

split-band speech processor

Design and construction
details of a split-band
audio speech processor
that features up to
15 dB of clipping
and low distortion

Speech processing, especially for SSB, can be a relatively inexpensive means of improving the effective "talk power" of a voice modulated transmitter. Much has been written about various devices and methods that can be used to gain this increase in effective talk power. The devices used have ranged from simple audio compressors to rf envelope clipper-filters. All of these devices attempt to reduce the peak-to-average ratio of the speech or rf waveforms, thereby overcoming the peak power limitations of the transmitter. Generally, the degree of improvement is proportional to the complexity of the processing method; the simpler circuits offer minimal improvement while the more complex effect substantial improvement.

This article will not attempt to present all the theory involved in speech processing; however, the interested reader is referred to excellent articles by Fisk,^{1,2} Kirkwood,³ Moxon,⁴ and Schreuer⁵ for more detailed overviews of the subject.

Until recently, rf envelope clipping has generally been accepted as the most effective SSB processing method. Distortion products are small, generally consisting only of intermodulation products. The primary disadvantage of rf processing is the circuit complexity involved, and the necessity of modifying the associated transmitter. When modifying the transmitter is out of the question, a processor using the audio-SSB-audio (Comdel) approach can be used. In this method, an SSB signal is generated, peak limited (clipped), filtered, and then demodulated back to an audio signal which then modulates the transmitter.

My initial efforts were directed toward designing and building a unit of this type. A breadboard model was constructed and evaluated under laboratory conditions. Performance was very good, and distortion was held to under 10 per cent at 20 dB of clipping. The circuit was, however, excessively complex. It required an audio preamplifier, two balanced modulators, an oscillator, a clipper, an rf amplifier, and an expensive mechanical or crystal filter.

By Wes Stewart, N7WS, 1801 East Canada Street, Tucson, Arizona 85706

At this point, Jim Metzger, W7TKR, suggested that I try the split-band approach. He had done some work with the process with considerable success and Fisk² had written in glowing terms about a similar unit available commercially from Maximilian Associates. This was inducement enough to build a bread-board model for evaluation.

basic circuit

Fig. 1 is a simplified block diagram of the split band clipper. The input signal is applied to an agc-controlled preamplifier which then drives the first set of bandpass filters (BPFs). The filters split the audio spectrum into four narrow bands which are then clipped and directed into the second set of BPFs, where the harmonics generated by the clipping process are filtered off. These filtered signals then go to the combiner stage where they are reassembled into the desired output.

Input amplifier. The design of the input amplifier is not particularly critical. The gain required will depend on the output amplitude of the source, the gain (if any) of the BPFs, and the limiting threshold of the clipper stages. If a very low output microphone is used, low noise may be of some importance. If, as in my case, active bandpass filters are used, the amplifier will also have to exhibit low output impedance. Automatic gain control is also desirable, as it helps maintain a high average clipping level, which in turn insures maximum talk power improvement.

Bandpass filters. As pointed out by Fisk, the optimum design for BPFs is a compromise between several conflicting requirements. Overshoot or ringing due to the near squarewave input from the clipper must be minimized, skirt selectivity should be good, and phase shift through the passband must be smooth and predictable. The latter point becomes important when the design of the combiner is considered, as will be seen later. Other very important factors to be considered are circuit complexity and reproducibility.

After pondering all of the above points, I decided on a two-pole Butterworth active filter. The Butter-

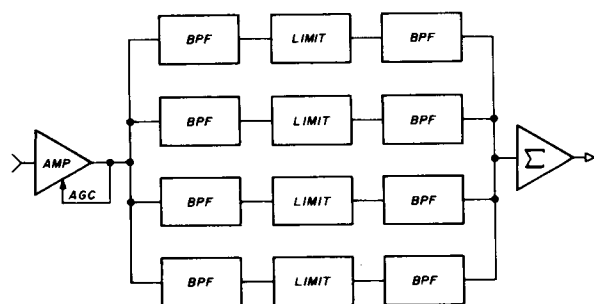


fig. 1. Block diagram of a split-band audio speech processor.

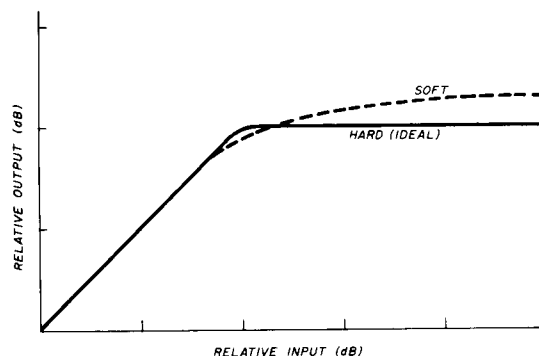


fig. 2. Comparison of "hard" vs "soft" limiting. Soft limiting is undesirable because of the uncertainty of the threshold point, making it hard to maintain constant output from the processor.

worth is not optimum when considering only impulse response and phase shift; however, when used in a low- Q configuration, it is a good compromise between filters with these attributes and those possessing superior skirt selectivity.

