Tales of Speech Processing

—including a practical design

Tolerating the screamers and whisperers.

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Conversation overheard on 20 meter band, SSB: "Old man, I'd like you to give me a report—I want to switch in my processor and see what it sounds like. . ."

"OK, switch it on. You're about 5 and 9 now."

"★'#"&'#★'&2&?"

". . . Ah. . . Yeah. . . Ah. . . Sounds pretty good. . . Really brought my S-meter up. But I think I missed the question . . . Try me again."

Anyone who works even a little SSB regularly has heard that conversation, usually many times. At the same time, we are all familiar with the low duty cycle characteristics of human speech. This attribute of speech has led to many schemes, some wilder than others, but all aiming to improve information transfer by speech. And listening on the bands tells one that some of the more elaborate designs can sound as awful as some of the more rinky-dink ones.

A short history of speech processing is probably in order. The basic character of speech has been known since at least the advent of the oscilloscope; and in the old AM days, several transmitters (Heath/Johnson/others) incorporated speech clipping followed by a suitable filter. The reason for the filter was obvious: when the top is lopped off a signal, harmonics are generated, increasing the modulation bandwidth and causing a fuzzy sound in the recovered audio. Some of these clipper/filters were very simple and straightforward and some of them sounded very good, with a tremendous improvement in intelligibility; some of them sounded awful.

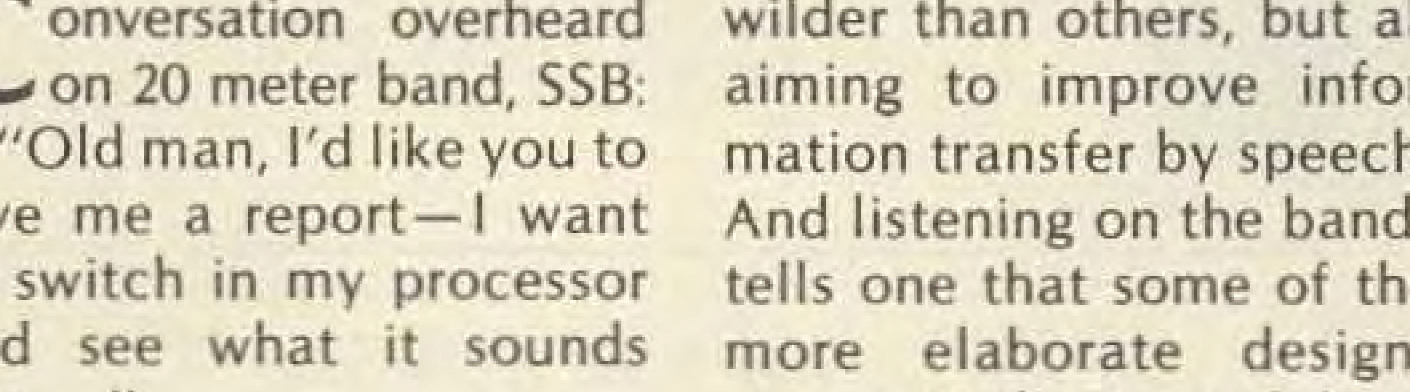

Fig. 1. Demonstration clipper/filter.

Then SSB came along, and at first it sounded awful enough to the AMers without complicating the whole thing with speech clipping/processing. In fact, in the great SSB vs. DSB controversy of the 1950s, reported in the proceedings of the IRE and other journals, it was alleged that one of the problems of the then "new" SSB was that it didn't lend itself to simple speech processing. This attitude persisted for many years, even though some unreconstructed mavericks were using speech clippers of one kind or another on SSB, and they could see a difference on the plate current meter. Some of them neglected to mention to their contacts that they were using clippers. Possibly there were some guilt feelings, especially after hearing conversations such as the one above.

A hairy mathematical proof made the rounds and found its way into the *Handbook* (ARRL). It demonstrated to everyone who had been through first year trig that clipping at audio for SSB was wrongheaded and possibly dangerous. It had terms like $Sin^{n}X$, where n was between zero and one. Oh, it was wonderful! Mathematicians rejoiced at the elegance of it.

