Allpass Networks in a Speech Chain

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Updated 4 January 2017

Intro

It is a well-known fact that if one observes a speech waveform using an oscilloscope it is quite commonly "lopsided." By this we mean that one side of the waveform, say the "top" side, has a greater peak amplitude than the other side. The degree of this asymmetry is highly dependent on the voice of the individual involved. This paper will explore this phenomenon and outline some problems which may arise from it and methods for minimizing those problems.

Possible problems - entirely linear systems

In a high quality public address system where the speaker's voice is simply being 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 in a noncompetitive situation, and if the modulation is adjusted to be relatively low, then again such a waveform would offer no problem. However, it has been found best in ordinary AM systems to set the polarity of modulation such that the peaks with the greater amplitude are "upward." This minimizes distortion found in the typical envelope demodulator in the usual radio receiver.

But 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 kind of contest between various broadcast equipment manufacturers to see which one 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 or less. Following the overload they would restore the gain to normal over a period of time of perhaps a few seconds. They were without question better than a human in controlling modulation levels.

It was impressive to view the output of these devices on an oscilloscope. There was no visible clipping or other artifacts added to the waveform, just a (usually) well-controlled modulation level. The winner in the war of speed used a system which 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 which were not the same as that to which the human ear responded. Rephrased, **they were controlling modulation, not volume**. And maximizing volume was becoming an issue at the time. So the outputs of these fast-acting devices looked nice on an oscilloscope but they didn't do what the broadcaster was wanting.

Further, if a lopsided waveform were to be applied to one of these units the peak with the greatest magnitude (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 which does not increase the peak-to-peak value.

Transient clipping

Research [CBS Labs] showed that if the AGC loop could have a reaction time (attack time) of a few milliseconds and a recovery (release time) of perhaps 200 milliseconds that such an AGC system would best match the human ear. Such an AGC system could control loudness and it would match the ear, allowing the volume to be maximized. This is what the broadcaster was 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 which escaped it. The first commercial unit to use these techniques was the CBS Laboratories' "Volumax." Such a relatively long attack-time system was quite a departure from the conventional wisdom of the time.

In a system with such a relatively 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. And if one audio peak is clipped more than the other we have a DC (or at least subaudible/syllabic) component exiting 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. It ends up interfering with the FM transmitter's Automatic Frequency Control 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 when a clipper is involved.

To minimize this problem the waveform should be processed in a manner such that prior to application to the clipper the peaks are made symmetrical, preferably without increasing the peak-to-peak value.

Let us now look at ways to handle the problem.

A standardized waveform

First for this paper let us generate a standard waveform that can be reproduced easily and will allow us to compare various approaches to processing. The proposed signal here has an approximate 10 dB (3:1 voltage-wise) positive to negative 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 we will use in this paper to generate a speech waveform is shown here as it appears in LTspice:

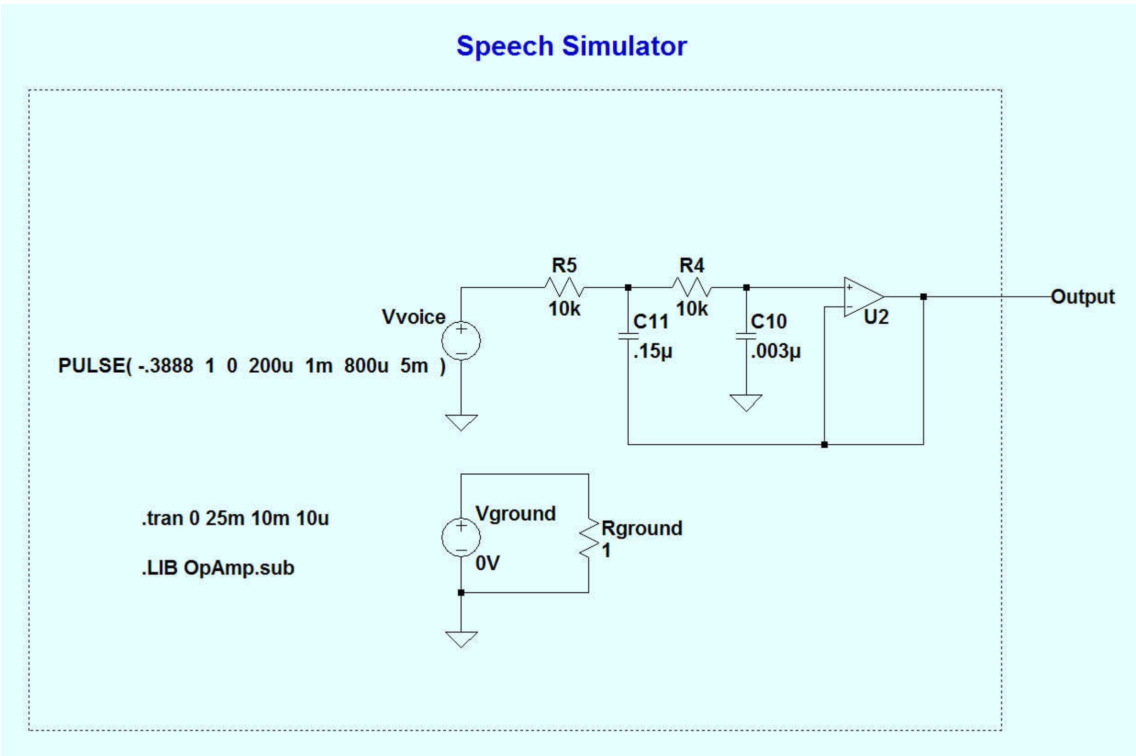

Fig 1 - Speech Simulator

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 signal as generated is shown here**:**

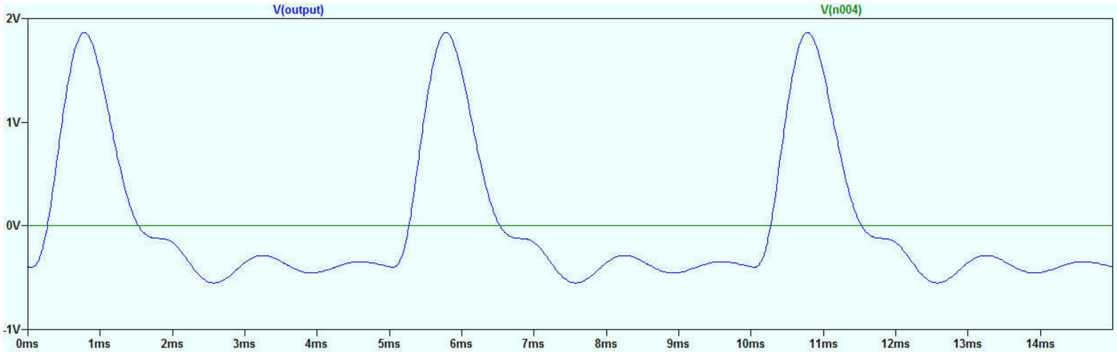

Fig 2 - Waveform from the speech simulator

There is about 10 dB of asymmetry in this waveform and there is no DC component. As can be seen, the areas above and below the centerline are precisely equal.

If this waveform is used 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.

A solution: use a highpass filter

One way to accomplish the task of making the waveform symmetrical "top to bottom" would be to apply it to a highpass filter. Indeed, this may be a part of the speech-processing chain already. Such a highpass filter would be doing double duty. It would be removing those components that serve no purpose if transmitted, and in fact may be causing mischief. The highpass filter might also make the positive and negative peaks more nearly equal. If the speech signal were applied to a highpass 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 subaudible components generated by asymmetric clipping. And the ear normally tolerates clipping of both modulating waveform peaks better than clipping only one side of the waveform.

Here we see our standard speech waveform generator connected to a 100 Hz highpass filter:

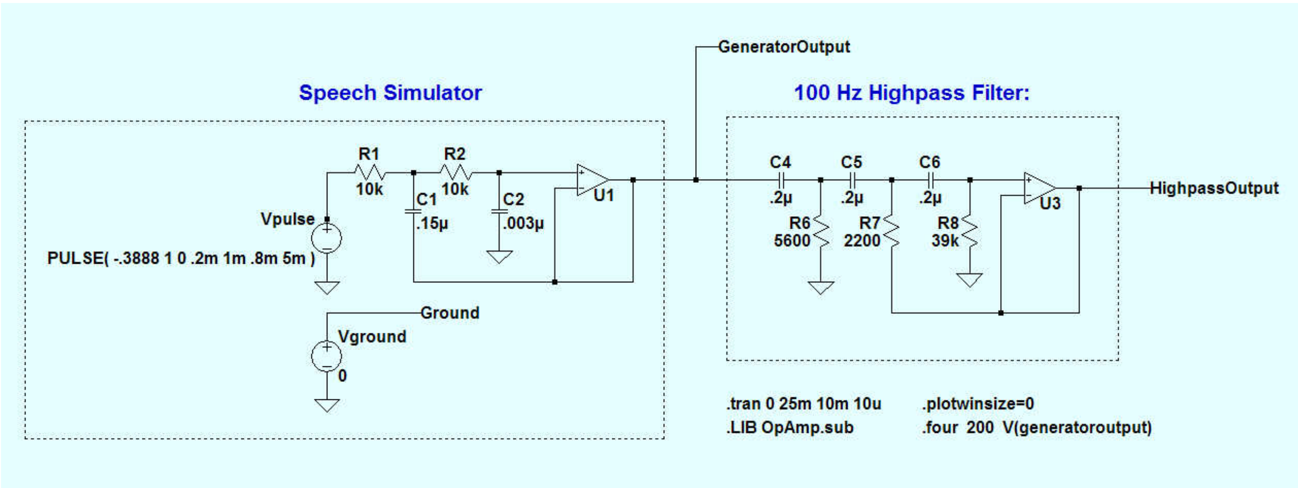

Fig 3 - Speech simulator and a 100 Hz highpass filter

The output of the filter will appear as shown in this waveform pair**:**

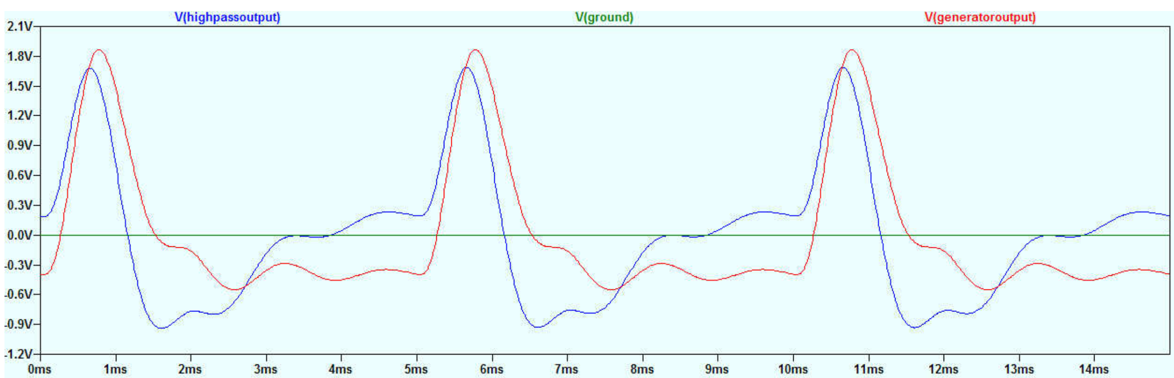

Fig 4 - Output of the highpass filter

The generator output is the red waveform and the highpass filter output is the blue waveform. But observe that the peak to peak voltage value has actually increased; this is not our object. The use of a highpass filter is not helpful in this respect.

A Better solution: use an *allpass* network

Another way to process the speech signal is to apply it to a network which has a flat frequency response but rearranges the relative phases of the signal to make it less asymmetric (more symmetrical). Because such a network passes all frequencies equally, it is called an allpass network.

Let us look at an active version (using opamps, resistors, resistors and capacitors) of such an allpass. 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 opamps, resistors and capacitors). Shown here is one form of an active allpass circuit:

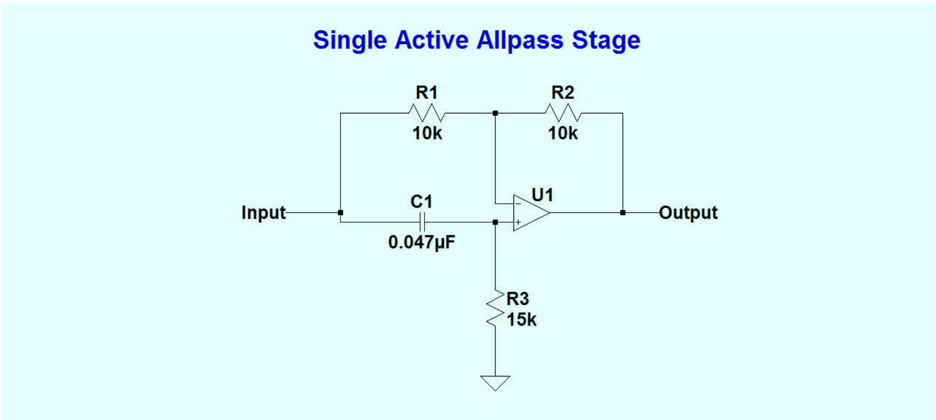

Fig 5 - A single stage active allpass network

This network will have a flat frequency response if R1 and R2 have the same value. Resistor R3 and capacitor C1 can be interchanged. The performance regarding delay will be identical; only the phase will be inverted one way compared to the other.

The Kahn SymmetraPeak

The concept of using an allpass network to make an audio signal more symmetrical is certainly not new. The first commercial product to provide this function was called the SymmetraPeak and was placed on the market by Kahn Communications about 1959. It was a lumped-element device for the simple reason that at the time opamp circuitry was not available. The schematic of the SymmetraPeak as entered into LTspice is here:

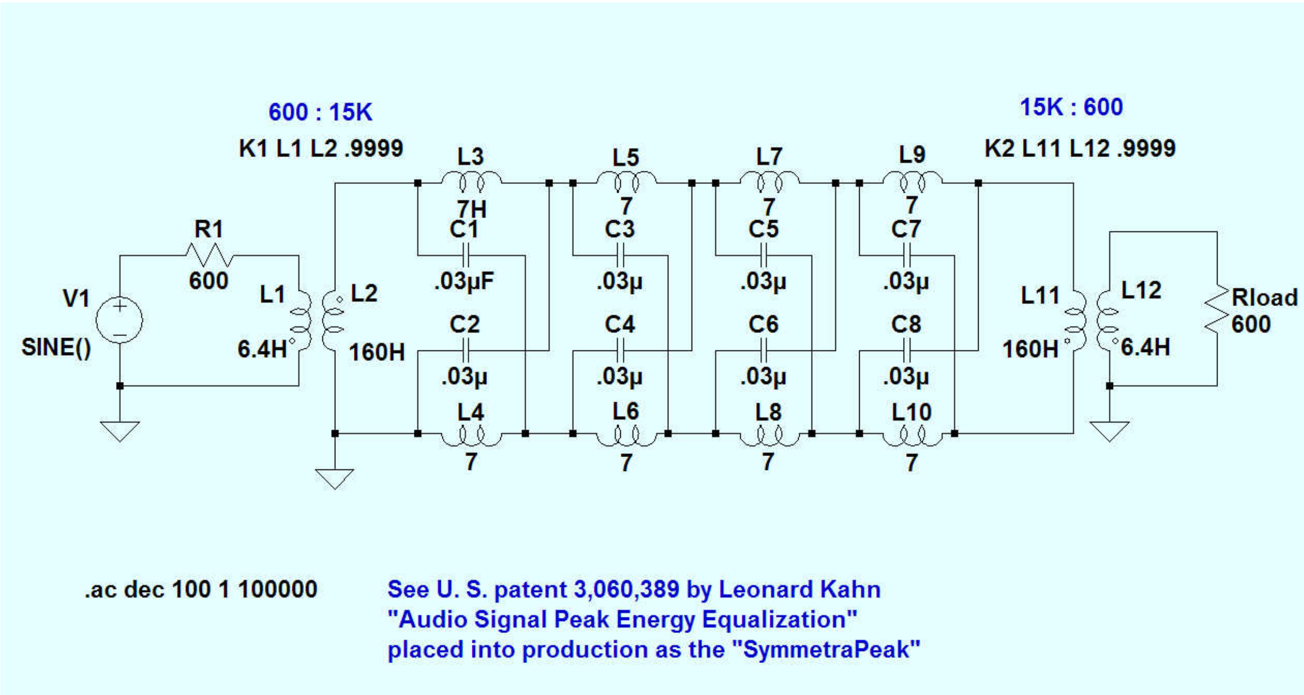

Fig 6 - The Kahn SymmetraPeak

The output of the SymmetraPeak is shown here. Our quite asymmetric test waveform has become relatively symmetrical:

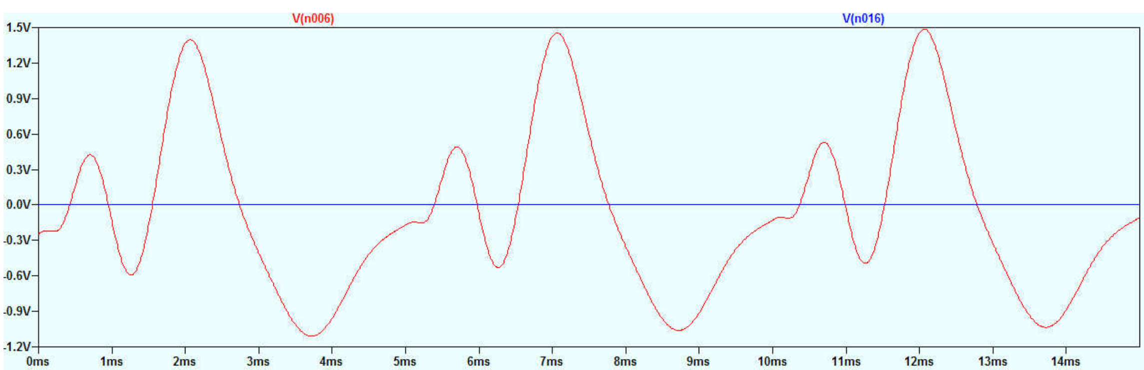

Fig 7 - Output of the SymmetraPeak

Active Circuitry

The SymmetraPeak was bulky and expensive. When opamp circuits were developed which accomplished the same thing it faded away.

An example of an opamp equivalent is shown here as designed by Gary Blau, amateur radio callsign W3AM**:**