The final circuit is configured as a multiple-feedback type.⁶ These filters are relatively insensitive to component variations, allowing the use of 5 per cent tolerance components and inexpensive operational amplifiers. Detailed design data for the selection of center frequency, gain, and Q will be given later.

Peak clipper. The clipper may seem to be one of the least critical parts of the circuit, but, in fact, its requirements are quite stringent. One of the most important factors in the performance of the clipper is that of clipping symmetry. Perfect symmetry insures that only odd harmonics are generated; second-order products would be too much for the two-pole filters to handle. An important point is that the only place clipping should occur is in the clipper. Clipping or limiting elsewhere in the circuit cannot be easily controlled and must be avoided. This may seem easy to do, but if the clipping threshold is too high, limiting may occur in a preceding stage when large amounts of clipping are in use. For example, if a clipping threshold of one volt is used and 20 dB of peak clipping is desired, the preceding stage must be able to have an output voltage swing of 20 volts peak-to-peak. If this stage is running off a single 12-volt power supply, this will of course be impossible.

Another important aspect is that of how "hard" the limiting is. Many of the circuits initially examined, which included limiting differential amplifiers, shunt-diode clippers, and operational amplifiers with shunt diode feedback, had rather "soft" limiting characteristics. That is, the threshold was ill-defined and the slope of the transfer function continued to change over a wide range of input levels. **Fig. 2** graphically shows the difference between hard and soft limiting.

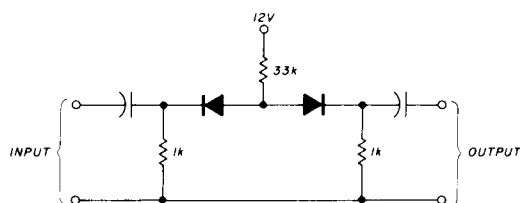


fig. 3. Circuit diagram of the amplitude limiter (clipper) used in the final design. With the resistor values shown, the output will be limited to approximately 300 mV p-p.

Soft limiting is undesirable because it makes it difficult to maintain a constant peak output level.

The circuit finally selected for this application, as best satisfying the above requirements as well as using a minimum of parts, is shown in fig. 3. This will be recognized as a variation of the old series automatic noise limiter used in receivers. By suitable selection of resistor values and bias voltage, the clipping threshold may be adjusted over a wide range.

The performance of this circuit is demonstrated in fig. 4. This is a multiple-exposure oscilloscope photograph taken of the output of the clipper. The inner, near sinusoidal, trace was obtained by increasing the input signal until a 3-dB increase caused only a 2-dB change in output. This point was defined as the clipping threshold. The middle trace represents a further input increase of 4 dB, and the outermost trace was obtained with a total input overdrive of 15 dB. The photograph shows the nearly flat peak output and the exceptional symmetry. A further test of symmetry was made by examining the frequency spectrum of the clipper output with a Hewlett-Packard 302A wave analyzer. With 15 dB of clipping, the second harmonic remained more than 40 dB below the fundamental output.

Combiner. The combiner has the job of taking the four BPF outputs and putting them back together again while maintaining their original phase relationships. Improper phasing will result in excessive passband ripple being generated. As described by Fisk, the Maximilian unit incorporates phase shift networks before the combiner to compensate for the phase shifts through the BPFs. As will be shown later, these networks can be eliminated by the judicious selection of filter characteristics and the use of a simple summing and differencing amplifier.

circuit description

Fig. 5 is the complete schematic of the system. The input is applied to Q1, an FET source follower, used to match high impedance microphones. The follower output drives U1, a Plessey SL1626 gain-controlled amplifier. This IC maintains a nearly constant output of slightly less than 100 mV RMS over an input range of 1 to 100 mV.

The SL1626 is used as recommended by the data sheet, except for the addition of R6 and C10, which are necessary to suppress a high-frequency oscillation. R4 lowers the sensitivity about 20 dB and may be unnecessary in some applications. Front panel adjustment of the clipping level is possible via R7.

