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External Processing for Controlled Envelope Single Sideband

It is now possible to separate the CESSB processing from the transmitter.

In my Nov/Dec 2014 *QEX* article on controlled envelope single sideband (CESSB), I stated that generation of the CESSB signal is best integrated into the SSB modulator of a radio, rather than being done in an external box.[1] It is possible to separate CESSB generation from a radio, however, if the radio SSB modulator is designed with this in mind.

The SSB modulator must be linear phase, and must have a bandwidth sufficient to pass the CESSB spectrum, including its spectral skirts. If an otherwise conventional SSB modulator meets these requirements, then the peak control obtained by the CESSB process will be preserved.

This will make it possible to use external processing to create CESSB. The radio may be used for conventional SSB if an external CESSB processor is not available.

The envelope control problem with single sideband is that limiting audio peaks does not accurately limit SSB envelope peaks. The envelope of an SSB signal is basically the vector magnitude of the modulating audio signal plus its Hilbert Transform. The Hilbert Transform is an audio phase shift of 90° for all frequencies within its bandwidth. The Hilbert Transform overshoots, making RF envelope amplitude control difficult.

CESSB is a way of controlling the inevitable RF envelope overshoots caused by the Hilbert Transform. These Hilbert Transform overshoots occur regardless of the method used to generate SSB. A phasing method SSB modulator produces a Hilbert Transform directly, by means of audio phase shift networks. Filter and Weaver method SSB modulators produce the Hilbert Transform indirectly.

[1]Notes appear on page 12

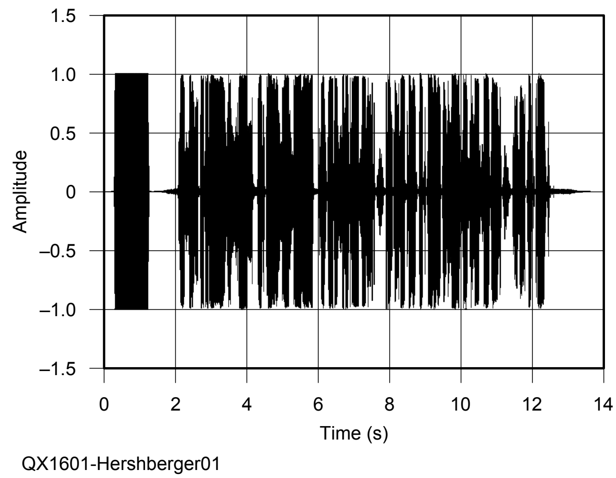

Figure 1 — An externally processed CESSB signal, reproduced by a linear phase Hilbert Transform SSB modulator.

If the envelope overshoots are not reduced, then ALC or manual transmit gain control will reduce the SSB signal amplitude, such that there is no flat-topping. This reduces average transmitted power.

Conversely, if the Hilbert Transform-induced envelope peaks are reduced or eliminated, then the average transmitted power of an SSB signal can be significantly increased. A 2.5 dB increase in average transmitted power is typical, compared with advanced look-ahead ALC systems.

Discussion

The intermediate output of the CESSB process is a pair of audio baseband signals. These are often known as "I" and "Q" signals, for in-phase and quadrature. If the I and Q audio signals are applied to a pair of mixers driven with quadrature RF, then the sum of the two mixer outputs will be SSB.

Another characteristic of the I and Q signals is that they are interrelated by a Hilbert Transform, or a negative Hilbert Transform. In other words, the audio signals are 90° out of phase between I and Q at all frequencies. In that regard, there is redundancy in I and Q.

One way to separate the CESSB process would be to pass the two baseband I and Q audio signals to a radio. It would be important to maintain accurate amplitude and phase matching for the two audio signals. It is not necessary to pass both audio signals into an SSB transmitter, however.

Because the two I and Q outputs of the CESSB system contain redundancy, you can throw one of them away and then regenerate it if necessary. The remaining signal has a special characteristic. The vector magnitude (or modulus) of itself plus its Hilbert Transform, is accurately amplitude limited. That vector magnitude function is proportional to the RF envelope amplitude of the SSB signal.

$$e(t) = \sqrt{a^2(t) + H^2\left[a(t)\right]} \qquad \text{[Eq 1]}$$

where:
$e(t)$ is the envelope signal
$a(t)$ is the input audio signal
$H[a(t)]$ is the Hilbert Transform of the input audio signal.

What Equation 1 suggests is that we could discard either the I or the Q signal, and pass just one audio baseband signal as $a(t)$ from the external CESSB processor to the radio. The radio could then regenerate the missing signal with a Hilbert Transform (either directly or indirectly). If this is done with linear phase and flat amplitude response, then the regeneration of the discarded signal will be perfect.

For this to work, the radio must have a linear phase response in its SSB modulator. That means flat time delay versus frequency. Also, the frequency response of the SSB modulator must be equal to or greater than the skirt bandwidth of the CESSB I and Q signals.

So, if the CESSB signal has a response of 300 to 3000 Hz, with descending filter skirts extending to 150 Hz at the low end and 3150 Hz on the high end, then the SSB modulator in the radio should have flat amplitude and linear phase from 150 to 3150 Hz. As long as those conditions are met, the radio will transmit accurately controlled envelope peaks using an external CESSB processor.

Unfortunately, most of the analog SSB transmitters in use today do not have linear phase response. A conventional radio with a crystal or mechanical filter for SSB generation might be wide enough, but it will have group delay peaks near the band edges. On the other hand, some SSB transmitters using DSP may very well have linear phase response. Those radios, if they exist, could be converted to CESSB operation with an external CESSB processor.

