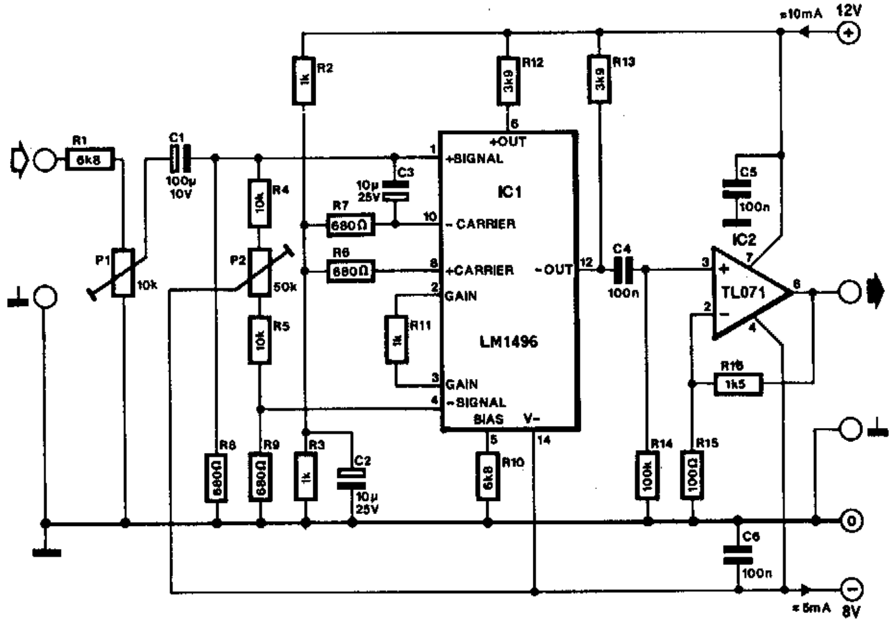
5

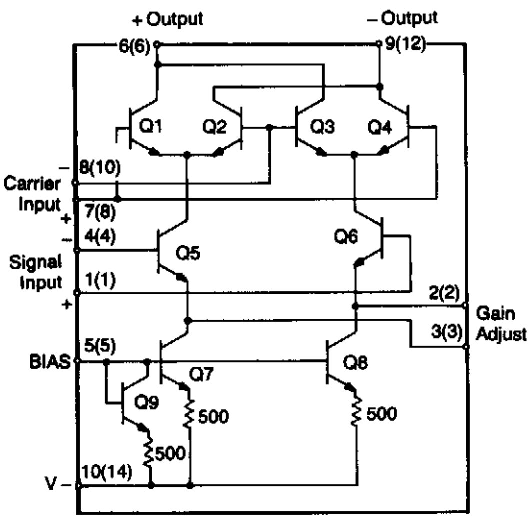
Audio Effects Circuits

The sources of the following circuits are contained in the Sources section, which begins on page 660. The figure number in the box of each circuit correlates to the entry in the Sources section.

AUDIO-FREQUENCY DOUBLER



HANDS-ON ELECTRONICS



Internal circuit of the LM1496.

ELEKTOR ELECTRONICS *Fig. 5-1*

Often the frequency of a signal must be doubled: modulator/demodulator chip LM1496 is an ideal basis for this.

From trigonometry it is well known that:

$$2\sin x\cos x = \sin 2x$$

and:

$$\sin^2 = 1 - x\cos 2x.$$

These equations indicate that the product of two pure sinusoidal signals of the same frequency is one signal of double that frequency. The purity of the original signals is important: composite signals would give rise to all sorts of undesired products.

The LM1496 can only process signals that are not greater than 25 mV: above that level, serious distortion will occur. The design is therefore provided with a potential divider at its input. This addition makes it possible, for instance, to arrange for a 500-mV input signal to result in a signal of only 25 mV at the input of the LM1496.

To provide a sufficiently high output signal, the output of IC1 is magnified by op amp IC2, which is connected as a noninverting amplifier. Because the output of IC1 contains a dc component of about 8 V, the coupling between the two stages must be via a capacitor, C4.