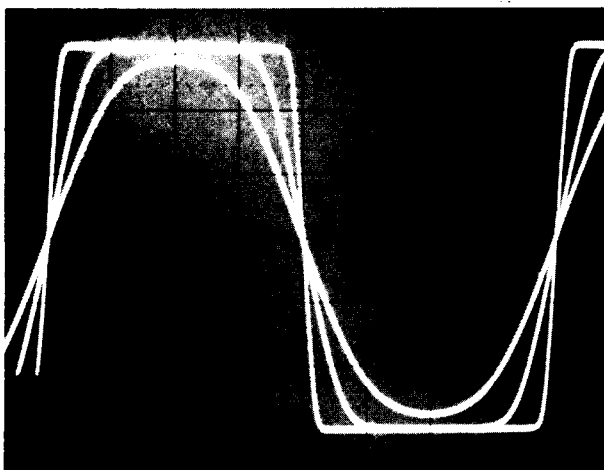
Amplifier U2A, one section of an LM324, develops a small amount of additional gain and serves as a low-impedance source to the following BPFs. The resistors used on the outputs of all the LM324s are necessary to eliminate cross-over distortion.⁷

All of the bandpass filters are operated at the same gain and Q ; only the center frequency (f_o) differs from channel to channel. For simplicity, all capacitors are of the same value; the center frequency is adjusted by choice of resistor values. Using the given values, the overall frequency response will be approximately 350 to 3000 Hz at -6 dB, with no greater than 3 dB of passband ripple. If other cutoff frequencies are desired, appendix 1 gives the equations necessary to calculate new values of f_o and Q . Appendix 2 gives the equations for calculating the parts values for the individual filters.

The clipping stages, as described earlier, use a pair of forward-biased diodes. With the bias resistor values shown, the clipped output will limit at about 300 mV p-p. The shunt-bias resistor values are kept low enough to insure that the input impedances of the second BPFs remain fairly constant even when the clipping diodes turn off.

The second set of BPFs are identical to the first. Their outputs are combined in another section of an LM324, which delivers the system output through a resistive divider. By adjusting the resistor values, the output amplitude can be set approximately the same as that of the microphone, allowing the clipper to be

fig. 4. Performance of the clipper stage shown in fig. 3. The sinusoidal trace was made at the threshold point (1-dB clipping). The middle trace shows 5 dB of clipping, and the outer trace was made with 15 dB of clipping. The vertical sensitivity is 50 mV/div.