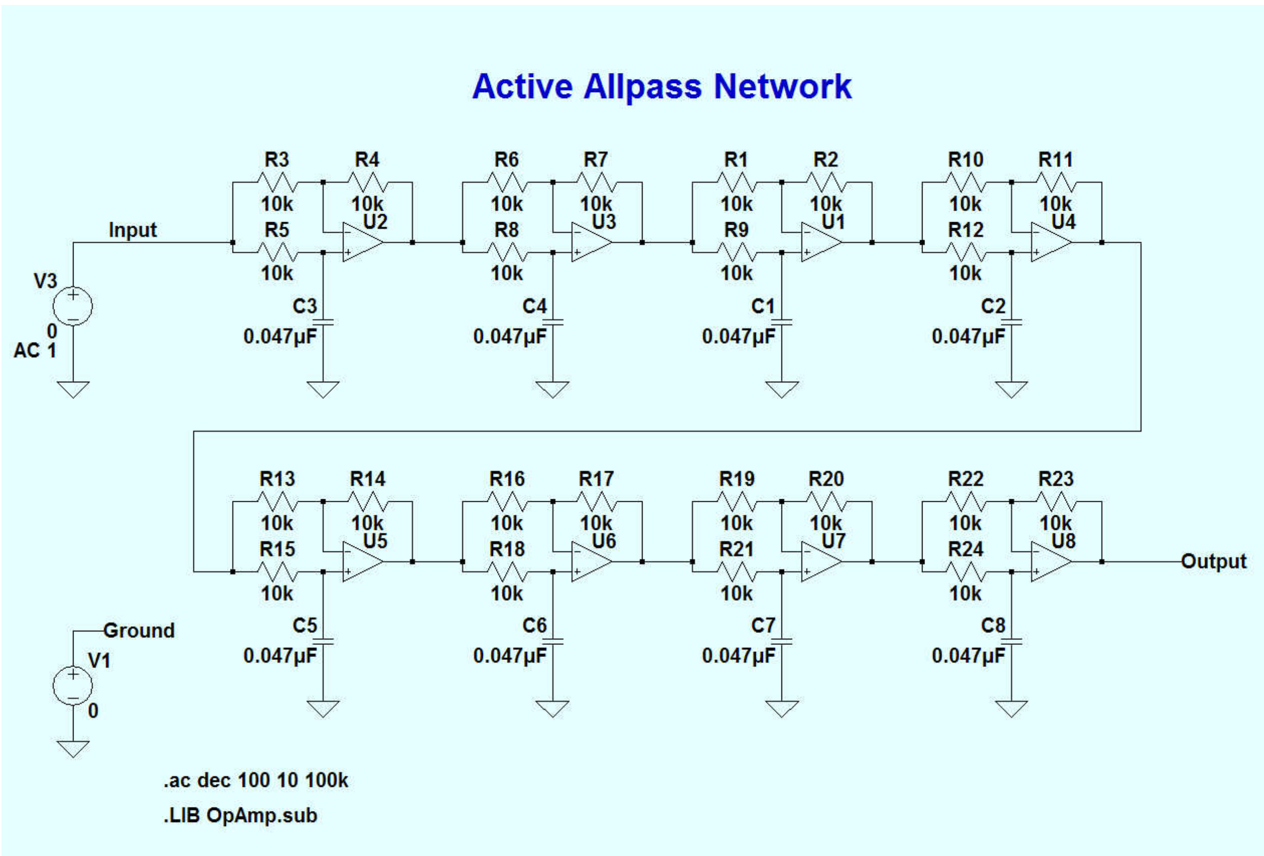

Fig 8 - An active allpass network

Our standard signal as it exits from that active allpass is shown here - even more symmetrical than from the SymmetraPeak:

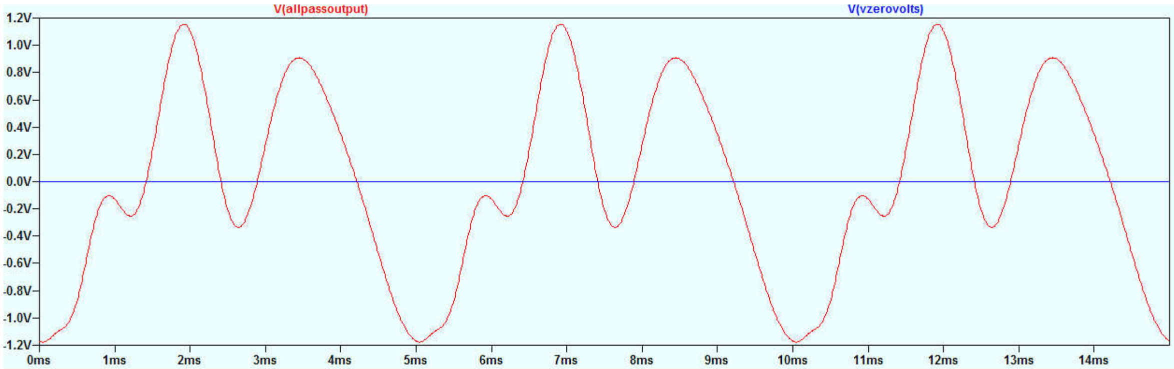

Fig 9 - Output of the active allpass network

The signal's 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 subaudible 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 ("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 as regards the AGC portion of the speech processing. But an RMS-sensing AGC unit *must* be followed by a clipper to catch those waveform excursions which escape the AGC unit. Be advised those excursions will be of significance. But if the clipper operates on one side of the waveform more than the other, a DC or subaudible 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.

Here again the allpass can come to the rescue. Insert the allpass between the RMS-sensing AGC block and the clipper. By adding the allpass at this point the clipper will clip symmetrically and no subaudible components will be involved.

Update: a *Tunable Allpass*

A ***tunable*** single-stage allpass network has been devised which may very well be suitable in most cases. The schematic is shown here as it underwent development in LTspice:

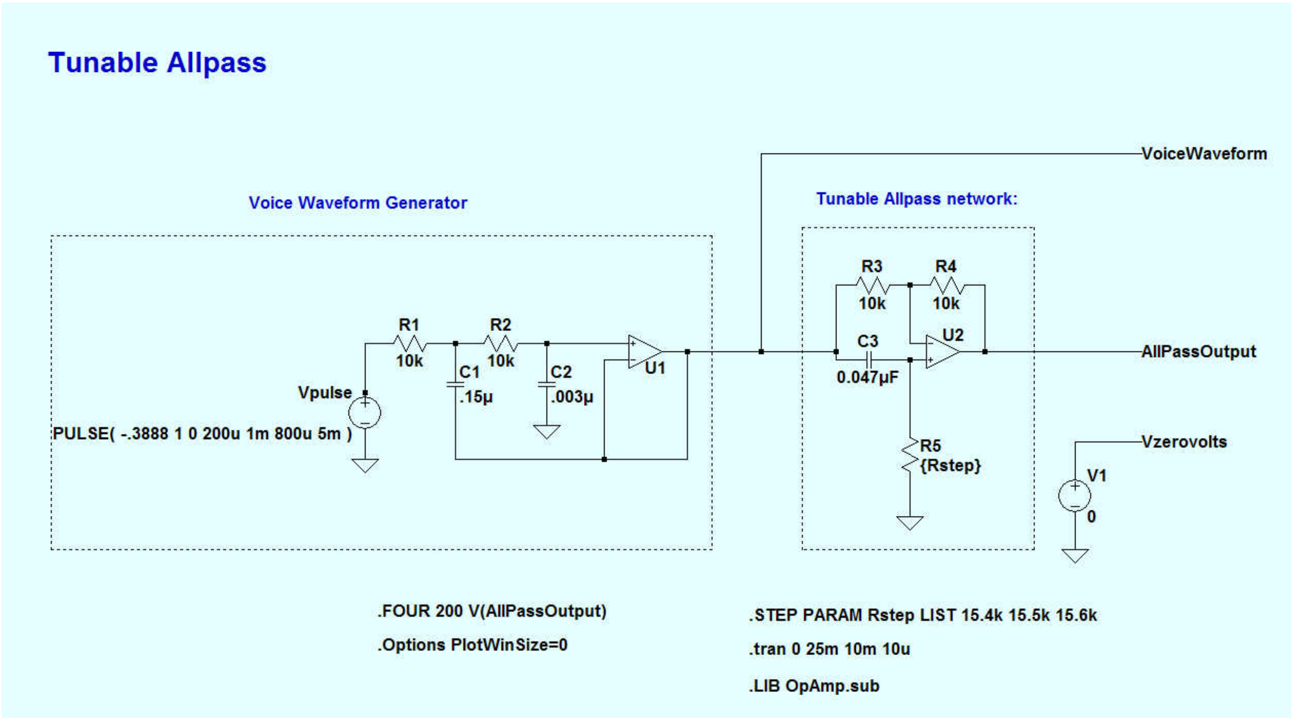

Fig 10 - Schematic of a tunable allpass network driven by the waveform generator

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. But this simple circuit does give a significant amount of performance as seen in these waveforms.

Here is the output of that tunable allpass circuit:

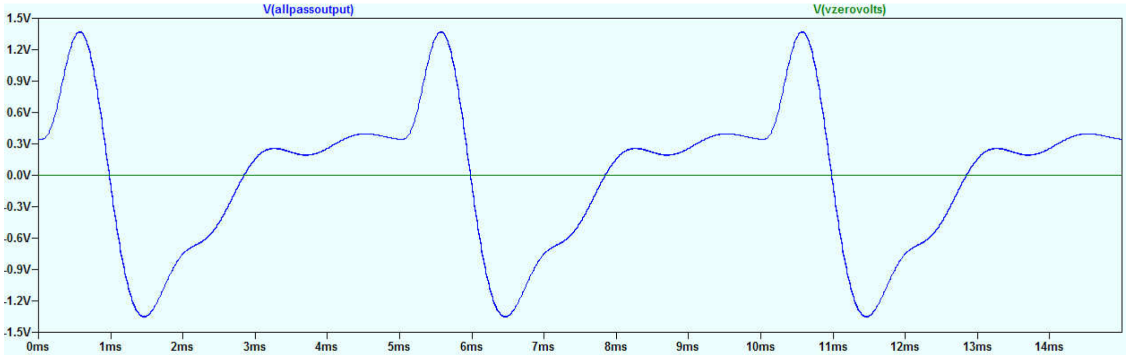

Fig 11 - Output of the simple tunable allpass network

The output from this allpass network is quite competitive 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 allpass network is not a cure-all. If the applied waveform is symmetrical top-to-bottom in the first place (i.e. has no even-order harmonic content) then the allpass can actually ***increase*** the signal's peak-to-peak amplitude. Here we see an example of an applied waveform with top-to-bottom symmetry**:**

