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All-Pass Networks in a Speech Chain

W4ENE discusses asymmetrical audio waveforms, the problems this creates, and ways of minimizing the asymmetry.

It is a well-known fact that a speech waveform viewed on an oscilloscope is quite commonly lopsided. That is, one side of the waveform, say the top side, has a greater peak amplitude than the bottom side. The degree of this asymmetry is highly dependent on the voice of the individual. Here, we will explore this phenomenon and outline some problems that may arise from it. We will also discuss methods for minimizing the problems.

Possible Problems – Entirely Linear Systems

In a high quality public address system where the speaker's voice is simply augmented by a power amplifier and loudspeaker, there would be no problem if the volume levels are such that the system is entirely linear. The lopsided waveform would pass with its asymmetry unnoticed.

If such a lopsided waveform were to be used to modulate an AM radio transmitter, and if the modulation index is adjusted to a relatively low level, then again such a waveform would offer no problem. However, it has been found best for ordinary AM systems to set the polarity of modulation so that the peaks with the greater amplitude are modulated upward. This minimizes distortion found in the typical AM signal envelope. However, in a purely linear system, asymmetric waveforms are not in themselves a problem.

AGC Loops with Fast Attack Times

There was at one time — circa 1950s, 1960s — a competition among various broadcast equipment manufacturers to see who could develop the fastest-acting AGC system to control the modulation in a broadcast transmitter. These devices were commonly called volume limiters or volume limiting amplifiers. They generally reacted to an overload situation within a millisecond. Following the overload they would restore the gain to normal over a period of perhaps a few seconds. They were without question better than a human controlling the modulation levels.

It was interesting to view the output of these devices on an oscilloscope. There was no visible clipping or other artifact added to the waveform, just a (usually) well-controlled modulation level. The winner in the war-of-speed used a system that had a zero attack time. It used a delay line to delay the audio signal while the gain-controlling voltage was being generated. But in every one of these units there was an annoying tendency for the device to respond to signals that were not the same as that to which the human ear responded. Rephrased, they were controlling modulation, not volume. Maximizing volume was becoming an issue at the time. So, while the outputs of these fast-acting devices looked nice on an oscilloscope, they didn't accomplish the broadcaster's needs.

If a lopsided waveform were to be applied to one of these units, the peak with the greatest magnitude, whether positive or negative, would cause the generation of AGC voltage. If those peaks could be made equal in amplitude then less AGC voltage would be generated and modulation would increase. This must be done, however, in a manner that does not increase the peak-to-peak value.

Transient Clipping

Research at CBS Laboratories showed that if the AGC loop could have a reaction time (attack time) of a few milliseconds, and a recovery time (release time) of perhaps 200 ms, such an AGC system would best

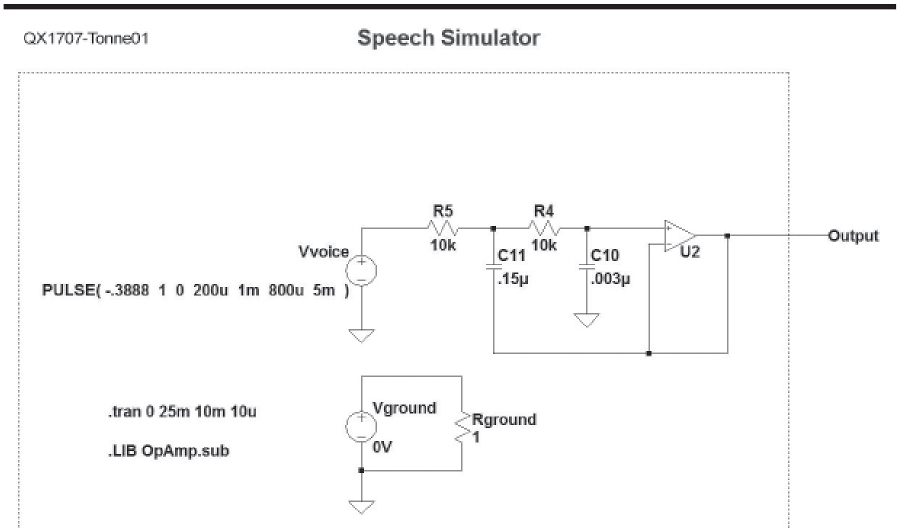

Figure 1 — LTspice rendering of the speech simulator.