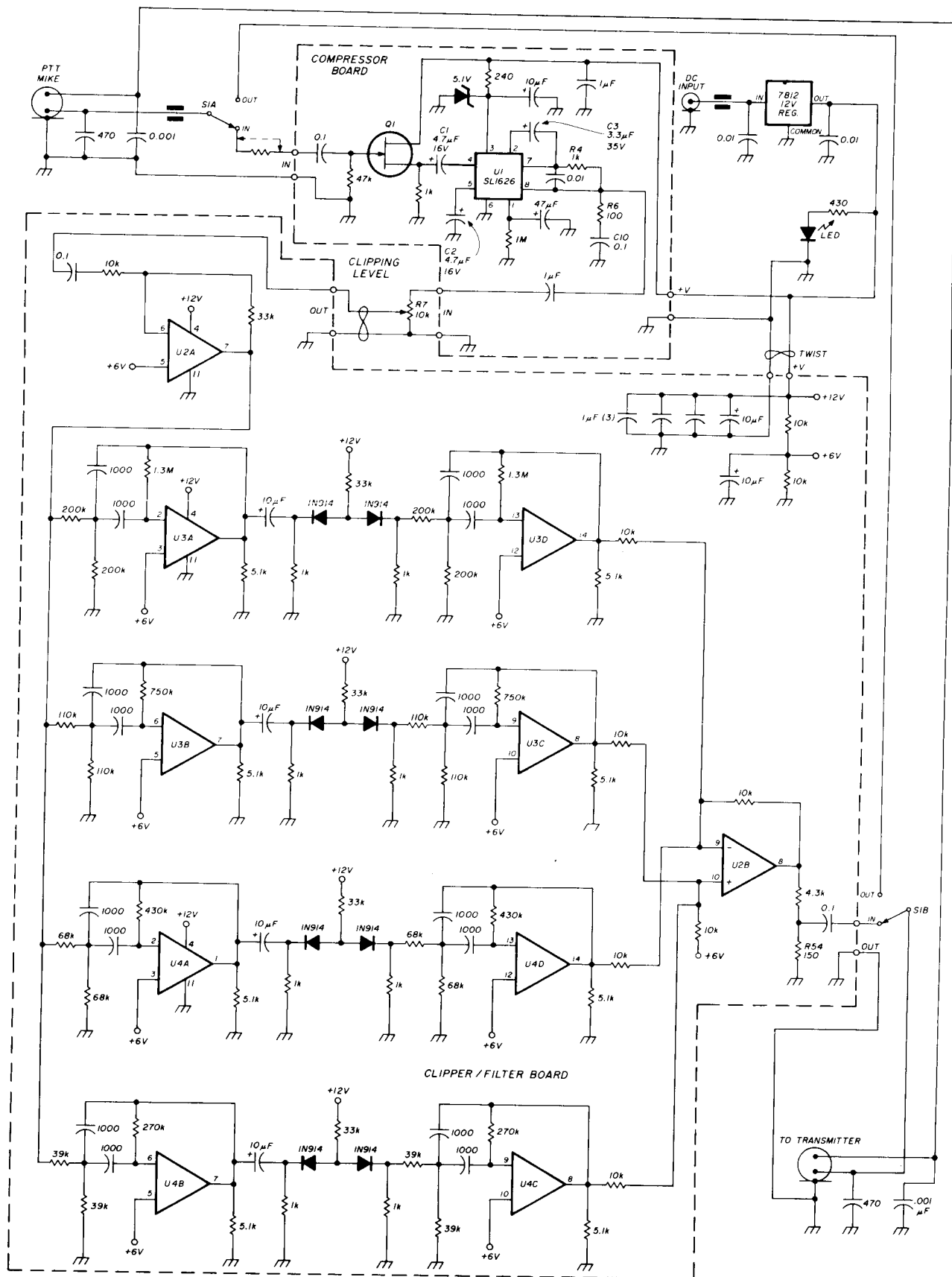


fig. 5. Complete schematic diagram of the split-band audio processor. Q1 is a 2N4392 or equivalent. U2, U3, and U4 are LM324s. C1 and C2 are dipped tantalum capacitors (RS 272-1409). C3 is also a dipped tantalum (RS 272-1408). All other polarized capacitors are tubular tantalums or electrolytics. The remaining capacitors are ceramics, with the exception of the 1000-pF capacitors, which are 5 per cent dipped micas. All resistors are 1/4-watt, 5 per cent, carbon composition.

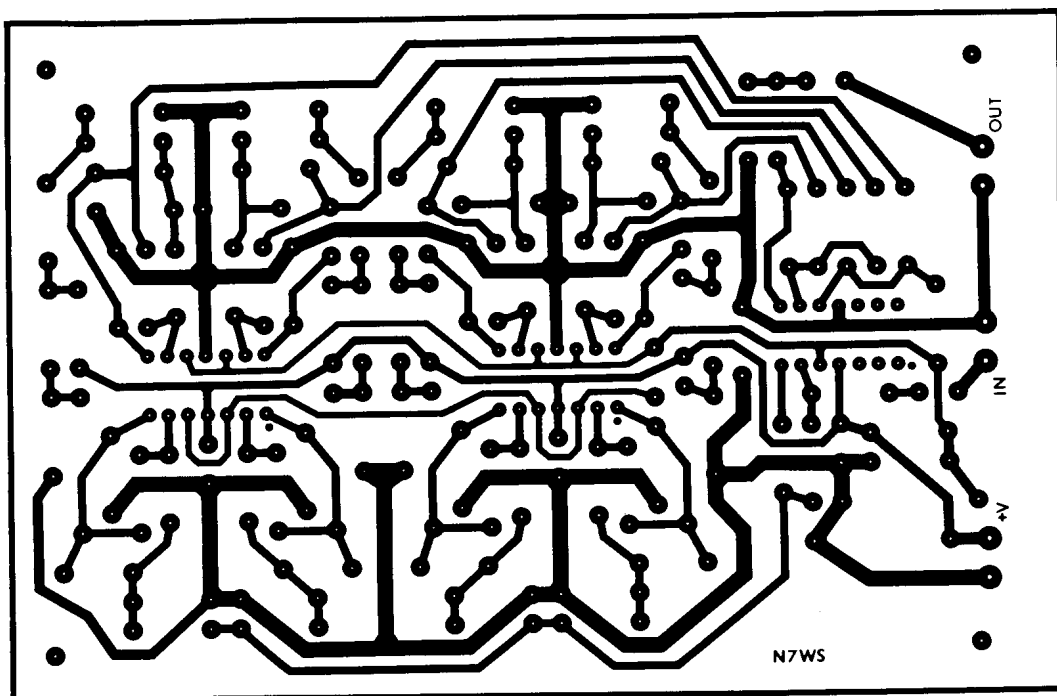


fig. 6. Full-size printed-circuit layout for the audio clipper board.

switched in and out without a gain change in the transmitter. Note that the BPF outputs are alternately connected to the plus and minus inputs of the combiner. The next section will demonstrate why this is done.

