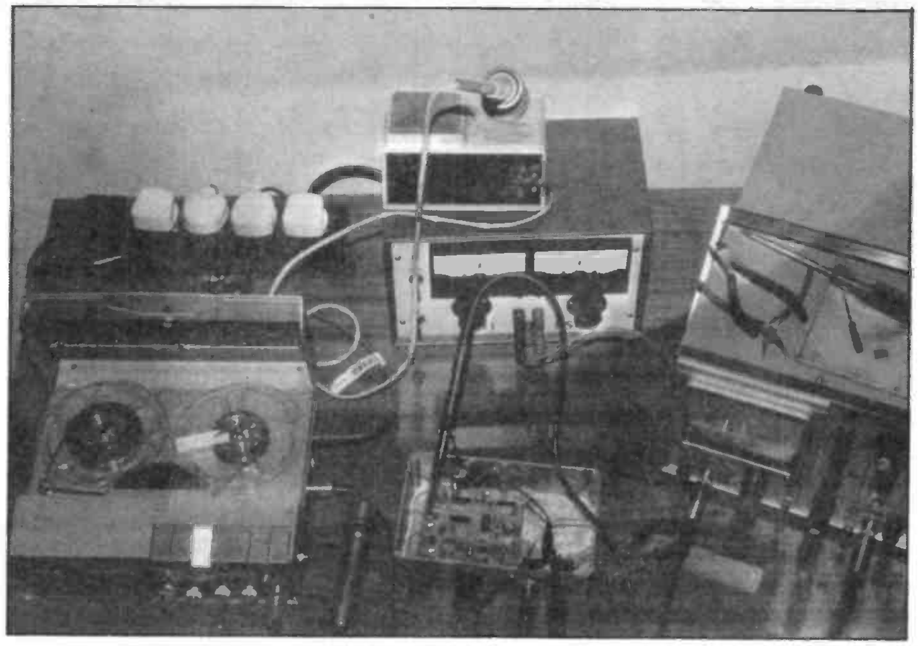
spectrum. Although a bandwidth of 2kHz or so is needed for intelligibility, and about 3kHz for easy recognition of the speaker, most speech energy — the vowel sounds — consists of frequency components below 1kHz. It is the information-carrying, transient, consonents that need the extra bandwidth.

It is possible to improve the intelligibility of speech transmissions by attenuating the low frequency components. This, as we shall see later, also helps reduce distortions produced by processing the amplitude of the speech waveform.

Thus there are two characteristics of speech — amplitude distribution and frequency distribution — that can be altered to improve intelligibility. Good speech processing uses a combination of both techniques.



Test set-up used to get the oscilloscope photo on opposite page

Overmodulation

In practice, any transmitter has some form of protection against overmodulation. (At least it should have.) If the audio level going into the transmitter is too high, then the protection system will be working continuously, by either clipping or compressing the signal. As it is impossible to predict exactly what level your voice is going to peak to, it is inevitable that either the transmitter will be undermodulated, or the protection system will occasionally drop the level.

Compression and clipping, AF & RF

Fig. 1 is the classic textbook graph comparing the four basic methods of processing the amplitude of speech. As it claims to show the precise improvement in intelligibility for a given peak-to-peal level, I can only assume that the graph is based on trials with *tape recordings* of speech. It is only by recording that you can predict the exact peak level of the sample of speech that you use for your experiments.

Real, 'live' speech cannot be predicted so precisely. You have to either undermodulate or compress/clip. In other words, **Fig. 1** should be treated with a king-sized pinch of salt. If AF compression only produces one solitary dB of improvement, ie. hardly any improvement at all, why does every BBC and IBA medium wave transmitter use AF compression?

Amplitude processing

There are two basic methods of improving the average-to-peak ratio of speech — compression and clipping. Both of these methods can be used in either AF or RF stages in the transmitter. I will avoid the term limiting as it means different things to different people — to the RF minded it means clipping, to audio buffs it is a severe form of compression.

Compression is achieved by using the output level of a stage to control its gain (see **Fig. 2**). If the output is too loud the gain is turned down. The output of the rectifier charges up the capacitor in the time constant circuit, which increases the control (sidechain) voltage, reducing the gain of the voltage controlled amplifier.

It is important that the compressor can respond to sudden loud peaks extremely rapidly if a significant 'overload' of the transmitter is to be avoided. This 'attack' time constant needs to be around a millisecond or less. On the other hand, when the gain is recovering during a pause in the speech input, it is important that the gain recovers slowly. If both attack and 'decay' (recovery) had very rapid time constants then the sidechain control voltage would tend to follow the audio waveform. The compressor would then act either as an amplifier with negative feedback, or as a clipper, depending on the 'threshold'. (The threshold of a compressor is, roughly speaking, the level at which gain reduction begins.)

A fast attack and slow decay

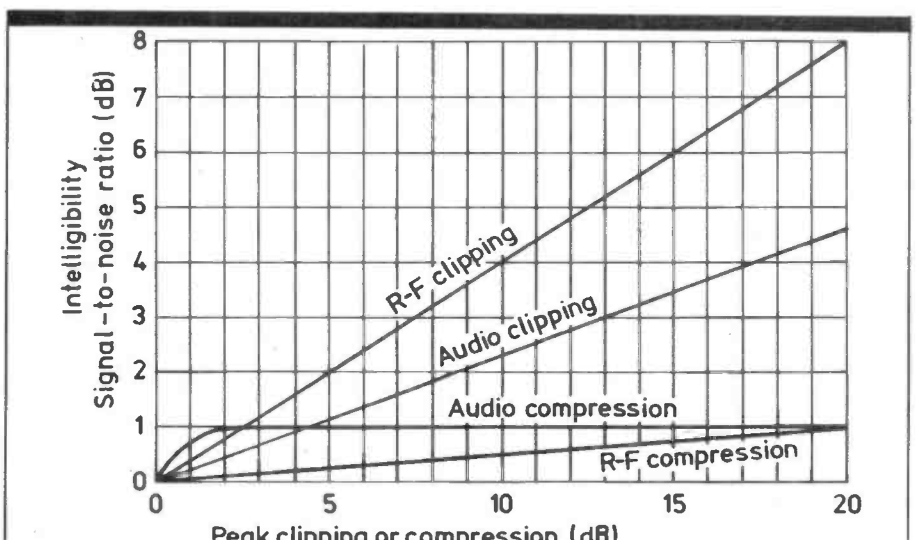


Fig. 1. Theoretical comparison of speech processing techniques often found in the textbooks