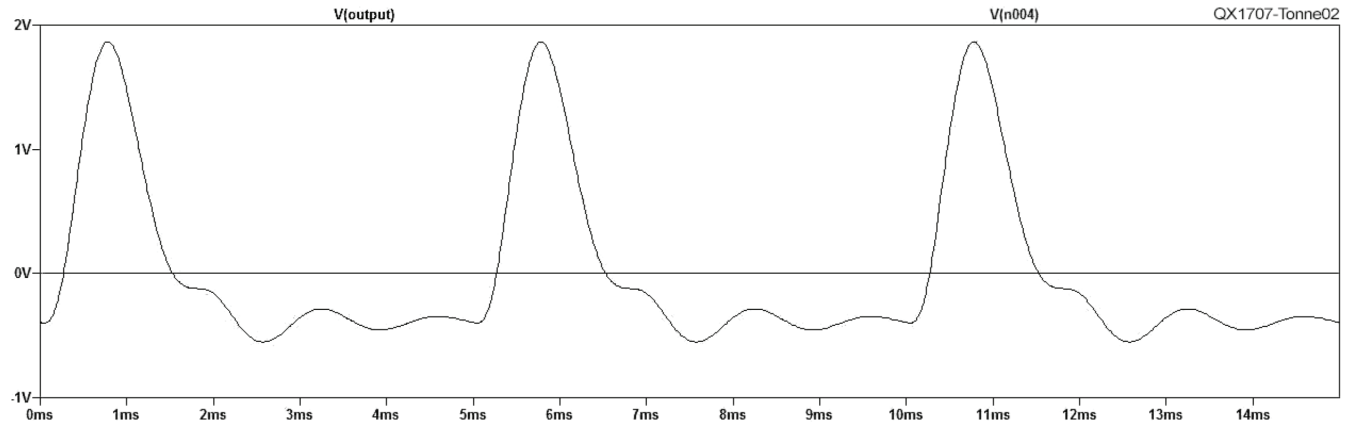

Figure 2 — Waveform from the speech simulator.

match the response of the human ear. Such an AGC system could control loudness and it would match perception by the ear, allowing the volume to be maximized. This is what broadcasters were looking for.

There was a drawback to such a scheme. The relatively long attack time required that such an AGC system must be followed by a clipper to catch the transients that escaped the AGC system. One of the first commercial units to use these techniques was the *CBS Volumax* system. Such a long attack-time system was quite a departure from the conventional wisdom of the time.

In a system with such a long attack time, an asymmetrical applied waveform causes less AGC voltage generation than in a system with a short or fast attack time. However, the signal from the AGC system must then be applied to a clipper. Clipping one audio peak more than the other results in a dc, or at least a sub-audible syllabic, signal component from the clipper. In an AM transmitter this appears as a form of amplitude "carrier shift". In an FM transmitter this appears as a center frequency shift, and interferes with the FM transmitter Automatic Frequency Control (AFC) system. It is certainly disconcerting to watch an analog frequency meter on an FM broadcast transmitter kick violently when such a unit is used to control the modulation with an applied program containing asymmetric waveform components involving a clipper.

To minimize this problem, the waveform should be processed in a manner such that prior to clipping, the peaks are rendered symmetrical, preferably without increasing the peak-to-peak value. Next we look at ways to handle the problem.

A Standardized Waveform

Let us generate a standard waveform that can be reproduced easily and will allow us to compare various approaches to processing. The proposed signal has an approximate 10 dB (3:1 voltage ratio) positive to negative amplitude ratio, but it has no dc component. The areas under the curve for the positive and for the negative portions of the waveform are equal. A circuit to generate such a speech waveform is shown in the LTspice model rendered in Figure 1. The signal generated by that circuit has a fundamental frequency of 200 Hz and is a believable replica of a steady speech signal, perhaps the sound "ohhhh". The generated signal is shown here in Figure 2.

