

## Comments on *A Voice-Balancing Audio Peak Clipper*

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This note is prompted by my concern about the article by Professor Burger, WB6VMI, in the July 1998 issue of QST. From its tone, the article appears to be directed to beginners and I believe it does them (and their ham neighbors) a disservice.

In addition to some of the phrases such as, “*sidebands that deviate by more than they should from the reference frequency of a carrier*” and “*audio transformers and filtering capacitors that smooth the flow of an audio waveform*”, the most troublesome aspect of the article is the implication that the building and adjustment of an audio peak clipper is a simple process.

Although Professor Burger’s interests may focus on older tube-type DSB-AM equipment where simple audio speech clipping is more appropriate, today’s beginning ham is much more likely to be using an SSB transmitter. In the later case, audio speech clipping is most certainly non-trivial and beyond the scope of a simple beginner’s project.

Professor Burger states that overmodulation can result in “splatter” and that by using an audio peak clipper this can be prevented. He also claims that human voice waveforms are asymmetrical and that this condition needs special treatment. I would like to comment on these claims in turn.

### **Clipping and SSB**

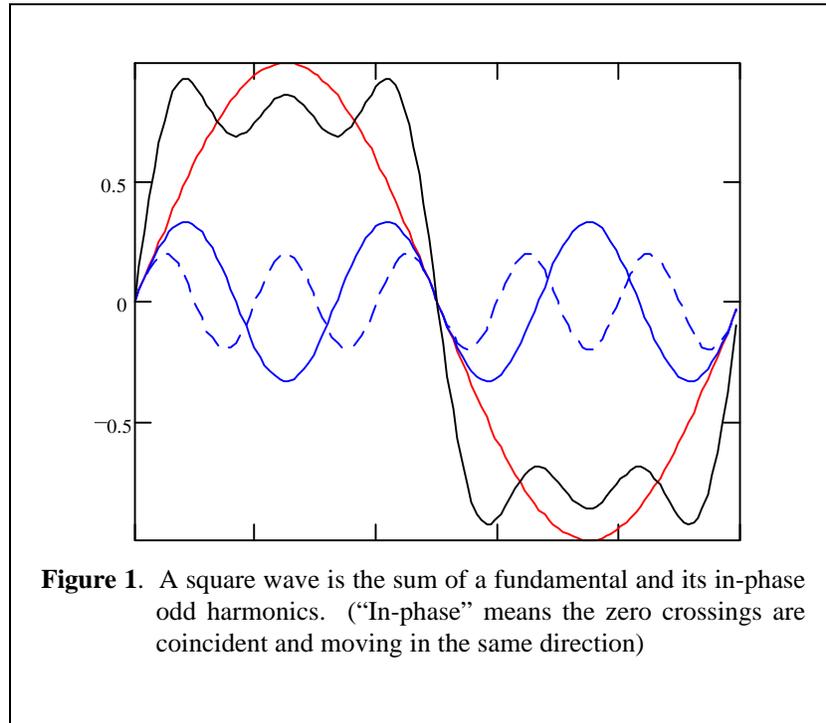
It is true that splatter<sup>1</sup> and excessively broad signals can result from overdriving the audio input (or any other stage) of a transmitter and that in some cases, judiciously applied audio peak clipping can be of assistance in alleviating the problem. However, it is equally true that the cavalier use of peak clipping can actually be counterproductive in the case of an SSB transmitter.

When a sine wave is symmetrically clipped or hard-limited, odd-order harmonics of the fundamental are generated, with the level of the harmonics increasing as the clipping becomes more severe. If there is any asymmetry, there will also be even-order harmonics produced. A sine wave that has undergone a large amount of symmetrical clipping approaches the appearance of an ideal square wave. In fact, Figure 3 in Professor Burger’s article is an example of this. By scaling the waveforms in the figure, I estimate that at least 15 dB of clipping is occurring in the “simple” clipper. This is a significant amount of clipping and is in my opinion, beyond the realm of a device used to reduce “accidental overmodulation.”