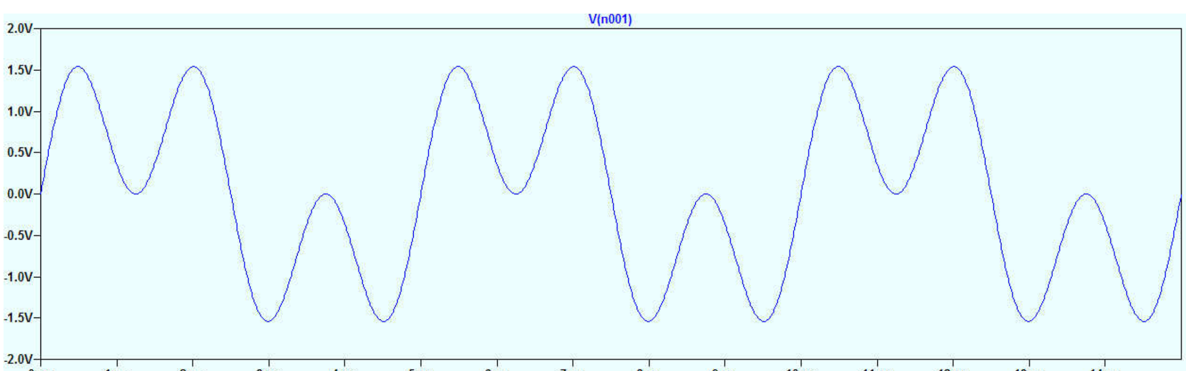

Fig 12 - A waveform with top-to-bottom symmetry

This signal consists of a 200 Hz fundamental and an equal amount of 600 Hz (third harmonic).

Now let us apply that signal to an allpass network. Here we have applied it to the Blau circuit. The signal at the output of that network appears as shown here**:**

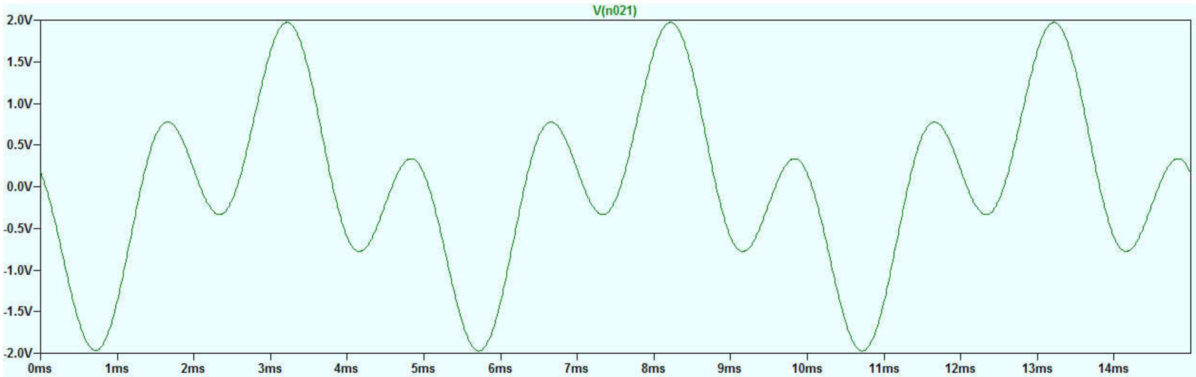

Fig 13 - Illustrating allpass failure

This signal still has top-to-bottom symmetry but the peak amplitude has actually **increased**. This can be seen by comparison of this waveform with the previous waveform (Fig 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 use of an allpass network might not be beneficial and in fact will be harmful to our effort.** This aspect of the allpass is usually glossed over and is being pointed out for completeness.

Suggestion

An allpass block should be placed in an appropriate point in a speech processor used in a radio transmitter**:**

It should be placed ahead of an AGC circuit which is essentially instantaneous-acting
or
it should be placed ahead of a clipper if the preceding AGC circuit is low-acting or especially if it is RMS-sensing.

The allpass network is inexpensive (if constructed using opamps and associated components) and generally insures lower distortion by virtue of less clipping or at least symmetrical clipping (which sounds less offensive to the human ear). Symmetrical clipping always causes less "mischief" than does asymmetrical clipping. Placement of the allpass 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 allpass network. The only other amateur radio writing on this subject seen by *ye scribe* has been by Gary Blau, W3AM.

Go to the W3AM site
Go directly to the W3AM allpass writeup