There is about 10 dB of asymmetry in this waveform and there is no dc component. The areas above and below the centerline are precisely equal. If this waveform were to modulate a transmitter directly, the upward peaks would require 10 dB more power than the downward peaks. If the transmitter could not handle this degree of asymmetry, then the modulation level would have to be reduced until the positive peaks were in a linear region and the negative peaks would be reduced in amplitude. This is certainly an inefficient use of transmitter capability.

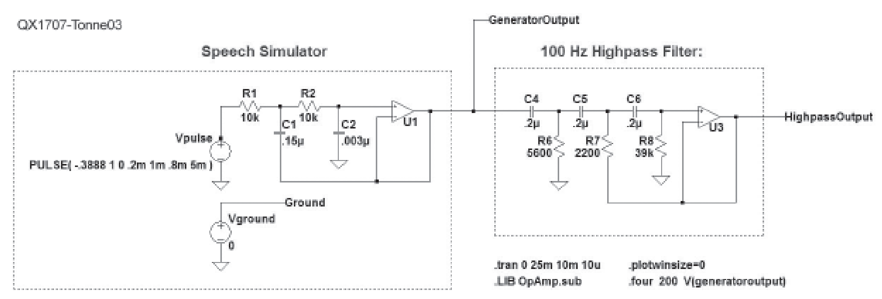

Figure 3 — LTspice rendering of the speech simulator of Figure 1 followed by a 100 Hz high-pass filter.

A Solution– Use a High-pass Filter

One way to make the waveform symmetrical "top to bottom" would be to apply it to a high-pass filter. Indeed, this may be a part of the speech-processing chain already. Such a high-pass filter would do double duty. It would remove those components that serve no purpose if transmitted, and in fact might cause mischief. The high-pass filter might also make the positive and negative peaks more nearly equal. If the speech signal were applied to a high-pass filter and then to a clipper, there would be similar amounts of clipping applied to the positive and negative peaks. There would be a reduction in axis shift due to any sub-audible components generated by asymmetric clipping. The ear normally tolerates clipping of both modulating waveform peaks better than clipping just one side of the waveform.

In Figure 3 we see our standard speech waveform generator of Figure 1 connected to a 100 Hz high-pass filter. Figure 4 shows the output of the filter (lagging waveform) compared with the output of the generator (leading waveform). The horizontal line depicts the zero voltage level. Observe that the peak-to-peak voltage value has actually increased. This is not our objective. The use of a high-pass filter is not helpful in this respect.

A Better Solution: Use an All-pass Network

Another way to process the speech signal is to pass it through a network that has a flat frequency response but rearranges the relative phases of the signal frequency components to make it less asymmetric. Because such a network passes with all frequency amplitudes equally, it is called an all-pass network.

Let us look at an active version — using op-amps, resistors and capacitors — of such an all-pass filter. At various frequencies the phase through the circuit shifts. If a complicated waveform, made up of a fundamental and various harmonics, is applied to the circuit, the harmonics have their phases altered relative to the fundamental. This can be accomplished with a lumped-element circuit — using inductors and capacitors — or it can be accomplished at much lower expense using active circuitry using op-amps, resistors and capacitors as shown in the Figure 5.

This network will have a flat frequency response if R1 and R2 have the same value. R3 and C1 can be interchanged. The delay performance will be identical, just the phase will be inverted.

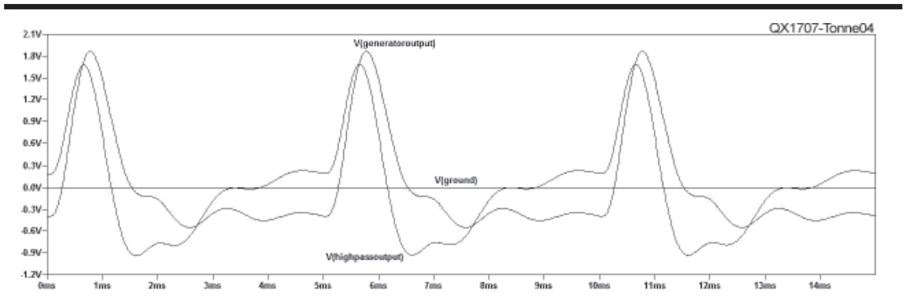

Figure 4 — Output of the filter (lagging waveform) compared with the output of the generator (leading waveform). The horizontal line depicts the zero voltage level.