Assuming the equations shown in the appendix were used to determine the f_o and Q of the individual filters, adjacent filters will share a common -3 dB

frequency. Eq. 1 demonstrates how an input signal at this frequency is shifted $+45$ degrees in one channel and -45 degrees in the other:

$$\theta = 90 - \arctan \left(\frac{2Qf}{f_o} + \sqrt{4Q^2 - 1} \right) - \arctan \left(\frac{2Qf}{f_o} - \sqrt{4Q^2 - 1} \right) \quad (1)$$

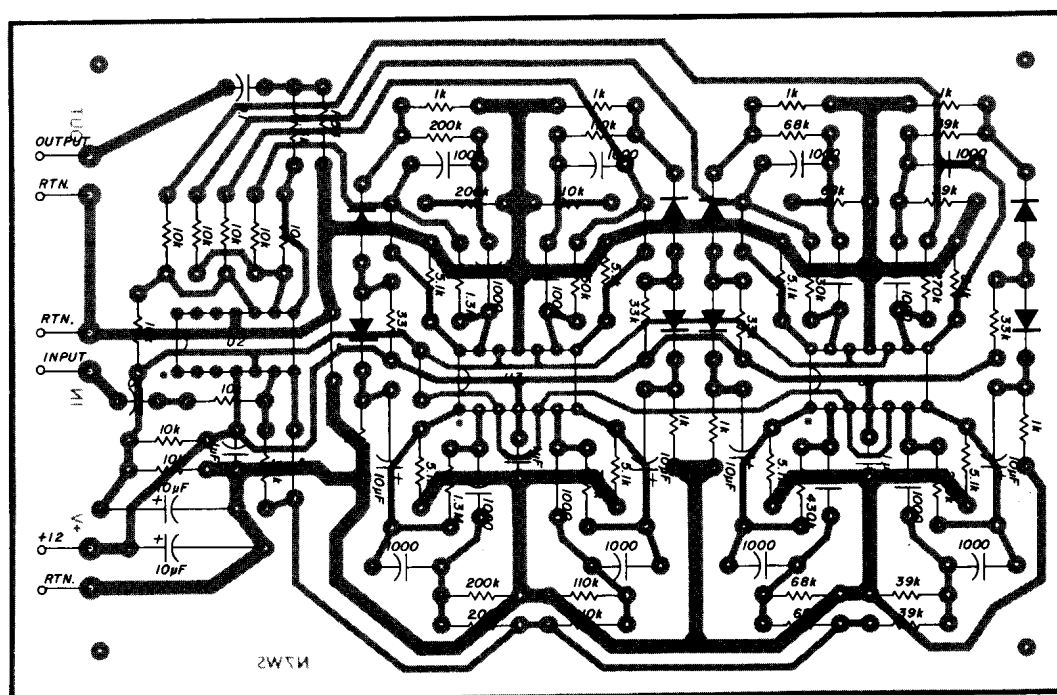


fig. 7. Component location on the audio clipper board.

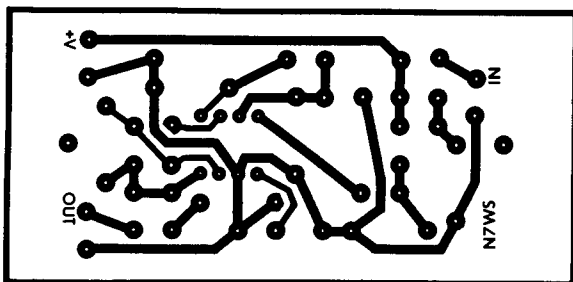


fig. 8. Full-size circuit board layout for the compressor board of the split-band processor.

where θ is the phase shift in degrees
 f_o is the filter center frequency
 f is the frequency of interest

After cascading the two filters in each channel, this shift will be doubled to ± 90 degrees. Clearly, if these two signals are vectorially added, their sum will be zero because they are of equal amplitude but 180 degrees out of phase. A simple solution to this problem is to invert the phase of one signal. This is effectively what is done by the combiner.

Solving eq. 1 for other frequencies will yield a phase error that increases with distance from the -3 dB point. This error is less important, however, because the amplitude difference also increases, so the larger signal dominates when the summation is made.

construction

For added versatility, the circuit is constructed on two etched circuit boards; the input compressor on one, the clipper-filter on another. This allows either one to be used alone in other applications. Figs. 6 and 8 are full-size layouts of the foil sides of the two

fig. 10. View of the prototype split-band speech processor. The circuit boards are mounted using metal spacers and machine screws. Room is available for mounting an ac power supply; an external supply was used for this model.

