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QST Issue: Dec 1983
Title: Try This Speech "Decompressor"
Author: Eric Nichols, KL7AJ

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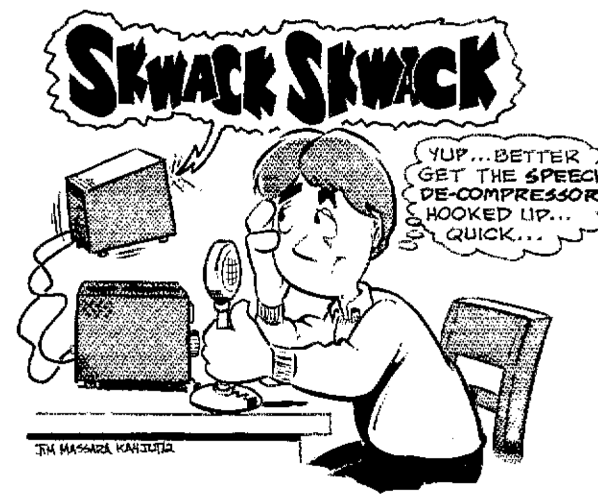
Try This Speech "Decompressor"



Tired of listening to unpleasant or monotonous speech-processed signals? Restoration, in part, can be accomplished at your receiver output with this interesting gadget.

By Eric Nichols,* KL7AJ

Speech processing for a transmitter is probably as good as the circuit design or the operator's ability to adjust the processor correctly. Unfortunately, neither condition is met most of the time. But, even when the overall system is functioning as prescribed, our ears or minds tend to tire of the restricted-speech sound. It's unlikely that we would be so brash as to ask the other station, "Could you please turn off your processor for awhile?" A more practical alternative is to "decompress" the speech at the output of our receiver or transceiver.

It appears that speech processing is here to stay. Practically all new transmitters and transceivers contain processors of some form. Isn't it surprising that no manufacturer incorporates the natural complement to speech compression — audio expansion at the receiver end? Audio expansion is no stranger to the hi-fi enthusiasts among us. These dynamic-range enhancers have been around for years. They compensate for the lack of "head room" on audio recordings. These devices are expensive when designed for music systems, but a simpler and less expensive expander can be devised for speech-only reception.

Here we will consider a simple decompressor which, when connected between a low-level audio output of a receiver or transceiver and an external power amplifier, can effectively restore to the original product even the most "squashed" audio. This project can be assembled in an evening or during a weekend.

Circuit Information

Fig. 1 shows the circuit of the decompressor. We should be aware that all processors — rf or audio — achieve the same end results: to reduce the level of difference between audio peaks and to bring the average power to a higher plateau. This is done with some type of nonlinear amplification.

D1 and D2 of Fig. 1 cause U1 to operate nonlinearly in the opposite sense. As the level into the diodes increases, the amplifier gain also increases. This effect accentuates the difference between the peak voltages. The values for R1 and R2 can be changed to any resistance from 0 to 10 kΩ to tailor the expansion to the amount desired. The values specified in Fig. 1 provide approximately 20 dB of expansion.

Laboratory Tests

I recently demonstrated this compressor at a club meeting. The test system consisted of a closed-circuit audio chain with speech that was prerecorded. The audio was first compressed with 20 dB of instantaneous peak clipping (the most "trashy" method of speech processing). Next, the clipped audio was expanded and fed to a quality audio amplifier; the difference between the original and the expanded audio (after clip-

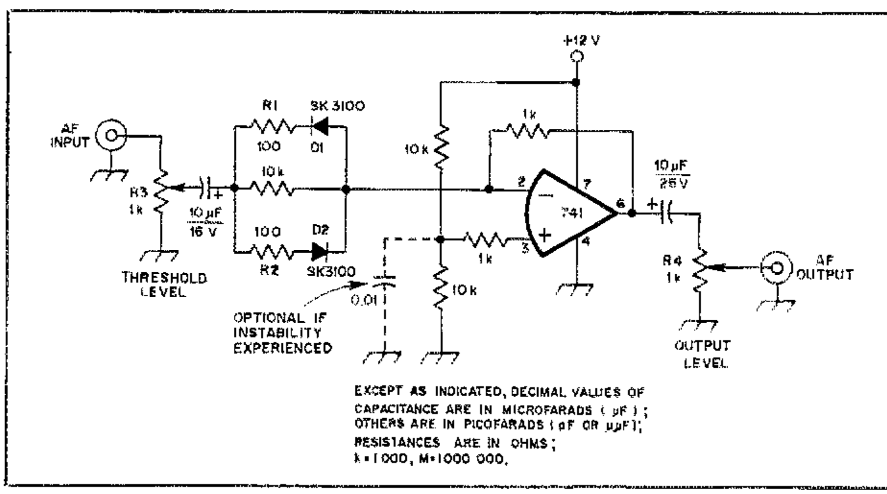


Fig. 1 — Schematic diagram of the speech decompressor. Resistors are ¼- or ½-W carbon composition. V_{cc} can be from 9 to 13.5, positive. R3 and R4 are linear-taper carbon controls. D1 and D2 are small-signal silicon diodes (1N914 or equiv.).