The Kahn SymmetraPeak Circuit

Using an all-pass network to make an audio signal more symmetrical is certainly not new. A commercial product to provide this function was called the SymmetraPeak and was marketed by Kahn Communications about 1959. It was a lumped-element device for the simple reason that at the time op-amp circuitry was not available. The schematic of the SymmetraPeak, rendered in LTspice, is shown in Figure 6. Our quite asymmetric test waveform has become relatively symmetrical, as seen in Figure 7.

Active Circuitry

The SymmetraPeak was bulky and expensive. When op-amp circuits were developed that accomplished the same thing it faded away. An example of an op-amp equivalent (Figure 8) was designed by Gary Blau,W3AM. Our standard signal as it exits from that active all-pass is shown in Figure 9. It is even more symmetrical than from the SymmetraPeak.

The signal peak-to-peak amplitude is not changed. The magnitude of the higher-amplitude peak has been reduced while at the same time the magnitude of the lower-amplitude peak has been increased. There is no axis shift, and no sub-audible components have been added. The areas under the curve above and below the zero axis are equal. Clipping of such a waveform would cause a minimum of "mischief" compared with clipping of the original asymmetric waveform. The phase shift of this network goes from near zero degrees at very low audio through near 1440 degrees at extremely high audio frequencies.

RMS-sensing AGC and Clipping

If the modulation level in the transmitter is controlled by a peak-sensing audio AGC unit (a "limiter"), that AGC system will respond to the peak with the highest instantaneous magnitude. But if the modulation level is controlled by an rms-sensing AGC unit, asymmetry does not enter into the picture at all with respect to the AGC portion of the speech processing. But an rms-sensing AGC unit must be followed by a clipper to catch those waveform excursions that escape the AGC unit. Be advised that those excursions will be of significance. But if the clipper operates on one side of the waveform more than the other, a dc or sub-audible component will be developed by the clipper. This will normally cause trouble in the modulator proper. We have a situation wherein the modulator must be direct-coupled to properly handle the signal.

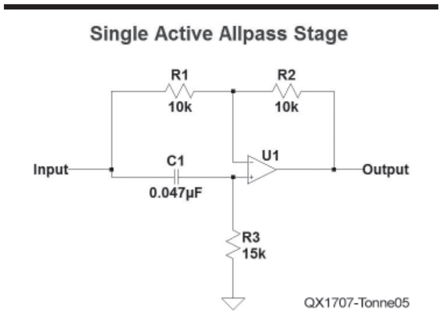

Figure 5 — A single stage active all-pass network rendered in LTspice.