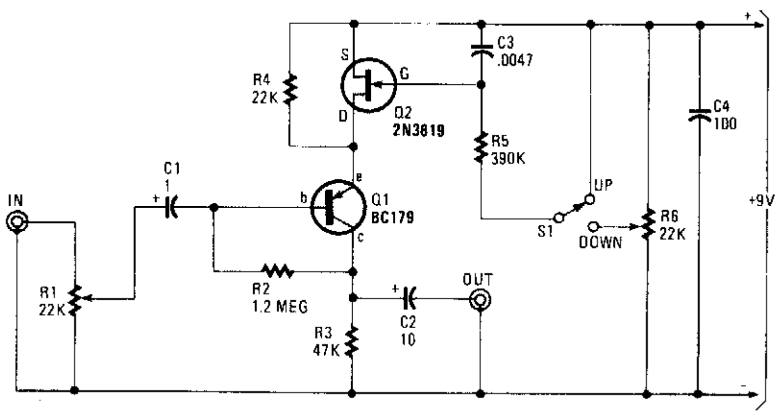
With values of R15 and R16 as shown, IC2 gives an amplification of 16 (24 dB). The overall amplification of the circuit depends on the level of the input signal: with an input of 1.2 V, the amplification is unity; when the input drops to 0.1 V, the amplification is just 0.1. The value of the input resistors has been fixed at 680 Ω: this value gives a reasonable compromise between the requirements for a high input impedance and a low noise level.

To ensure good suppression of the input signal at the output, the voltages at pin 1 and pin 4 of IC1 must be absolutely identical to P4. It is possible, with the aid of a spectrum analyzer, to suppress the fundamental (input) frequency by 60 to 70 dB.

The output signal at pin 12 is distorted easily, because the IC is not really designed for this kind of operation. The distortion depends on the level of the input signal. At a frequency of 1 kHz and an input level of 100 mV, the distortion is about 0.6%; when the input level is raised to 500 mV, the distortion increases to 2.3%, and when the input level is 1 V, the distortion is 6%. The signal-to-noise ratio under these conditions varies between 60 and 80 dB.

The circuit draws a current of 10 mA from the positive supply line and 5 mA from the negative line. The phase shift between the input and output signals is about 45° (output lags). Finally, although the normal output is taken from pin 12, a similar output that is shifted by 180° (with respect to that at pin 12), is available at pin 6.

