audio compressor

By R. A. Penfold

AN AUDIO compressor is an audio amplifier which is designed to provide a constant output level, from a wide variety of input levels. Thus it is sometimes referred to as a constant volume amplifier. It merely consists of an audio amplifier which is fitted with some form of automatic gain control.

AUDIO COMPRESSION

Reasons for using audio compression vary, as it can be used in several applications. It is often used in tape recording when something such as a debate is to be recorded, and only one microphone is to be used.

The use of compression obviates the need to re-adjust the recording level each time a different person speaks, as, once the level is set for one speaker, the correct modulation depth will be obtained for all the others. This is of course providing that all the speakers are close enough to the microphone, to provide a sufficient output to operate the compressor. This technique also removes the possibility of overmodulation at unexpectedly high volume levels.

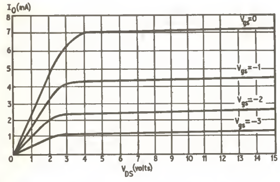


Fig. 1. Drain current plotted against drain to source voltage for a typical *n*-channel f.e.t.

Speech compression is used in some amateur transmitters in order to maintain a high average modulation level, without running the risk of overmodulating an a.m. transmitter, or exceeding the maximum power rating of the power amplifier of an S.S.B. transmitter.

Simple peak clipping circuits are sometimes used instead, but these introduce a comparatively high degreee of distortion, and are not as effective.

USING AN F.E.T.

When subject to a low voltage between the drain, and source terminals, an f.e.t. exhibits the characteristic of an ordinary resistor. This is illustrated in Fig. 1, which shows typical transfer characteristics of an *n*-channel f.e.t., at various gate voltages.

It will be seen that the value of the resistor formed by the f.e.t. can be varied by altering the gate bias voltage. It can be varied from a few hundred ohms to many megohms.

THE CIRCUIT

A circuit diagram of an audio compressor utilising an f.e.t. in a voltage controlled attenuator is shown in Fig. 2. The input impedance to the unit is high (typically 2·5 megohm), and is suitable for use with a crystal microphone. The output is at a low impedance, and will drive virtually any amplifier. For low level inputs (i.e. below the level at which compression begins) a voltage gain of about 275 is available with the gain control at maximum.

In order to obtain the required high input impedance, the input transistor, TR1, is operated in the emitter follower mode. This is direct coupled to TR2, which is a common emitter amplifier.

For TR1 to produce a very high input impedance it must have a fairly high impedance in its emitter circuit. R4 is therefore used to raise the input impedance to TR2, in order to achieve this.

The bootstrapping technique has been employed in order to virtually eliminate the shunting effect the biasing resistors, R1, R2, and R3 would otherwise have on the input impedance. C3 is the bootstrapping capacitor.

Transistors TR1 and TR2 are used mainly as a buffer amplifier, and provide only a small voltage gain.

VOLTAGE CONTROLLED ATTENUATOR

The output from TR2 is fed via C4 to the voltage controlled attenuator. R7 and R8 form a tap on the main supply rail, and produce a suitably low supply voltage for the f.e.t. TR3. The drain to source impedance of TR3, and R9 form an attenuator.

With no negative bias at TR3 gate, the drain to source impedance is very low, and the attenuation factor of the circuit is very low. By giving a negative bias at TR3 gate, the drain to source impedance can be greatly increased, and the attenuation factor of the circuit thus also greatly increased. A voltage controlled attenuator is thus formed.

The output from the attenuator is fed via C5 to the input of a very high gain common emitter amplifier, TR4, which is followed by an emitter follower stage, TR5. From the emitter of TR5, some of the signal is fed via C8 to the volume control, VR1, and then to the output socket. The remainder of the signal is used to produce the biasing voltage for the attenuator.

RECTIFYING CIRCUIT

It is fed via C6, and R14 to a rectifying circuit, consisting of D1 and D2. This arrangement is used as it provides a fast attack speed, but with a long decay. C7 smoothes the a.f. half cycles to a d.c. negative bias, which is then fed to the gate terminal of the f.e.t.

There is not a linear relationship between the gate bias voltage, and the drain to source resistance of the f.e.t. With a low level input, only a small bias voltage will be produced, and this will only alter the value of the resistor formed by TR3 by a few tens of ohms. This will have a negligible effect upon the attenuation factor of the circuit.

Raising the input level slightly will increase the bias voltage, and due to the logarithmic relationship between bias voltage, and drain to source resistance, this will cause a much larger increase in this resistance, say a few hundred ohms. This will result in a noticeable, although still only small increase in the attenuation factor of TR3 and R9.

INPUT LEVEL

It is at this point that raising the input level will begin to have a very noticeable effect on the voltage controlled attenuator, as only a very small change in bias is required to cause an increase of several kilohms in the drain to source resistance of TR3. Thus an increase in the input level causes the gain of the amplifier to drop considerably, and so reduce the output level. The output level will therefore tend to remain almost constant, even though the input level may vary considerably, providing the input is above the level at which compression commences.