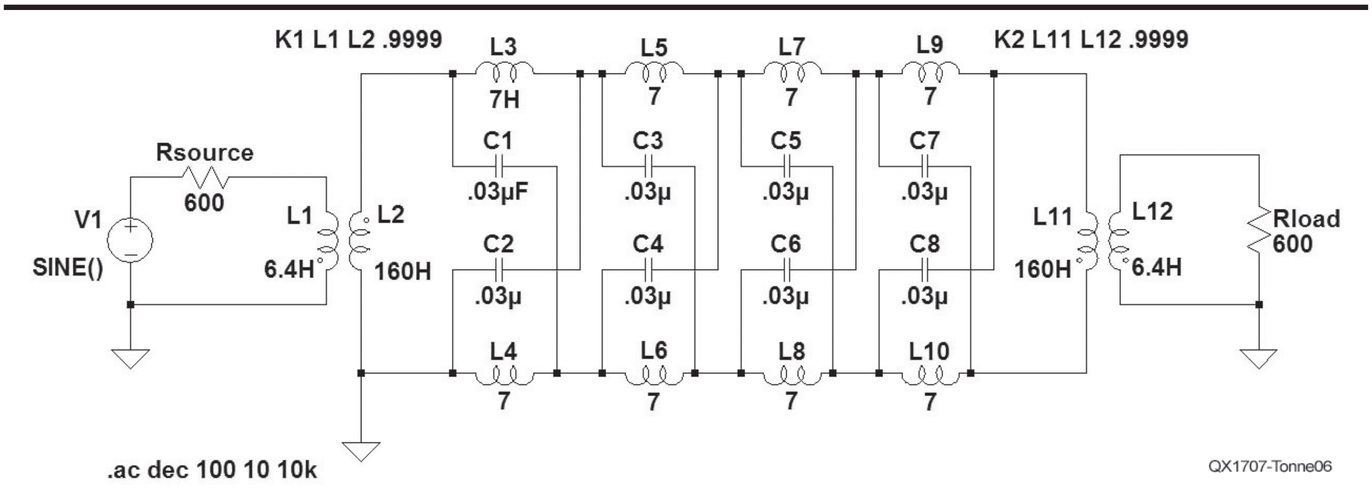

Figure 6 — The Kahn SymmetraPeak circuit rendered in LTspice.

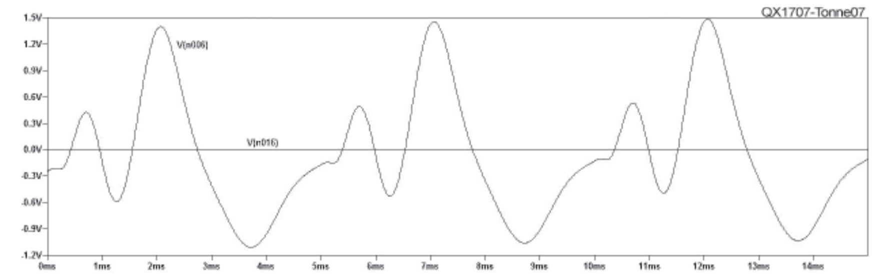

Figure 7 — Output of the SymmetraPeak. The trace is V(n006), and the straight line is V(n016).

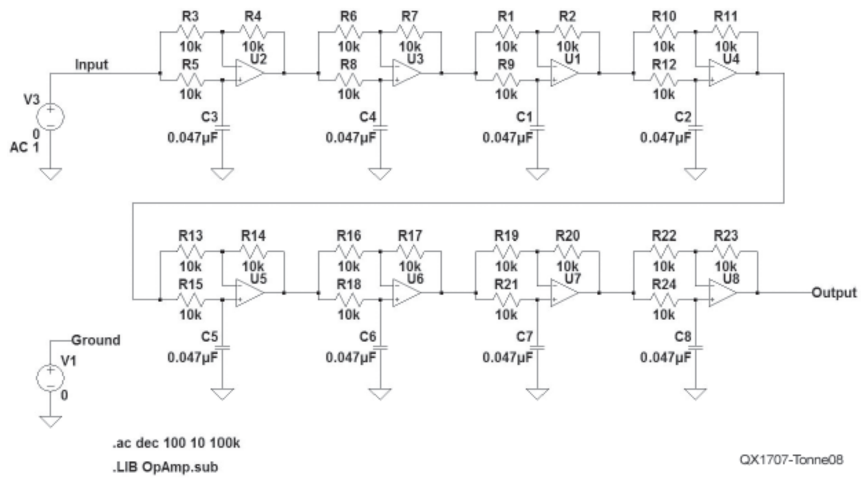

Figure 8 — An active all-pass network rendered in LTspice.

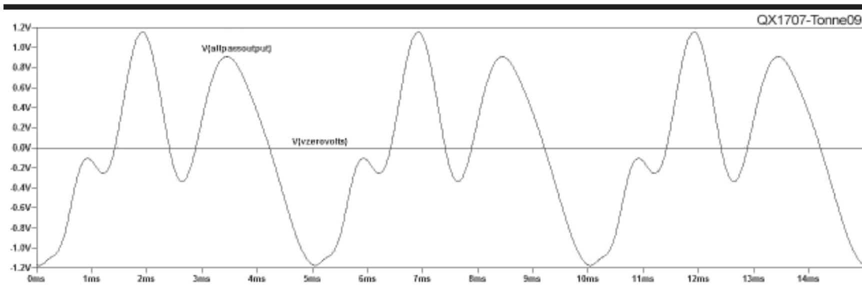

Figure 9 — Output of the active all-pass network. The trace is V(allpassoutput), and the straight line is V(vzerovolts).