AUDIO FADER

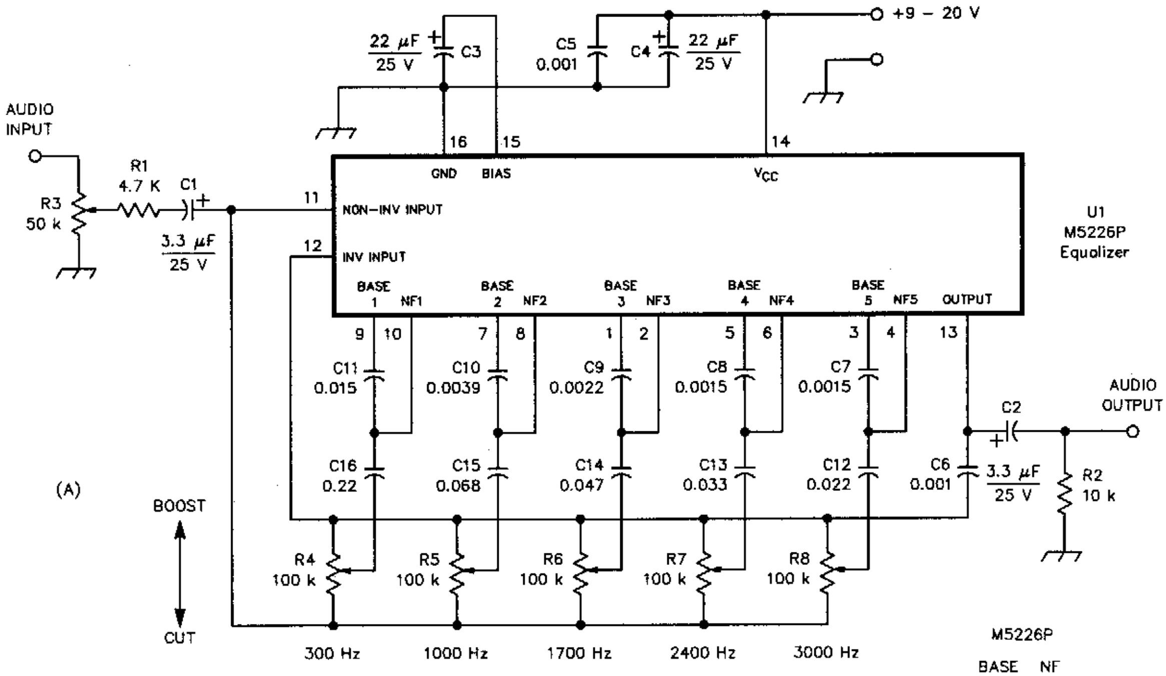


HANDS-ON ELECTRONICS

Fig. 5-2

In this circuit, Q1 is a simple amplifier that has its gain controlled by a variable emitter resistance supplied by FET Q2. In the up position of S1, C3 discharges through R5 and the gain of Q1 decreases because Q2 is driven toward cut-off. In the down position, Q2 conducts more, depending on the setting of R6, which causes a gain increase. By varying R5 or C3, various fade rates can be obtained.

AUDIO EQUALIZER

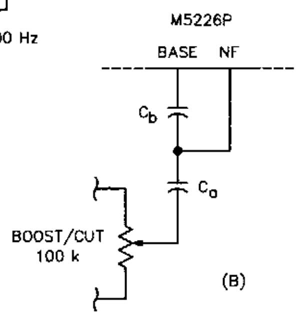


PC–Board Connections

Terminal	Connector Finger
AUDIO INPUT	5, 8, 14, 21
AUDIO OUTPUT	4, 7, 13, 20
+9 – 20 V	2
(ground)	1, 6, 9, 12, 11, 6, 19, 22

Except as indicated, decimal values of capacitance are in microfarads (μF); others are in picofarads (pF); resistances are in ohms; k=1,000.

This schematic represents one of four sections (channels) on the PC board.

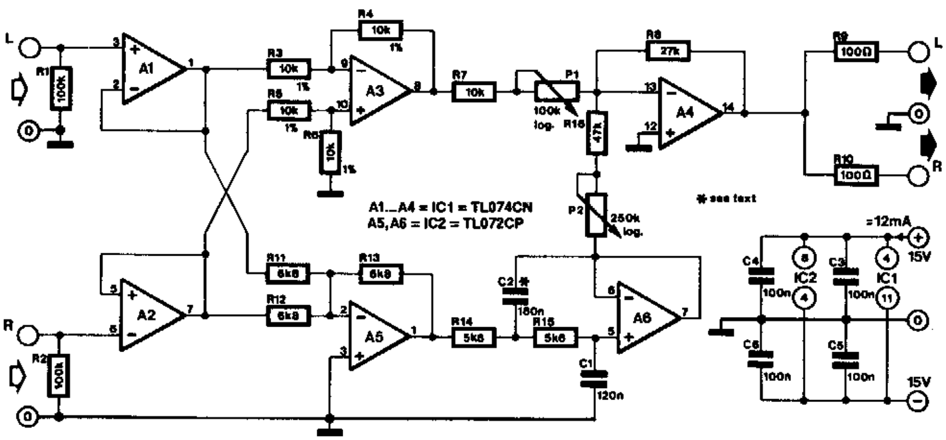


QST

Fig. 5-3

Designed for communications use, this equalizer circuit uses a Mitsubishi M5226P audio equalizer IC to adjust frequency response. It runs from a 9 to 20 V supply. C6 through C16 are polyester film capacitors of ± 5% tolerance.

VOCAL ELIMINATOR



ELEKTOR ELECTRONICS

Fig. 5-4

Otherwise properly mixed sounds often suffer from a predominant solo voice (which might, of course, be the intention). If such a voice needs to be suppressed, the present circuit will do the job admirably.

The circuit is based on the fact that solo voices are invariably situated "at the center" of the stereo recordings that are to be mixed. Thus, voice levels in the left- and right-hand channels are about equal. Arithmetically, therefore, left minus right equals zero; that is, a mono signal without voice.

