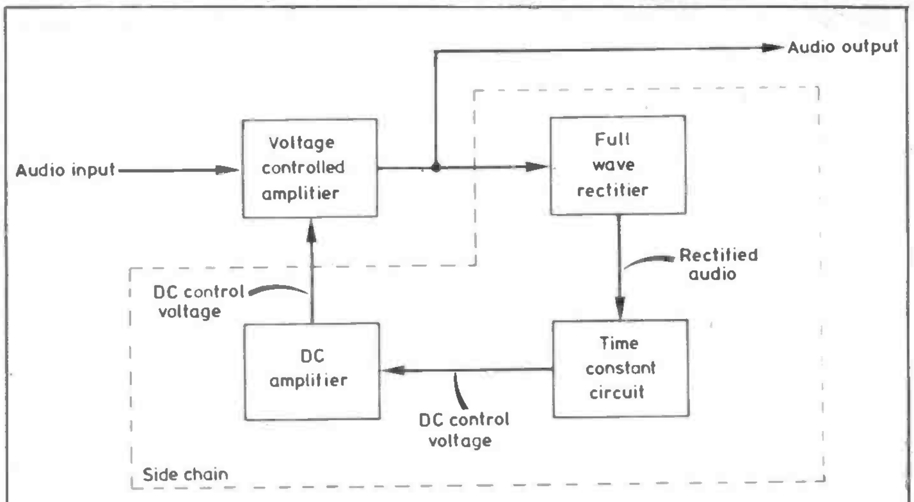

Fig. 2. Block diagram of AF compressor

characteristic can be achieved by charging the time constant capacitor from a low impedance source, and loading it with a high impedance (hence the DC amplifier in Fig. 2). One way of doing this is shown in **Fig. 3**. TR1 is a phase splitter stage that drives emitter followers TR2 and TR3 with audio signals that are equal in amplitude and opposite in phase. These antiphase signals, after having their DC components matched by R7, 8, 9 and 10, and C3 and C4, are rectified by D1 and D2. The DC output is used to charge C5. The source impedance is merely the (low) output impedance of the emitter followers, C3/4 and D1/2. This source impedance, when multiplied by the value of C5, gives the attack time constant. Assuming a very high input impedance for the DC amplifier, the decay time constant is the product of C5 and R11. As a rough guide, a decay time constant of, say, 200mS is about right for speech. (The IBA uses a time constant of about 500mS for medium wave commercial radio — this is a compromise between ideal figures for speech and music. More sophisticated broadcast compressors use a dual time contant that boosts the average level, prevents overloads and at the same time preserves the contrast between loud and soft passages of music!)

Compression can also be done in the RF stages. In this case, the DC control signal is derived from the PA. When it is on the point of being overdriven, the output valves (sorry, yes, valves!) will begin to draw grid current. This will make the grid go negative. This voltage, suitably filtered, can be used to reduce the gain of earlier stages. Such RF compression is normally termed Automatic Level Control, ALC.

The great problem with compression, whether it is done at AF or RF, is that it cannot follow the speech waveform rapidly enough to boost consonants much. It does, however, make sure that the transmitter is fully modulated on speech peaks.

Audio clipping

Audio clipping is just a question of amplifying the whole signal and then lopping off any bits that are too big. See **Fig. 4**. By comparing (a) and (c), it is clear that although the loud peaks have been savaged unmercifully, the low-level, transient sounds are a lot stronger. Note that although the signal has been amplified by a factor of two (6dB), the peak level of the clipped signal is the same as the peak level of the input signal. Thus there is 6dB of clipping.

AF clipping does produce severe harmonic distortion, but there are a number of ways of alleviating the damage. Firstly, by making sure that the clipping is symmetrical — ie. both positive and negative peaks are clipped to the same level — only odd harmonics will be generated. Secondly, careful audio filtering at both the input and the output will reduce the distortion products — more of this later.

6dB of clipping is not a lot. The Voice of America, for example, uses 9dB on its HF broadcasts. The clipping is obvious but not objectionable. Amateur stations tend to use still higher clipping levels without problems.

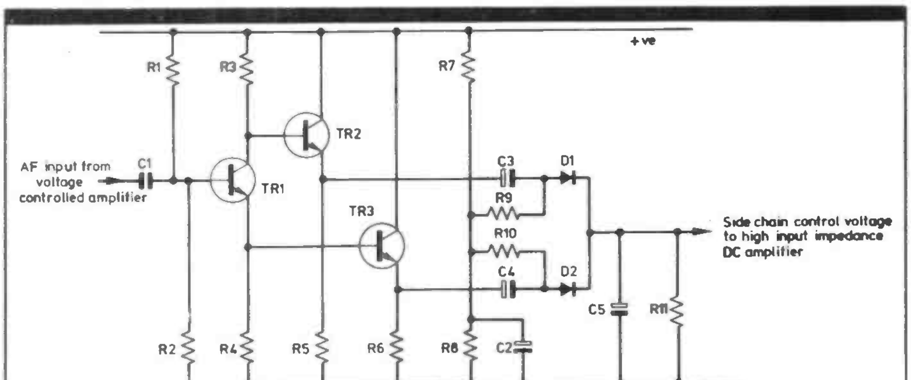

Fig. 3. Circuit for achieving fast attack and slow decay characteristic in AF compressors

The latest RF clipper from Datong. This one features automatic control of the level going into the clipping stage — as does Dave Howes' design in the next article