There appeared to be one unwarranted assumption, however, and that was that the operator would attempt to modulate an SSB transmitter with these (nearly) square-topped waveforms. And as the argument proved, you can't reproduce square waves directly using

SSB. Neglected was the fact that most operators would have used a filter after the clipper which would have rounded the sharp square edges by removing the harmonic energy.

Most of us are aware of the fact that a square wave is composed of a fundamental frequency and a whole drove (infinite number) of harmonics. Some have waded through the Fourier series analysis, and some can see it intuitively. But if you have never seen it on a scope—even if you have been through Fourier analysis frontward and rearward—you should hook up a simple clipper, followed by a sharp filter that cuts off just above the frequency you are clipping. See Fig. 1 for a sample hookup.

Try this circuit; it is very dramatic. It also serves to illustrate one of the problems with audio speech clipping. The clipped waveform is cleaned up, that is, restored to a single frequency, only if the filter cutoff is relatively close to the frequency being clipped. For instance, if you clip a 200 Hz sine wave, and pass it through a 2 kHz filter, the nice sine wave does not come back. What you get is a mess; now the waveform is still sharp-edged but is usually tilted as well, due to the phase shifts through the filter.

And since the filter for an audio speech system cannot cut off before about 2000 Hz, there is an irreducible problem. Do not despair, however, there is a compromise solution which is well worthwhile. It is possible to have an audio clipper which does not sound bad.

Why do so many sound bad? One reason is obvious. The operator can't stop turning the level knob soon enough—depending on other stations to set clipping levels is haphazard at best.