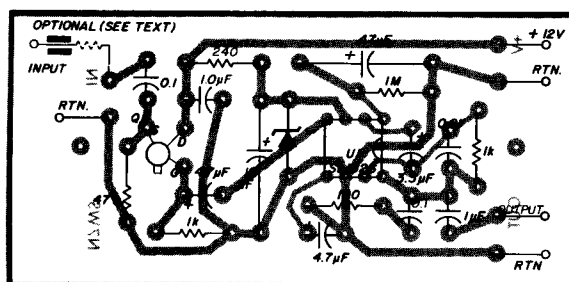
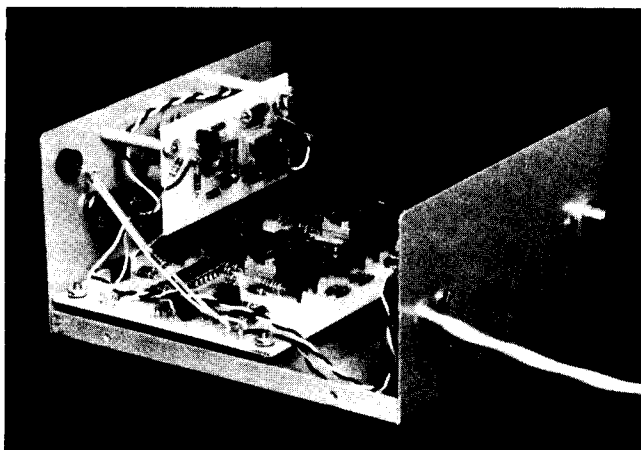


fig. 9. Parts placement diagram for the compressor circuit board.

boards, while figs. 7 and 9 show the component placement. These boards have been laid out with considerable attention to preventing ground loops. A hand-wired board should be built with the same attention.

The prototype shown in fig. 10 was constructed in a Radio Shack enclosure (270-253). Sufficient space remains for the inclusion of an ac-operated power supply. Fig. 11 is a schematic diagram of a suitable supply. Liberal use of ferrite beads and bypass capacitors on all leads entering the enclosure eliminates any chance of problems with rf interference.

performance

As fig. 12 shows, the frequency response is very close to what was calculated, despite the use of 5 per cent components. By adjusting R7, the clipping level can be varied from 0 to 15 dB. Greater amounts of clipping can be had by increasing the gain of either U2 or the BPFs, or reducing the clipping stage bias to lower the clipping threshold.

Caution should be exercised before deciding on greater amounts of clipping, however. This could turn out to be too much of a good thing. Increased clipping does continue to reduce the peak-to-average ratio, but at the same time distortion increases rapidly. This is shown graphically in fig. 13. As pointed out by Moxon,⁴ most of the improvement is obtained by the first 6 dB, with little to be gained by increased amounts. My on-the-air tests seem to indicate that 10 to 12 dB is about optimum with this system. All of this is rather subjective, but the whole topic of speech intelligibility and recognition is pretty subjective, so take it for whatever it's worth.

Total harmonic distortion was measured with an HP 331A distortion analyzer at various frequencies and clipping levels. The results of these measurements are shown graphically in fig. 13. As the figure indicates, distortion begins to rise rapidly as the clipping level approaches 15 dB.

These measurements were of necessity made with single frequency inputs which represent worst-case conditions. Because clipping is occurring on every

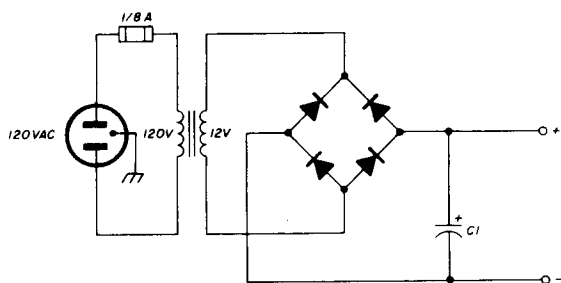


fig. 11. Schematic diagram of an ac power supply suitable for use with the processor. The transformer can be a Radio Shack 273-1385, the diode bridge a 276-1151, and C1 either 272-1019 or 272-1032.

half cycle, harmonic generation is maximum. With speech, clipping occurs much more randomly, with proportionally less total distortion.

On-the-air tests have been extremely gratifying. Reports have indicated substantial increases in apparent signal strength without noticeable distortion or loss of naturalness as long as the clipping level was held around the 10- to 12-dB point. Some loss of naturalness seems to occur above this point, but up to 15 dB, the sound is still not too objectionable. No tests have been run at levels in excess of 15 dB.

operation

Operation is very simple. The agc amplifier holds the clipping level constant, relaxing the operator requirements considerably. Some adjustment of the input sensitivity may be necessary if the microphone used has either a very high or very low output. While the dynamic range of the compressor will handle a higher input, the rise in background noise between speech pauses will be annoying to the listener. In this case, a series resistor may be added to the input which, in combination with R1, forms an attenuator. In the case of a very low-output microphone, increasing the value of R4 will increase the sensitivity. Highest gain occurs with R4 omitted entirely.