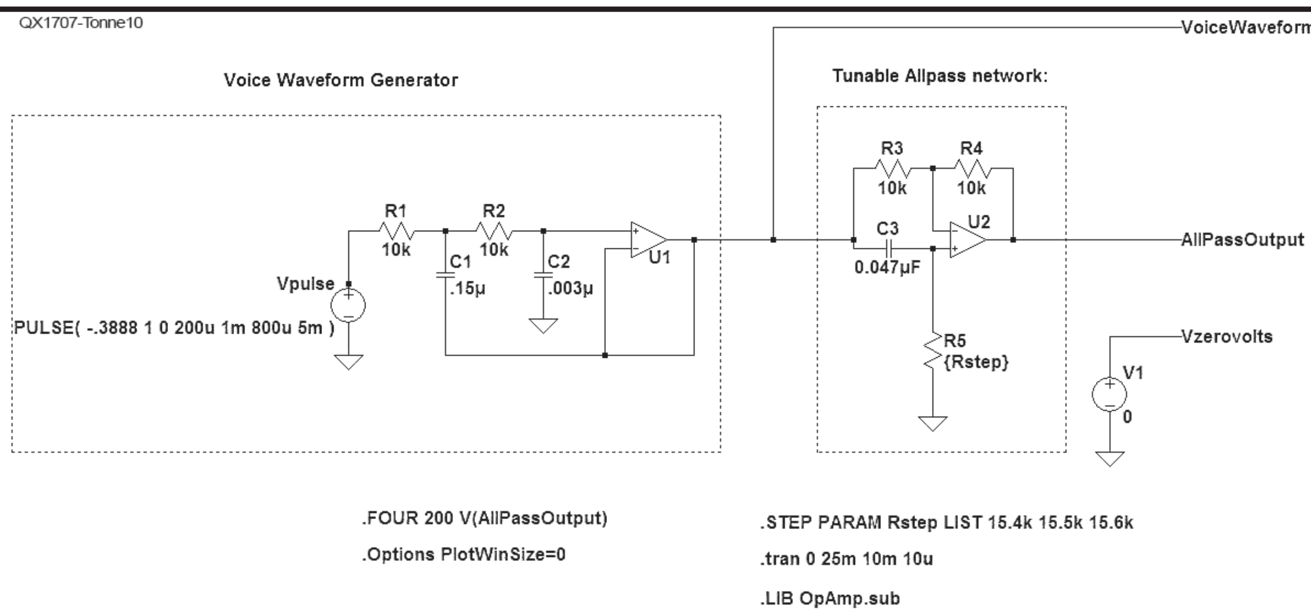

Figure 10 — Schematic of a tunable (with Rstep) all-pass network driven by the waveform generator.

Here again the all-pass can come to the rescue. Insert the all-pass between the rms-sensing AGC block and the clipper. By adding the all-pass network at this point, the clipper will clip symmetrically and no sub-audible components will be involved.

A Tunable All-pass Network

A tunable single-stage all-pass network has been devised that may very well be suitable in most cases. The schematic is shown Figure 10 as it underwent development in LTspice. By using an oscilloscope, R5 (“Rstep”) can be adjusted for best waveform symmetry on the individual’s voice. However, the peak-to-peak amplitude will not be as nicely controlled as with the more complex network. This simple circuit does give a significant amount of performance as seen in the output waveforms of Figure 11.

This all-pass network is quite comparable with the more complex circuits for this speech waveform. It is the customization that allows the high degree of performance seen with this simple circuit. The caveat is that various speech waveforms may require readjustment of the potentiometer. This should not be a problem in the usual one-user application.

A Possible Failure Mode

The use of an all-pass network is not a cure-all. If the applied waveform is symmetrical top-to-bottom in the first place — it has no even-order harmonic content — then the all-pass can actually increase the signal peak-to-peak amplitude. In Figure 12 we see an example of an applied waveform with top-to-bottom symmetry. The signal consists of a 200 Hz fundamental and an equal amplitude 600 Hz third harmonic. Now let us apply that signal to a Blau all-pass circuit. The resulting output signal is shown in Figure 13.

The signal still has top-to-bottom symmetry but the peak amplitude has actually increased. This can be seen by comparing of this waveform with the waveform of Figure 12. This illustrates an interesting point. If the waveform to be corrected has no even-order components (unlikely in practice) and so is symmetrical top-to-bottom in the first place, then the using an all-pass network might not be beneficial, and in fact will be harmful. This aspect of the all-pass is usually glossed over.