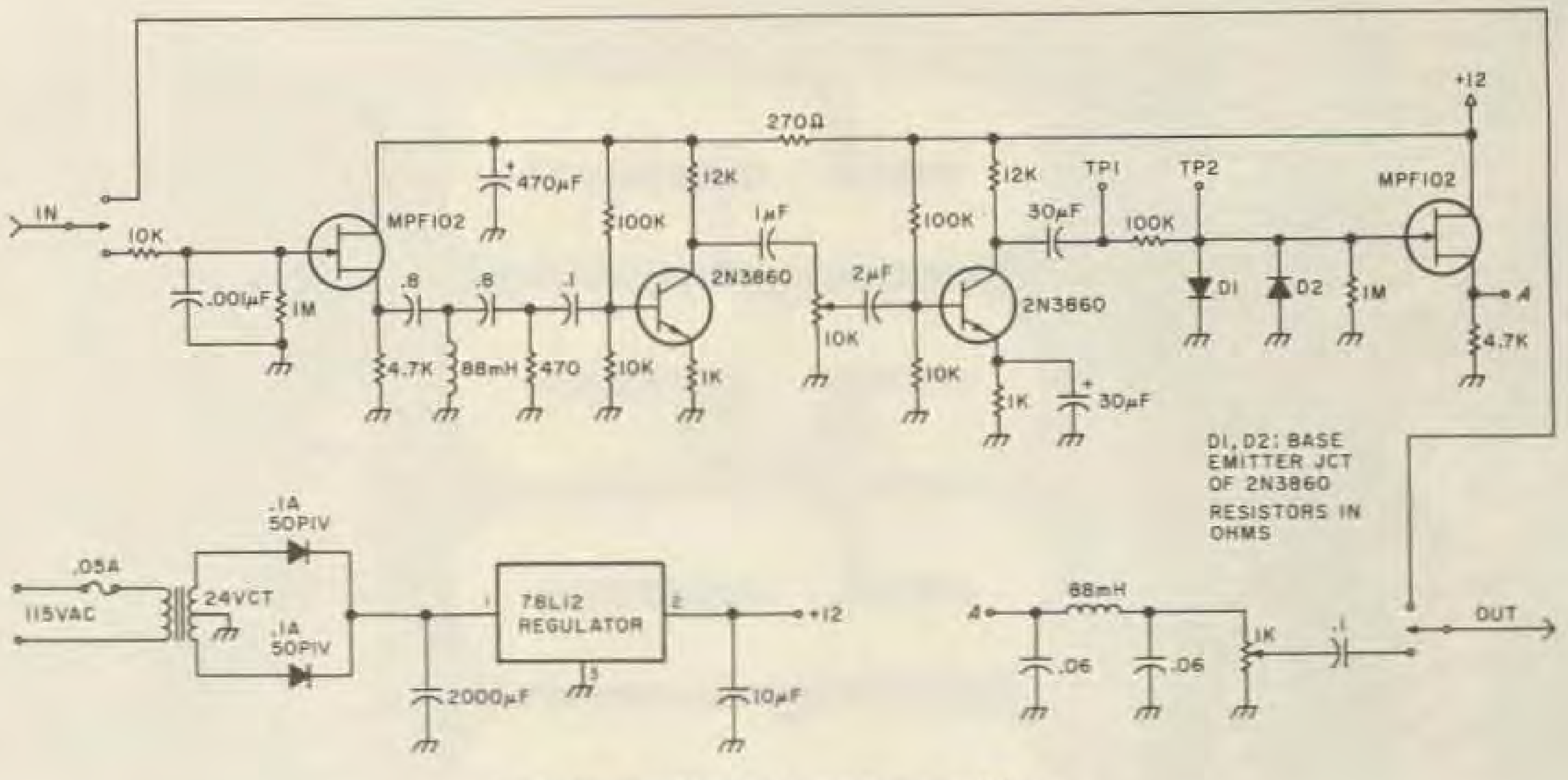

Fig. 2. Audio filter/clipper/filter.

Some indication can be obtained, however. You know you have gone too far when signals are 10 over 9, you are hearing no QRN/QRM and the other operator keeps asking you to repeat what you said. Many clippers, especially home brew ones, suffer from rf pickup. Rf pickup can destroy an otherwise good clipper. In addition to these problems, the low frequency phase shift/tilt problem is often heard. And finally, some operators using transmitters with sweep tube finals have discovered the tubes were not able to stand the increased duty cycle.

In spite of these caveats, clippers, as well as other forms of speech processing, are becoming more common now. The new all-transistor rigs are as comfortable with 100% duty cycle as they are with 30%, and the FCC has started to meddle with linear amplifiers.

And—are you ready?—The *Handbook* (ARRL) has a graph on page 392 (Figs. 13-20) in the 1977 edition showing 15 dB of audio clipping improves the signal-to-noise ratio by nearly 4 dB. Now you wouldn't build a linear amplifier for a four dB gain, unless you were a CBer, or instructed to by the FCC, but with an audio clipper you can get 4 dB for peanuts. Four dB, just lying around waiting for you to pick it up, like loose change, like found money.

Another goody, but not quite as satisfying as found money, is the text in the 1977 *Handbook* (ARRL) on clipping, clippers, and related subjects. A rather elaborate processor is detailed. It is good to read about, even if you don't build it; in the 60s we called stuff like that mind-expanding.

But enough of that; let's build a clipper. It ought to be simple. It ought to be cheap so some money will be left to build something else. But it ought to sound good. The filter/clipper/filter in Fig. 2 satisfies these objectives.

Looking back to address the problems listed above:

1. Rf. The 10k resistor and the .001 capacitor form a low-pass filter which keeps out rf. The 10k resistor could be replaced with a 1 or 2 mH choke, but the 10k resistor is cheaper, and adequate.

2. Low-frequency square waves and tilt. This problem is addressed by using low-frequency rolloff. All frequencies below 500 Hz can be greatly attenuated or even eliminated. The first MPF 102 source-follower feeds a T-section high-pass filter which attenuates the low frequencies, before clipping.

3. Tweaky fingers, or Oops! My plates just melted. The prototype has no knobs on the outside. Knobs on the outside are OK, if you can restrain yourself. Otherwise, you are better off to set it and forget it. Use a scope.

Additional notes: TP1 and TP2 are used with a scope to initially set the clipper. You can set it for whatever clipping level you want, up to the power supply voltage limitations. Eight volts p-p at TP1 sounds good. D1 and D2 are silicon junctions, so the level at TP2 will always be about 1.2 volts p-p. However, it is interesting to look at this point anyway.

The second MPF 102 source-follower feeds the low-pass filter. Output level is set with the 1k pot. A DPDT switch is included for those people who feel insecure if they can't do a regular comparison with distant operators.

My filter is used maritime mobile, and I find it a lot easier to carry around than a linear amplifier. It is very handy when running phone patches for the crew; I can tolerate the screamers and the whisperers—without external knobs. It's not as effective as a 2 kW linear amplifier, but it's a lot easier to pack into my suitcase. ■