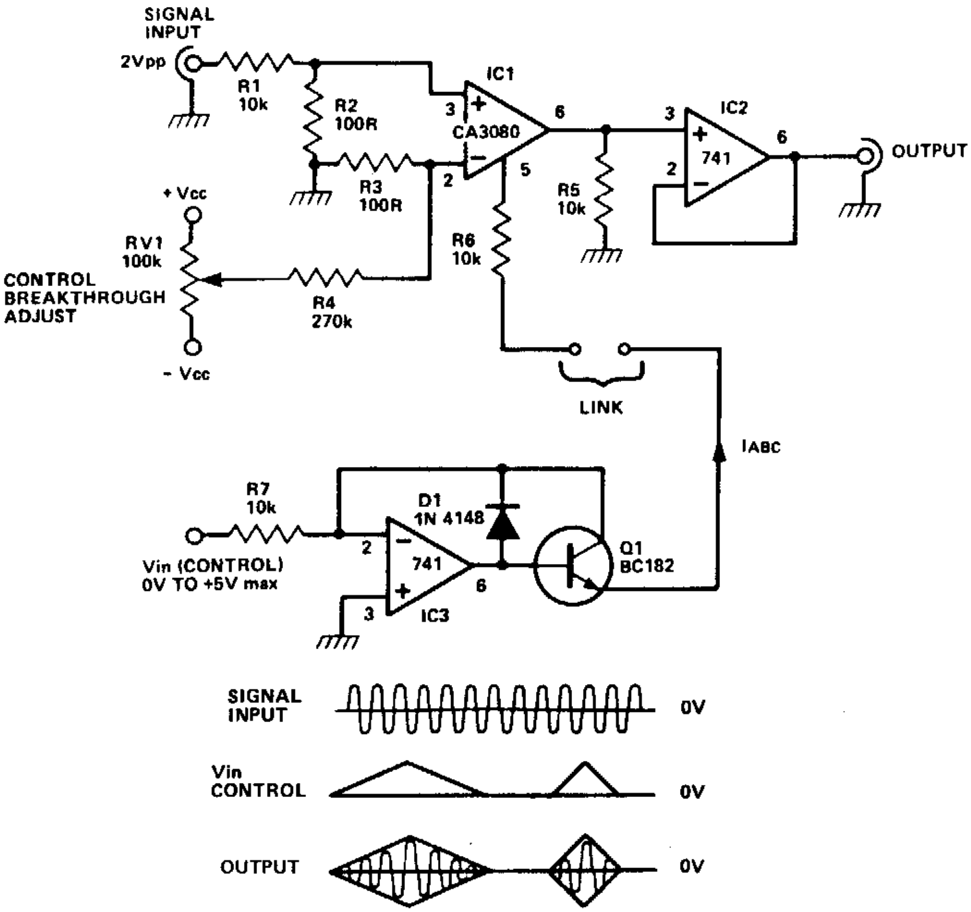
There is, however, a problem: the sound levels of bass instruments, more particularly the double basses, are also just about the same in the two channels. On the one hand low-frequency sounds are virtually nondirectional and on the other hand, the recording engineers purposely use these frequencies to give a balance between the two channels.

However, the bass instruments can be recovered by adding those appearing in the left + right signal to the left – right signal. The whole procedure is easily followed in the circuit diagram. The incoming stereo signal is buffered by A1 and A2. The buffered signal is then fed to differential amplifier A3 and subsequently to summing amplifier A5. The latter is followed by a low-pass filter formed by A6. You can choose between a first-order and a second-order filter by respectively omitting or fitting C2. Listen to what sounds best.

The low-frequency signal and the difference signal are applied to summing amplifier A4. The balance between the two is set by P1 and P2 to individual taste.

You have noticed that the circuit does not contain input or output capacitors. If you wish, output capacitors can be added without detriment. However, adding input capacitors is not advisable, because the consequent phase shift would adversely affect the circuit operation.