A Suggestion

An all-pass block should be placed at an appropriate point in a speech processor used in a radio transmitter. It should be placed ahead of a clipper if any preceding AGC circuit is slow-acting or especially if it is mis-sensing.

The all-pass network is inexpensive if constructed using op-amps and associated components, and generally ensures lower distortion by virtue of less clipping or at least symmetrical clipping. Symmetrical clipping always causes less “mischief” than does asymmetrical clipping. Placement of the all-pass block should always be prior to the point where clipping occurs or where it might occur.

The amateur radio fraternity has been relatively slow to pick up on the idea of using an all-pass network, although Gary Blau, W3AM, see **www.w3am.com/8poleapf.html**, also writes on this subject.

James L. Tonne, W4ENE, holds the Amateur Extra class license, and was first licensed in 1951. His current Amateur Radio interests are largely focused on speech processing and filter design. He has written several articles for QST *and* QEX *and was a major contributor to the RF and Filters chapter in the ARRL Handbook. He is the author of the Tonne Software package on the CD accompanying the ARRL Handbook and included as part of the downloadable package available on the ARRL web page.*

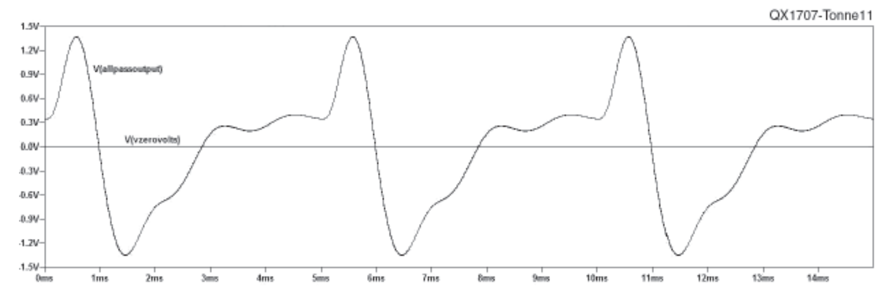

Figure 11 — Output of the simple tunable all-pass network. The trace is V(allpassoutput), and the straight line is V(vzerovolts).

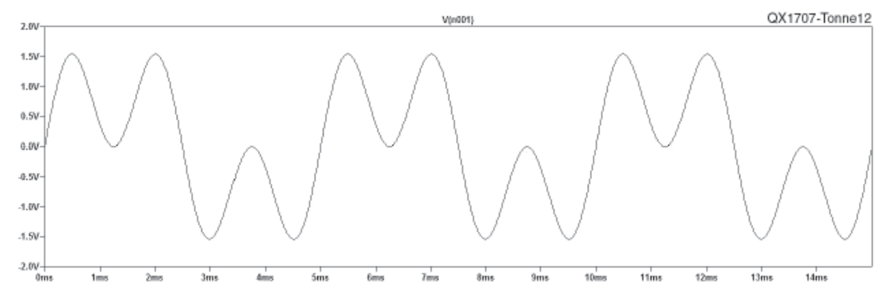

Figure 12 — A waveform V(n001) with top-to-bottom symmetry

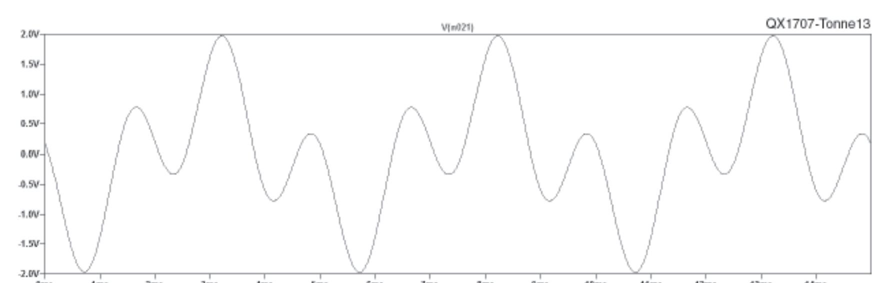

Figure 13 — Illustrating an all-pass failure, the result of applying the waveform of Figure 12 to a Blau all-pass circuit.