speech processor

To obtain the maximum efficiency when modulating a transmitter the modulation depth must be kept as high as possible for as much of the time as possible. This means that the modulating signal amplitude must be kept reasonably constant. Since speech most definitely does not have a constant amplitude some form of processing is called for.

The most commonly used methods of speech processing are clipping (cutting off the signal peaks) and dynamic compression (reduction of the dynamic range of the signal to achieve a reasonably constant signal level without distorting the waveform). The disadvantage of clipping is that it operates only on the amplitude peaks. It cannot boost a low level signal so that a low modulation depth may still occur. On the other hand, if the signal level is increased so that even low level signals give a reasonable modulation depth, then the peaks will be very severely clipped, resulting in distortion and loss of intelligibility.

Dynamic compression effectively boosts low level signals and cuts high level signals, thus achieving a fairly constant mean signal level. Unfortunately, due to the relatively slow response time of dynamic compressors, transient peaks may not be effectively suppressed, and overmodulation may occur.

The circuit described here overcomes these difficulties by combining both dynamic compression and clipping. The signal is first compressed to achieve a reasonably constant mean signal level, and is then clipped to remove any peaks.

T1 and T2 form the microphone amplifier. The gain of this stage depends on the impedance of the microphone used, so that a high impedance, high output crystal microphone will produce the same sort of output levels as a low impedance, low output dynamic microphone. This avoids large variations in the level of the signal fed to T3 when using different types of microphone.

R5, C5, D1 and D2 form a voltage controlled attenuator. A control voltage is fed back from the emitter of T4 to vary the forward bias voltage on D1. If the base voltage of T4 exceeds the voltage at the anode of D3 by about 0.5 V then the signal fed to the base of T3 is attenuated by R5, C5 and D1. R23 may be switched in or out to vary the compressor time constant. The compressed signal is taken from T3 via C8 and C10. Any peaks are clipped by D6 and D7. The degree of clipping depends on the ratio R8 : R9.

A low-pass filter is provided comprising T5, R17 - R20 and C11 - C14. The values given are suitable for operation in the 80 m band, where from 3 kHz upwards there should be a roll-off of at least 14 dB/octave. For working in other bands where the filter is not required points A and B may be joined and the passive filter components omitted. If a different turnover frequency for the filter is required then the capacitance values C11 - C14 should be multiplied by the factor $\frac{3}{f}$, where f is the required turnover frequency in kHz.

Thus, for a frequency of 6 kHz, the capacitance values would need to be halved.

As a final constructional point it must be stressed that diodes D1 to D7 should be of reputable manufacture. Many of the 'unmarked untested' diodes on the market have a forward voltage drop of up to 1 V, and the circuit will not operate satisfactorily with these. When using the specified 1N4148, the voltage at the anode of D3 should be about 1.5 V to 1.7 V.