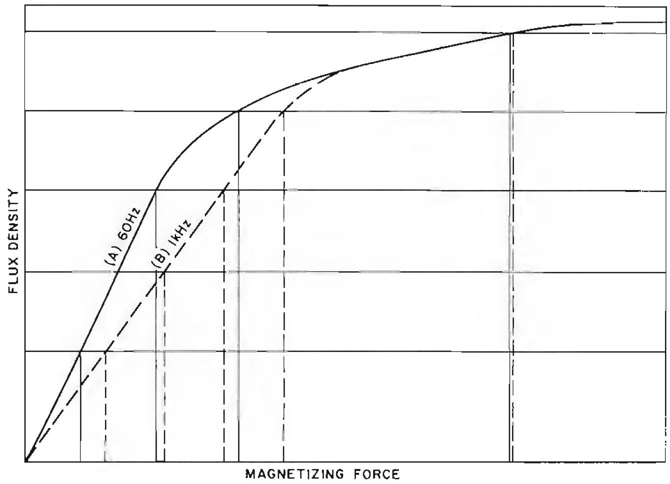


Fig. 9. Transformer core magnetization curves for two frequencies.

Most reverberation systems used in guitar amplifiers merely provide a large concert-hall effect because of the frequency response of the reverberation units. Fortunately, a depth control is included so that a slight amount of this reverberation can be made to sound fairly constant with frequency.

Ideally, the reverberation would vary as the mass of the string being played varies, so that one would have an effect similar to that obtained by the use of the damper pedal on a piano.

Interested in further increasing the versatility of their instruments, guitarists have demanded another form of tone modification—*tremolo* or *vibrato*. Both tremolo and vibrato are modulation of the carrier by a sub-audible frequency, usually in the range of 4 to 12 Hz. The method of modulation varies and will change the character of the tone.

In many cases, vibrato (frequency modulation) is accomplished by means of a lever attached to the tailpiece of the guitar. When this lever is moved back and forth by the player, it changes the pitch of the instrument to produce the desired effect. Tremolo (amplitude modulation) is often produced by means of a low-frequency oscillator in the guitar amplifier that modulates the output signal. Usually, both the frequency and depth of the modulation can be varied by the user. Some more expensive amplifiers use phase modulation in the high frequencies and amplitude modulation in the low frequencies.

Other methods of tone modification are also employed. Tone controls ranging all the way from a single treble boost and cut control to multiple bandpass filters are used. ▲