On the output side, changing the value of R54 will control the maximum output level. This interacts with the audio gain control on the transmitter, so

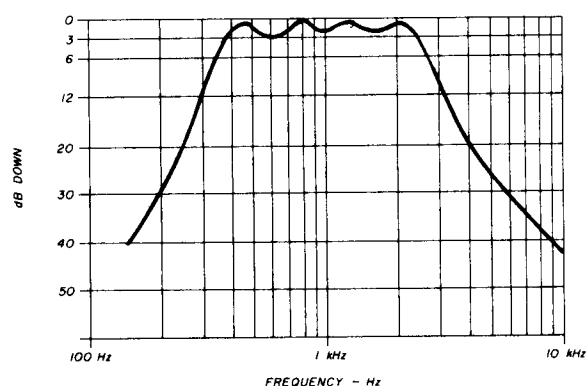


fig. 12. Measured audio response of the speech processor.

corrections can be made either place. I tried to pick a value that allowed the clipper to be switched in and out without having to readjust the microphone gain each time.

Finding the best setting for the microphone gain is best done with the aid of an oscilloscope on the transmitter output. With the clipping level set to maximum, adjust the transmitter gain so the peak output just approaches the level achieved with full carrier or excitation. If no oscilloscope is available, I find that just whistling into the microphone and setting the gain to the point that just activates the transmitter ALC works out very well. If you are not going to use the maximum amount of clipping available, then do the adjusting at the clipping level you intend to use. Even the best of clippers will not maintain a completely flat output vs input characteristic. Therefore, if you adjust your gain at 15 dB of clipping, then reduce it to 10 dB, your peak output will drop a little.

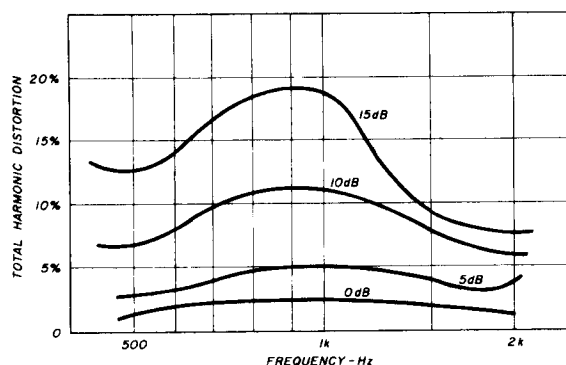


fig. 13. Total harmonic distortion vs clipping level. These curves were made with single tone inputs. Average distortion with speech input should be lower.

This effect can be explained as follows: As shown in fig. 4, sine waves subjected to 15 dB or so of clipping take on the appearance of pretty good square waves. As mathematical analysis can show, a square wave is composed of a fundamental frequency and all of its odd harmonics. We try to filter out these harmonics and retain only the fundamental. Unfortunately, the peak amplitude of this fundamental component is larger than the peak amplitude of the square wave by a factor of $\frac{4}{\pi}$, or 2.1 dB.⁸ It is this factor that causes a continuing increase in output despite the use of a "perfect" limiter.

I want to express my thanks to Jim Metzger, W7TKR, for his technical advice, to Frank Baker for his circuit-board layout genius, and to Don Scheick and Norm Keopfer for their assistance in the preparation of the circuit boards. Additional thanks go to the many others who offered advice and encouragement, to Norma Putney for the typing of the manuscript, and to my wife, Terry, for the many hours spent away from family affairs during this project.

appendix 1

For new passband limits, the values for Q and f_o can be found as in the following example:

1. Define the low frequency – 6 dB point, f_L (350 Hz)
2. Define the upper frequency – 6 dB point, f_H (3000 Hz)
3. Find the multiplying coefficient, L