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<sup>1</sup> R. P. Haviland, W4MB, “Single Sideband Speech Splatter,” *hr magazine*, September, 1975 page 28

In the ideal square-wave, the in-phase fundamental and odd-order harmonics sum together to produce the overall envelope. Figure 1 below demonstrates this. In the figure, the fundamental and its third and fifth harmonics are shown, along with their arithmetic sum. This could be for instance, a 500 Hz audio tone along with the 1500 and 2500 Hz harmonics, all of which would pass through a voice-bandwidth filter.



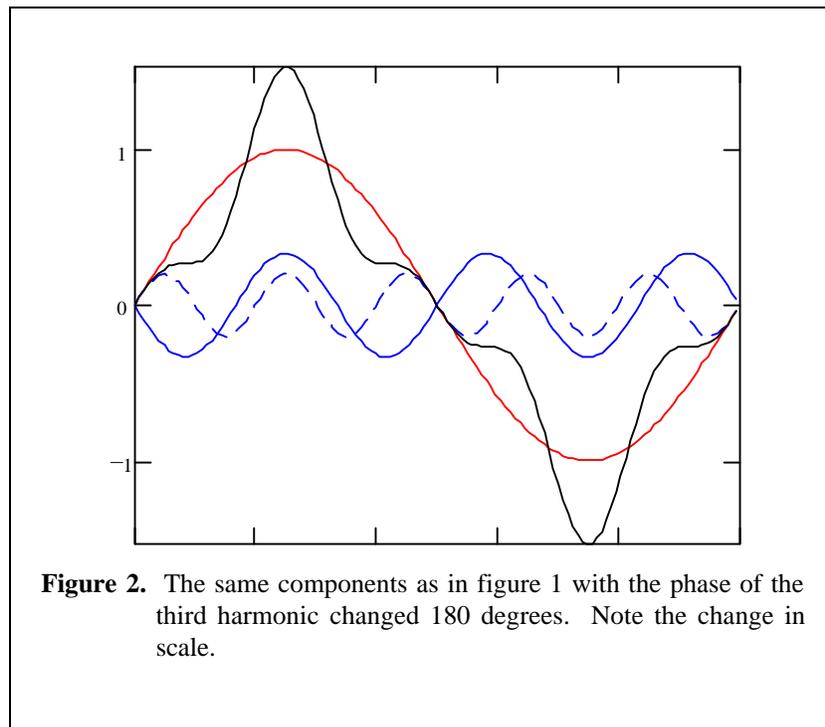
**Figure 1.** A square wave is the sum of a fundamental and its in-phase odd harmonics. (“In-phase” means the zero crossings are coincident and moving in the same direction)

The presence of these harmonics produces in-band distortion. If the intelligibility improvement due to the clipping increases at a faster rate than the intelligibility reduction due to the distortion, then there is a benefit to increased clipping.<sup>2</sup>

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<sup>2</sup> Harold G. Collins, W6JES, “Ordinary and Processed Speech in S.S.B. Application,” *QST*, January, 1969, page 17

The foregoing is true only when the harmonics due to clipping remain in phase with the fundamental. Unfortunately, after the audio frequency to radio frequency translation in an SSB transmitter this phase relationship is lost. Schreuer<sup>3</sup> points out that when a heavily clipped, low audio frequency sine wave is used to modulate an SSB transmitter, the resulting radio frequency envelope will actually have a 14 dB peak-to-peak variation, as opposed to the anticipated flat output. This comes about because the phases of the harmonics change with respect to the fundamental. (See Figure 2) The effect of this is to introduce another modulating frequency (at twice the fundamental). With voice, the phenomena is more random and occurs less often than in this steady-state example; however, the undesired peaking still occurs.