VOLTAGE-CONTROLLED AMPLIFIER



ELECTRONICS TODAY INTERNATIONAL

Fig. 5-5

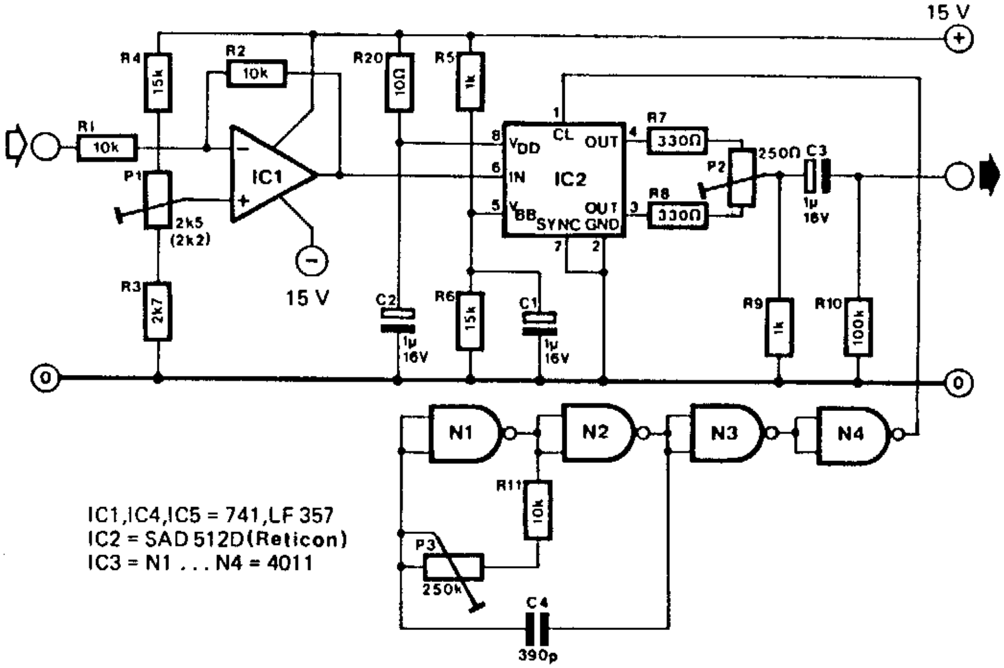
The CA3080 can be used as a gain controlling device. The input signal is attenuated by R1/R2 so that a 20-mVpp signal is applied to the input terminals. If this voltage is much larger, significant distortion will occur at the output. In fact, this distortion is put to good use in the triangle-to-sine wave converter.

The gain of the circuit is controlled by the magnitude of the current IABC. This current flows into the CA3080 at pin 5, which is held at one diode voltage drop above the $-V_{CC}$ rail. The gain of the CA3080 is "linearly" proportional to the magnitude of the IABC current over a range of 0.1 μA to 1 mA. Thus, by controlling IABC, you can control the signal level at the output. The output is a current output, which has to be "dumped" into a resistive load (R5) to produce a voltage output. The output impedance at IC1 pin 6 is 10 kΩ (R5), but this is "unloaded" by the voltage follower (IC2) to produce a low output impedance.

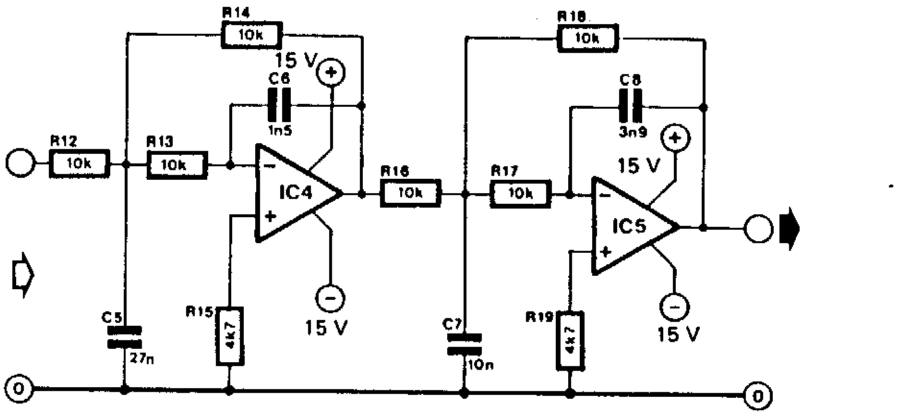
The circuit around IC3 is a precision voltage-to-current converter and this can be used to generate IABC. When V_{in} (control) is positive, it linearly controls the gain of the circuit. When it is negative, IABC is zero and so the gain is zero.

ANALOG DELAY LINE (ECHO AND REVERB)