$$L^4 = \frac{f_H}{f_L} = \frac{3000}{350} = 8.571$$

$$L = \sqrt[4]{8.571} = 1.711$$

4. Find the individual filter cutoff frequencies

$$f_L = 350 \text{ Hz}$$

$$Lf_L = 599 \text{ Hz}$$

$$L^2 f_L = 1025 \text{ Hz}$$

$$L^3 f_L = 1753 \text{ Hz}$$

$$L^4 f_L = 3000 \text{ Hz}$$

5. Find the individual center frequencies

$$f_{o1} = \sqrt{(350)(599)} = 458 \text{ Hz}$$

$$f_{o2} = \sqrt{(599)(1025)} = 784 \text{ Hz}$$

$$f_{o3} = \sqrt{(1025)(1753)} = 1340 \text{ Hz}$$

$$f_{o4} = \sqrt{(1753)(3000)} = 2293 \text{ Hz}$$

6. Determine required Q

$$Q = \frac{f_o}{BW}$$

$$Q_1 = \frac{458}{249} = 1.839$$

$$Q_2 = \frac{784}{426} = 1.840$$

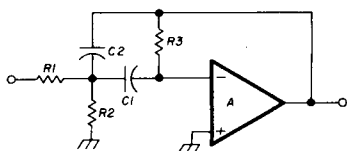
$$Q_3 = \frac{1340}{728} = 1.841$$

$$Q_4 = \frac{2293}{1247} = 1.839$$

Use $Q = 1.84$

appendix 2

The multiple feedback bandpass filter shown below may be designed by the following method (example in brackets):



Choose: $C = C_1 = C_2$

$$[C = 1000 \text{ pF} = 10^{-9} \text{ F}]$$

Let: $H = \frac{|A_o|}{Q}$

$$[H = \frac{3.39}{1.84} = 1.84]$$

where A_o = desired gain

$$Q = Q \text{ from appendix 1}$$

Calculate: $K = 2\pi f_o C$

$$[2\pi \cdot 458 \cdot 10^{-9} = 2.878 \cdot 10^{-6}]$$

$$R_1 = \frac{1}{HK}$$

$$\left[\frac{1}{1.84 \cdot 2.878 \cdot 10^{-6}} = 188.8k \right]$$

$$R_2 = \frac{1}{K(2Q - H)}$$

$$[2.878 \cdot 10^{-6} (1.84) = 188.8k]$$

$$R_3 = \frac{2Q}{K}$$

$$\left[\frac{3.68}{2.878 \cdot 10^{-6}} = 1.28M \right]$$

This completes the calculations; the final step is to select the nearest 5 per cent standard resistor values. If, as in the above example, A_o equals Q^2 , R_1 will equal R_2 , which minimizes errors due to tolerance variations.

The following program, written for an HP 25 calculator, will speed the design of the BPF:

HP-25 Program Form

Title Multiple Feedback Bandpass Filter

Switch to PRGM mode, press \square [PRGM], then key in the program.

DISPLAY	KEY ENTRY	X	Y	Z	T	COMMENTS	REGISTERS
00							R0 C
01	31	+				enter f_o	
02	02	2					
03	61	X					R1 R1
04	15 73	gT					
05	61	X					
06	24 00	RCL 0					R2 R2
07	61	X				defines K	
08	23 06	STO 6					
09	24 05	RCL 5					R3 R3
10	15 03	gABS					
11	24 04	RCL 4					
12	71	÷					R4 Q
13	23 07	STO 7					
14	24 06	RCL 6					
15	61	X				defines R1	R5 A
16	15 22	g 1/x					
17	23 01	STO 1					
18	24 04	RCL 4					R6 K
19	02	2					
20	61	X					R7 H
21	24 07	RCL 7					
22	41	-					
23	24 06	RCL 6				defines R2	
24	61	X					
25	15 22	g 1/x					
26	23 02	STO 2					
27	24 04	RCL 4					
28	02	2					
29	61	X					
30	24 06	RCL 6				defines R3	
31	71	÷					
32	23 03	STO 3					
33	13 00	GTO 00					
34							
35							
36							
37							
38							
39							
40							
41							
42							
43							
44							
45							
46							
47							
48							
49							

references

1. James R. Fisk, W1DTY, "Speech Processing in Amateur Equipment," *ham radio*, June, 1968, page 60.
2. James R. Fisk, W1DTY, "Novel Audio Speech Processing Technique," *ham radio*, June, 1976, page 30.
3. Barry Kirkwood, ZL1BN, "Principles of Speech Processing," *ham radio*, February, 1975, page 28.
4. Leslie A. Moxon, G6XN, "Performance of RF Speech Clippers," *ham radio*, November, 1972, page 26.
5. Walter Schreuer, K1YZW, "Speech Clipping in Single-Sideband Equipment," *ham radio*, February, 1971, page 22.
6. *Handbook of Operational Amplifier Active RC Networks*, 1st edition, Burr-Brown Research Corporation, Tucson, Arizona, 1966, pages 27-34, 78, and 79.
7. *Linear Databook*, 1978 edition, National Semiconductor Corporation, Santa Clara, California.
8. Rudolf F. Graf, *Electronic Databook*, 2nd edition, Van Nostrand Reinhold Company, New York, 1974, page 209.

ham radio