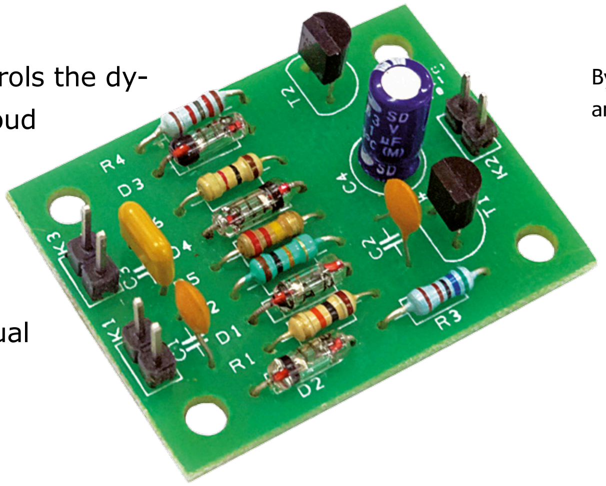
Small Dynamic Compressor

Feed-forward instead of Feedback

This simple compressor circuit controls the dynamics of a speech signal so that loud sounds are controlled to produce a more intelligible voice signal in noisy environments. This is achieved using a novel control mechanism instead of the more usual feedback loop.



By **Rolf Gerstendorf** and **Sunil Malekar**

This circuit uses dynamic feed-forward compression control rather than the more conventional feedback system which regulates the signal using the output signal level. Instead of a control loop this design uses a method of parallel control and despite its relative simplicity it's surprisingly effective. An input signal with a dynamic of approximately 50 dB produces a constant output signal level ±3 dB.

One diode does the trick

The compressor circuit shown in **Figure 1** uses just one active component (transistor T1) in the signal path. The input signal will normally pass unimpeded from the input at K1 over C1, R1, D1, C2 and R2 to the output at K2. An audio signal with a level in the range of 100 mV_{pp} up to 10 V_{pp} passing through the diode/detector made up of D3 and D4 will produce a control voltage for T1. The higher the signal level at the input, the more T1 will be driven into conduction and the greater will be the current through its emitter and diode D1. This causes D1 to conduct more and shunts more of the audio signal to ground. This simple regulation mechanism controls the signal dynamics. The output signal remains in the range of 20 to 50 mV_{pp}.
The germanium diodes D3 and D4 have a relatively low forward-conduction threshold and have an additional voltage bias via T2 and R4. This ensures the detector is effective at low levels of input signal and produces a signal threshold which is practically at zero volts.
The release-time (the control voltage fall-rate time constant) of the control mechanism is governed by the values of C4 and R5. The attack-time in this circuit (unlike feedback

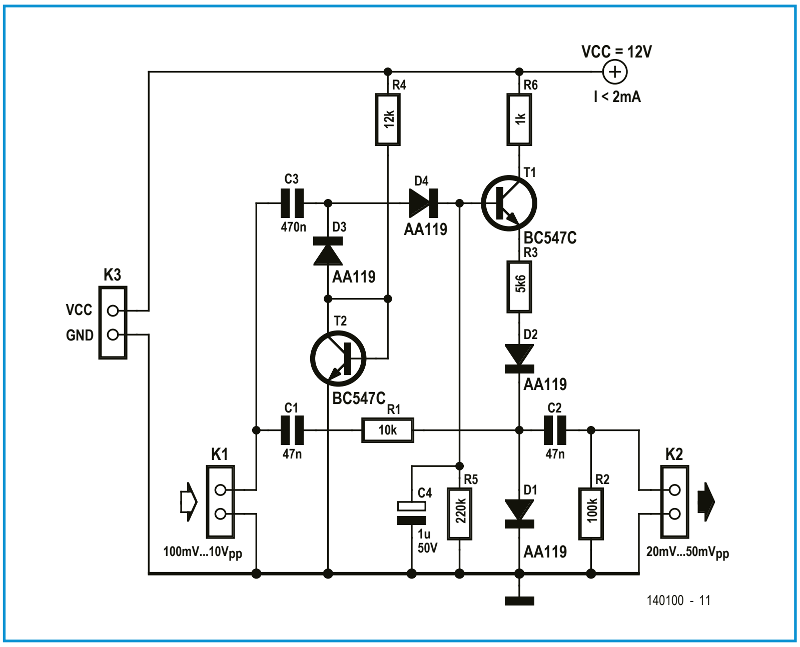


Figure 1.
The feed-forward dynamic compressor circuit.

controlled circuits) is very short and governed by the source impedance of the input signal (which is usually very low) and the value of C4. This characteristic is beneficial because it helps prevent overload distortion that can occur during the attack ramp up time. The circuit can be powered from a 12 V DC power source; it takes less than 2 mA.

The characteristics of diode D1 used to produce the compression are not exactly linear so distortion of a few percent is added to the audio signal. The effect can be seen in **Figure 2** (lower trace = input, upper trace = output). This level of distortion is not acceptable for the reproduction of high quality music but is hardly noticeable on a low quality voice signal. The overall effect of the compressor is to greatly improve speech intelligibility in a noisy communication channel environment. This makes the design particularly suitable for amateur radio applications but not for studio or for live stage performances.

A PCB has been produced for the compressor using the free layout program DesignSpark (**Figure 3**, you can find the layout files at [1]) where all the components neatly fit the board. Although the design doesn't use SMDs, the finished circuit takes up so little space there should be enough room to build it into the equipment or even into the microphone housing used in amateur radio rigs.

(140100)

Web Link

[1] www.elektor-magazine.com/post

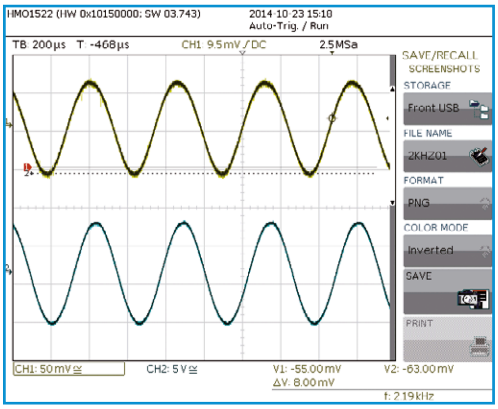


Figure 2.
The output signal (upper trace) shows the effect of compression, the distortion is evident.

Component List

Resistors
R1 = 10kΩ
R2 = 100kΩ
R3 = 5.6kΩ
R4 = 12kΩ
R5 = 220kΩ
R6 = 1 kΩ

Capacitors
C1,C2 = 47nF
C3 = 470nF
C4 = 1µF 50V, pitch 6mm

Semiconductors
D1,D2,D3,D4 = AA119
T1,T2 = BC547B (TO-92)

Miscellaneous
K1,K2,K3 = 2-pin pinheader
PCB 140100-1

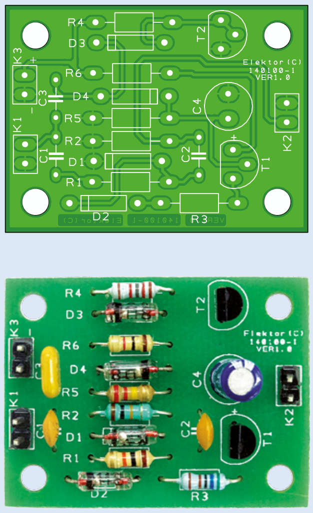


Figure 3.
There should be space inside most equipment to fit the small PCB.