1



2



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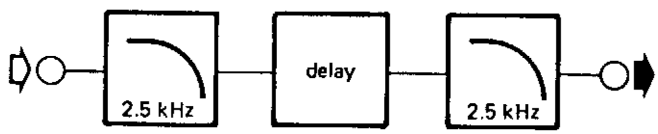
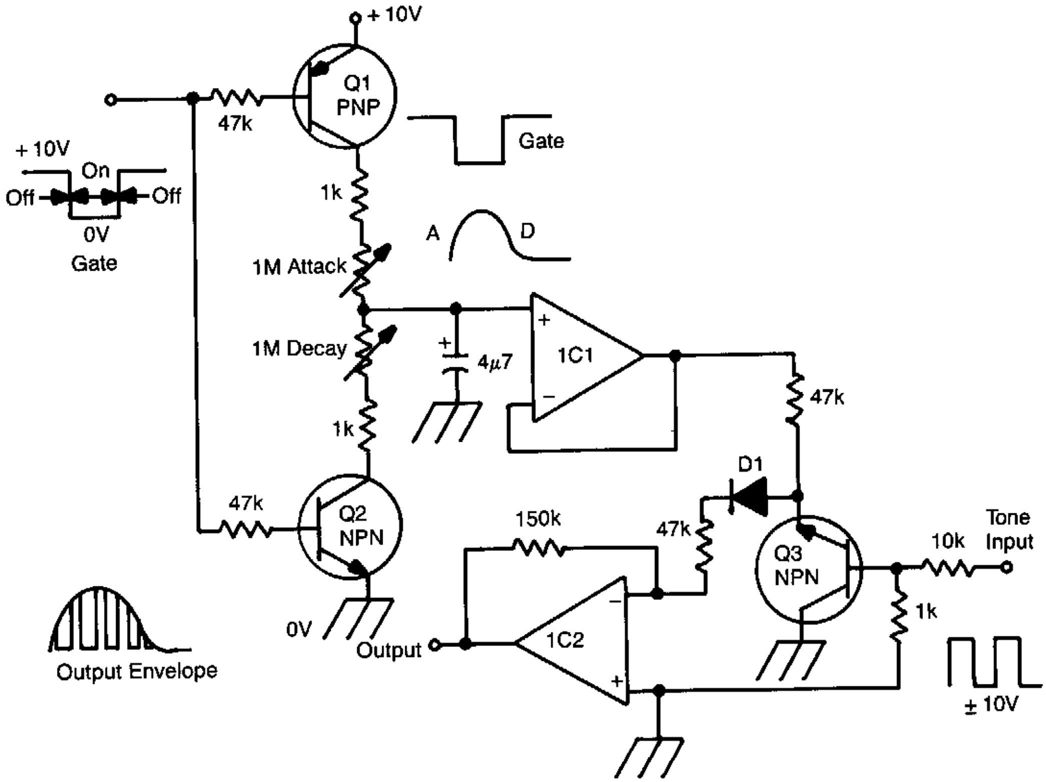


ELEKTOR ELECTRONICS

Fig. 5-6

This circuit uses an SAD 512D (Reticon) chip, which is a 512-stage analog shift register. By varying the clock frequency between 5 and 50 kHz, delay time can be set between 51.2 and 5.12 ms. The clock frequency must be at least twice the highest audio frequency.

MUSICAL ENVELOPE GENERATOR AND MODULATOR



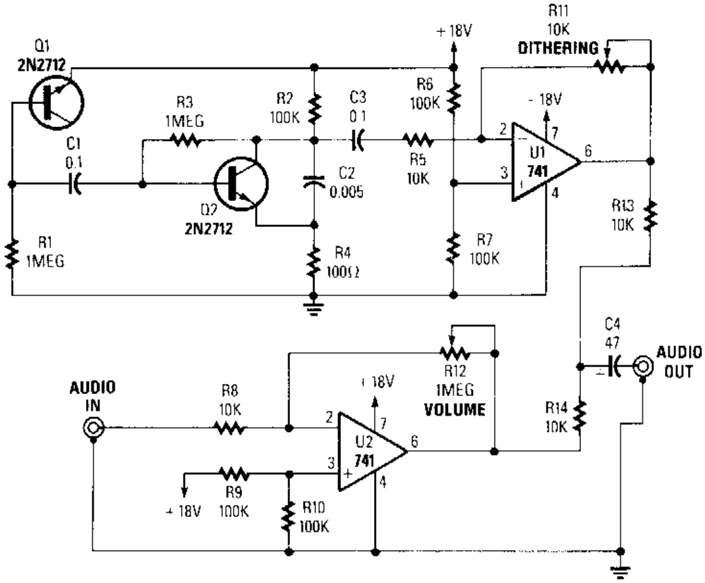
ELECTRONICS TODAY INTERNATIONAL *Fig. 5-7*

A gate voltage is applied to initiate the proceedings. When the gate voltage is in the ON state, Q1 is turned on, and capacitor C is charged up via the attack pot in series with the 1-kΩ resistor. By varying this pot, the attack time constant can be manipulated. A fast attack gives a percussive sound, a slow attack gives the effect of "backward" sounds. When the gate voltage returns to its OFF state, Q2 is turned on and the capacitor is then discharged via the decay pot and the other 1-kΩ resistor to ground. Thus, the decay time constant of the envelope is also variable.