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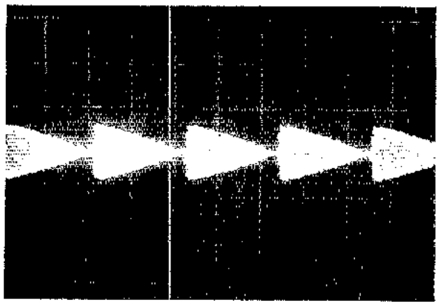

Fig. 2 — Original audio wave form without processing or expansion, as viewed on a scope.

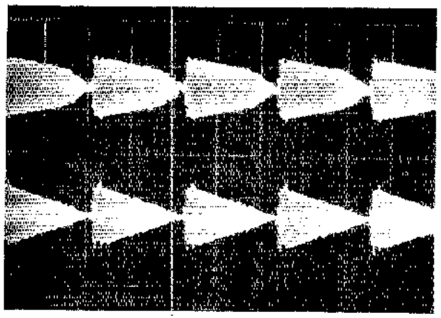

Fig. 3 — The upper trace shows the audio energy after being processed. Processed audio, recovered then expanded by the circuit of Fig. 1, is shown in the lower trace. The ratio between the peaks and ebbs of the audio signal is more pronounced in the lower waveform example.

ping had taken place) was indistinguishable to anyone in the room. Figs. 2 and 3 contain oscillographs of the unprocessed audio, compressed audio and expanded audio energy.

The circuit of Fig. 1 has an unexpected bonus. It reduces the background noise, particularly the high-frequency hiss, considerably. The speech information seems to jump right out of a noiseless background. This greatly enhances the listening quality.

Adjustments

The expander input level should be low. It can be taken from a low-level output point in the receiver. The 600-ohm phone-patch output on most transceivers is ideal for the take-off terminal.

First, we need to set the output level for a fairly high amount. Next, we advance the threshold-control setting until background noise can just be heard. Then the control is backed off just below this point. Now we can listen with pleasure to those hams who don't know that their processors have an "off switch" at our end of the line!

The decompressor has what might be considered a disadvantage: It requires an outboard audio amplifier. If you don't care for a collection of outboard equipment, you may want to try my "Mickey Mouse" way of achieving the necessary result. This method works only with solid-state rigs that contain complementary-symmetry output stages. It will not work with rigs that use conventional Class A output amplifiers.

Most of today's rigs have an af-output stage that is similar to the circuit of Fig. 4. D1 and D2 provide a slight forward bias for Q1 and Q2 to reduce crossover distortion. Sometimes a resistor is used in place of D1. All we need to do in order to introduce crossover distortion intentionally is to change the bias on Q1 and Q2 of Fig. 4. The crossover distortion provides the same results as is created by D1 and D2 of Fig. 1, surprisingly. If there are two such diodes in your rig, you may jumper one of them (as in Fig. 4), and that will do the

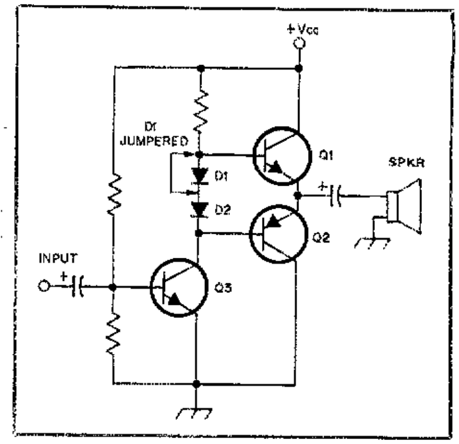

Fig. 4 — Typical circuit of a complementary-symmetry audio channel found in a modern receiver or transceiver (see text).

trick. As an alternative, you may replace the silicon diodes with germanium units, which have roughly half the forward-conduction voltage characteristic of silicon diodes. If a resistor is presently in use at D1 of Fig. 4, you may need to reduce the value of the resistor to approximately half the present value. Do not replace both diodes with resistors, because at least one diode is necessary to ensure thermal stability.

Final Comments

The human voice is a great thing to hear. The state of the art seems to require that the voice quality be "butchered." So, let's use the available technology to increase the effectiveness of our amateur communications. But let's also try to recover the pleasant character of the human voice.

Strays

Gary Firtick, K1EB, is one of about 25 members of the Hen House Gang in Bethlehem, Connecticut, who will be operating W1FHP until January 7, 1984, looking to contact hams in as many towns of Bethlehem across the country as they can. This will be their 28th Christmas season operation. See Special Events, Nov. 1983 *QST*, for more details. *(photo by KA1YP)*

CALL FOR PAPERS

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Deadlines are January 31, 1984, for the notice of intent, and March 31, for camera-ready manuscripts. Author's kits will be mailed by February 29. — *Jens Zander, SM5HEV*