Other texts point out that transmitting an ideal audio square wave requires infinite bandwidth and infinite peak power.<sup>4</sup> Of course, the square wave is never ideal, so these extremes are never reached. However, even in this case, the peaks will be very high. If a properly operating ALC is used, it will operate on these peaks and attempt to reduce them into the amplifier's linear region. This has the effect of *reducing* the average output power; exactly the opposite of the desired effect.

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<sup>3</sup> Walter Schreuer, K1YZW, "Speech Clipping in Single-Sideband Equipment," *hr magazine*, February, 1971, page 22.

<sup>4</sup> *Fundamentals of Single Sideband*, 2<sup>nd</sup> edition, Collins Radio Company, Cedar Rapids, Iowa, 1959, p 1-15.

Well-designed speech processors for SSB that utilize clipping either operate at the transmitter i-f<sup>5</sup> or if the clipping is done at audio frequency, use the “split-band” approach.<sup>6</sup> In the radio frequency clipper, the harmonics generated are harmonics of the i-f and are easily eliminated, which leaves only some small intermodulation distortion products. Several popular transceivers use this method, although, it does require a second i-f filter.<sup>7</sup>

The split-band approach clips at audio frequency, but does so in sub-octave frequency bands and follows the clippers with a second set of identical filters which remove the troublesome harmonics. This writer has found this technique to be extremely effective. The split-band technique is currently used in the Kenwood TS-870S.

The fact that these relatively complicated techniques have been developed for SSB use is a clear indication that audio clipping used in SSB applications is more than a “beginners” project. If it was simple, the developers and authors of the references would have spent their time doing something else.

### **Asymmetrical clipping and SSB**

Professor Burger suggests that speech waveforms are asymmetrical, with positive peaks exceeding negative peaks or vice versa. He then suggests that the corrective action for this is to set different positive and negative clipping thresholds. Assuming for the sake of discussion that the professor’s original premise that the waveforms are asymmetrical is true, I am still baffled as to why the clipping thresholds should be independently adjusted.

First, as stated above, with asymmetrical clipping we no longer have only odd-order harmonics to deal with. The even-order harmonics that are generated are an even more onerous filtering problem.

Second, in the SSB case, there is no correlation between the audio waveform and the radio frequency envelope. If a single audio sine wave is used to drive the SSB exciter, the ideal output is a single radio frequency sine wave. Clearly, there will be no way to perform any kind of “voice balancing” in this situation, particularly when considering the admitted interaction of the two “independent” adjustments. Presumably, this is why the professor suggests adjusting the clipping levels while transmitting AM. Unfortunately, in modern transceivers, the AM is hardly ever true DSB-AM, but is instead SSB with inserted carrier. The use of this mode for clipping level adjustment is still problematic; in addition to two clipping level adjustments, the reinserted carrier level becomes another variable.

The professor goes on to say that if the AM mode is unavailable, the builder should adjust the clipping controls to “avoid flat-topping.” I guess this assumes that the experimenter has available a wide-band oscilloscope with which to observe a cycle or two of the radio frequency signal.

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<sup>5</sup> William Sabin, W0IYH, “R.F. Clippers for S.S.B.,” *QST*, July, 1967, page 13

<sup>6</sup> Wes Stewart, N7WS, “Split-Band Speech Processor,” *hr magazine*, September, 1979, page 12

<sup>7</sup> Leslie Moxon, G6XN, “Performance of RF Speech Clippers,” *hr magazine*, November, 1972, page 26

Because any distortion (flat-topping) of this signal will generate harmonics of the carrier frequency, the bandwidth of the oscilloscope must be several times the carrier frequency. It will also be necessary for the observer to have a very keen eye in order to detect the onset of limiting (flat-topping). This is almost impossible to do, which is why a two-tone test is the preferred method for adjustment of final amplifier drive level.

In conclusion, I believe that Professor Berger's heart is in the right place and that he is truly trying to be helpful. However, considering the foregoing, I believe that it is evident that this is not a suitable beginner's project.