This envelope is buffered by IC1, a high-impedance voltage follower and is applied to Q3, which is being used as a transistor chopper. A musical tone in the form of a square wave is connected to the base of Q3. This turns the transistor on or off. Thus, the envelope is chopped up at regular intervals, which are determined by the pitch of the square wave.

The resultant waveform has the amplitude of the envelope and the harmonic structure of the square wave. IC2 is used as a virtual earth amplifier to buffer the signal and D1 ensures that the envelope dies away at the end of a note.

AUDIO DITHERIZING CIRCUIT FOR DIGITAL AUDIO USE



POPULAR ELECTRONICS

Fig. 5-8

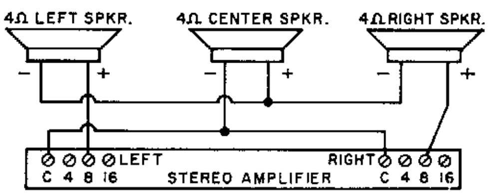
By adding a small amount of noise to a signal to be digitized (about 0.7 bit):

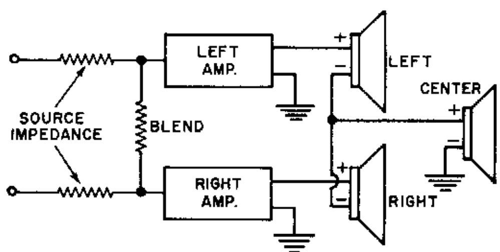
$$V_{\text{NOISE}} \approx 0.7\left(\frac{V_{\text{INPUT}}\text{p-p}}{2^N}\right)$$

where: $n = \#$ of bits. For example, 8 bits and 2 V p-p would be 0.0055 V.

This circuit uses a transistor (Q1) and an amplifier (Q2 and U1) to generate the noise signal. R11 controls the noise injection and R12 controls the gain of the system.

DERIVED CENTER-CHANNEL STEREO SYSTEM





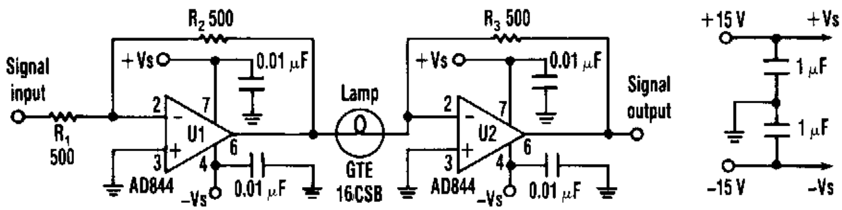
ELECTRONICS WORLD

Fig. 5-9(a)

Fig. 5-9(b)

A simple method of deriving a center or third channel without the use of an extra transformer or amplifier. (a) 4-Ω speakers are connected to 8-Ω amplifier taps. 8 and 16-Ω speakers connect to 16-Ω taps. (b) By blending the inputs it is possible to cancel out undesired crosstalk.

LOW-DISTORTION AMPLIFIER/COMPRESSOR



ELECTRONIC DESIGN

Fig. 5-10

Designers can build a 15-dB compressor with a miniature lamp and a current-feedback amplifier. The circuit possesses extremely low distortion at frequencies above the lamp's thermal time constant. This means that distortion is negligible from audio frequencies to beyond 10 MHz. There's also relatively little change in phase versus gain compared to other automatic gain-control circuits. Lastly, the circuit has many instrumentation, audio, and high-frequency applications as a result of its low distortion and small phase change.

The AD844 op amp is a perfect fit for this application because it's a current-feedback amplifier. Each stage of the circuit, U2, lamp, and feedback resistor compresses an ac signal by over 15 dB (see the figure). Cascading a number of stages delivers higher compression ranges.

Op amp U1 operates as a unity-gain buffer to drive the input to the compressor. However, U1 is optional if a low-impedance signal source is used. The lamp's resistance will increase with temperature, which reduces the ratio of resistor R3 to the resistance of the lamp. This ratio reduces the gain of U2. The lamp's cold resistance should be greater than the input resistance of U2 (more than 50 Ω) for proper operation. The lamp's resistance will change slightly for low input levels. Therefore, the ratio of R3 to the resistance of the lamp and the gain of U2 stays high.