Even with quite high input levels (up to about 0·25V r.m.s.) there will be only a small degree of distortion in the circuit. The use of modern silicon transistors in the input stage ensures a low noise level.

TIME CONSTANT

The attack of the a.g.c. circuit is very fast, being virtually instantaneous, but the time constant capacitor, C7, produces a fairly long decay time (about two seconds). For most applications this is very desirable, as it prevents the gain from rising during brief pauses in the signal, and the noise which would subsequently accompany this.

However, the decay time can be altered to suit individual requirements by altering the value of C7, the larger its value, the longer the decay time.

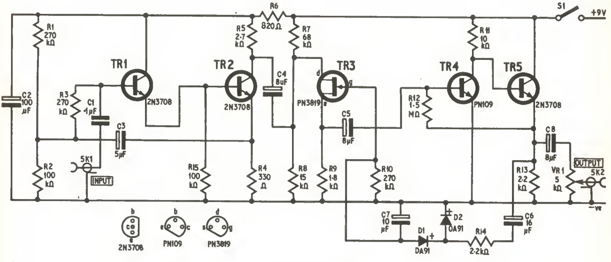


Fig. 2. Circuit diagram of the complete Audio Compressor

audio compressor

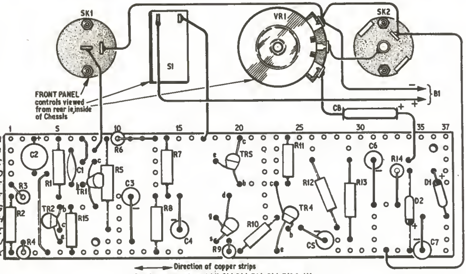


Fig. 3. Layout of the components on the Veroboard panel and interconnections to the other components. Note breaks in copper strips

COMPONENTS . . .

Resistors

R1 270k Ω
R2 100k Ω
R3 270k Ω
R4 330 Ω
R5 2·7k Ω
R6 820 Ω
R7 68k Ω
R8 15k Ω
R9 1·8k Ω
R10 270k Ω
R11 10k Ω
R12 1·5M Ω
R13 2·2k Ω
R14 2·2k Ω
R15 100k Ω

Miniature ¼W, ±10%

Potentiometer

VR1 5k Ω logarithmic

Semiconductors

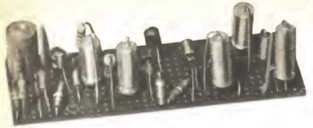
TR1, TR2, TR5 2N3708 (3 off)
TR3 PN3819 } (Arrow Electronics Ltd.)
TR4 PN109 }
D1, D2 OA91 or similar (2 off)

Capacitors

C1 0·1μF disc ceramic
C2 100μF
C3 5μF
C4 8μF
C5 8μF
C6 16μF
C7 10μF
C8 8μF
All 10V electrolytic except C1

Miscellaneous

S1 S.P.S.T. toggle switch.
Vero board panel, 0·1in matric
7in × 4in × 1½in
Aluminium chassis with base plate (H. L. Smith and Co.)
SK1, SK2 phono sockets (2 off)
Control knob, 9V PP6 battery and clips to suit, hardware



CONSTRUCTION

Constructional requirements will vary widely, as some constructors may wish to build the unit as an integral part of some piece of equipment, while others may wish to build it as a self-contained unit, as was the prototype. In either case the Veroboard layout shown in Fig. 3 can be used. 0·1in matrix board is used, and the copper strips run lengthwise. These are cut at a number of points as detailed in the diagram.

A 7in × 4in × 1·5in aluminium chassis fitted with a base plate is used as a case for the prototype. The Veroboard panel is mounted on stand-off insulators in order to hold it a little way clear of the metal case. A PP6 battery is used to power the unit, this particular type being a good fit in the case, and has virtually its shelf life with normal use.

Phono sockets were use for SK1, and SK2 on the prototype, but almost any type of two way socket is of course suitable. Due to the high input impedance of the unit it is essential that the input lead is screened, in order to avoid unwanted noise pick up.

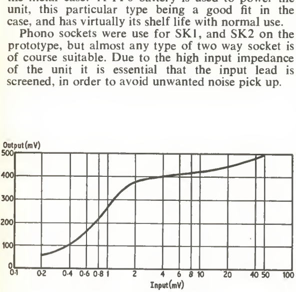


Fig. 4. Graph showing the relationship between the input and output voltages of the Audio Compressor. It can be seen that while the input changes from 2 to 50 millivolts the output only changes by 25 per cent

RESULTS

A graph of the results obtained on the prototype compressor is shown in Fig. 4. This shows input voltage versus output voltage. With an input of 1mV or less the gain is fairly constant at about 275, or a little less. Above this the gain decreases slightly as the input voltage is raised, until it reaches about 2·6mV, and increasing the input voltage above this level has very little effect upon the output. ★