Simulations

GNU *Octave* is an excellent simulation and signal processing tool.[2] I have written some GNU *Octave* code that simulates the external CESSB system. My GNU Octave code is available for download from the ARRL *QEX* files web page.[3] The *Octave* script reads in an audio WAV file, which has been accurately amplitude limited. CESSB processing is done first. Next, one of the two baseband audio signals produced by the CESSB process is discarded. (Actually, the script uses a linear combination of I and Q to produce a single output signal. Any linear combination will work, such as I + Q, I – Q, 0.5 × I – 0.866 × Q, and other combinations). The remaining CESSB audio baseband signal is applied to the following modulators:

1) A linear phase filter type SSB modulator.

2) A linear phase Hilbert Transform SSB modulator.

3) A linear phase Weaver method SSB modulator.

Each of these modulators produces an upper sideband signal at 12 kHz. The sampling rate for all signals in the *Octave* code is 48 kHz.

The *Octave* code inserts a shaped 1 kHz tone, one second long, at the beginning of the speech audio. The purpose of the tone is to create an amplitude reference at the PEP limit of the transmitter power amplifier. A single tone does not create overshoot in any SSB modulator. (Simultaneous multiple frequencies are required to produce overshoot.) Note that the amplitude of the tone is a normalized 1.0 in each of the simulations that follow. If CESSB is accurately preserved, then the amplitude of the speech will not exceed 1.0 either.

All of these modulators reproduce the CESSB signal accurately, with tight envelope peak control. As a result, Figures 1, 2, and 3 look almost identical, even though different SSB modulation methods were used to create them.

SSB Modulators that *Do Not* Preserve CESSB

Next the same audio signal is applied to some inappropriate SSB modulators:

1) A nonlinear phase filter type SSB modulator, using a crystal or mechanical filter (such as a Heathkit SB-102, Collins KWM-2, and similar transceivers).

2) A phasing type SSB modulator (such as the vintage Hallicrafters HT-37 transmitter).

These SSB modulators, typical of analog SSB transmitters, introduce linear distortions to the CESSB audio baseband, and they overshoot. Accurate envelope peak control is lost.

The phasing-type modulator simulation

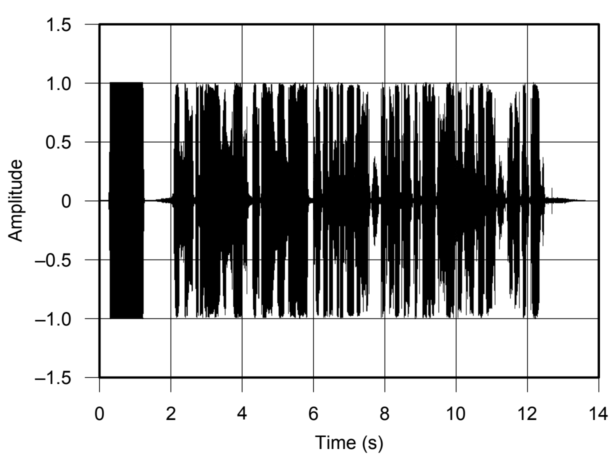

Figure 2 — An externally processed CESSB signal, reproduced by a linear phase bandpass filter SSB modulator.

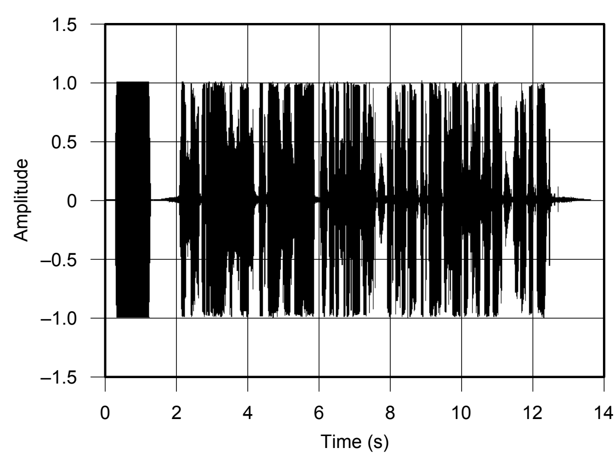

Figure 3 — An externally processed CESSB signal, reproduced by a linear phase Weaver SSB modulator.

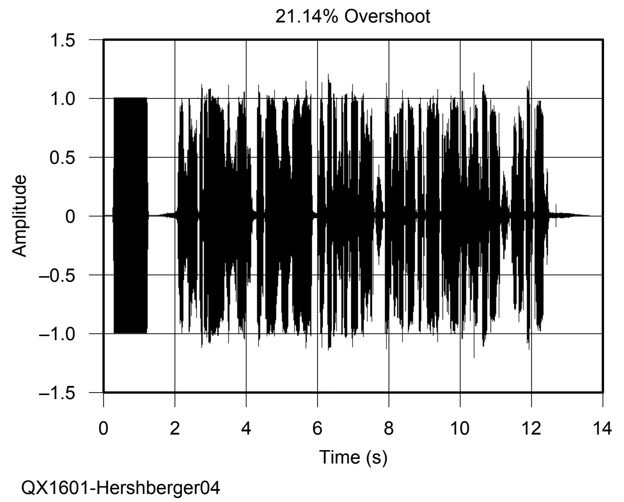

Figure 4 — An externally processed CESSB signal, reproduced by a nonlinear phase filter SSB modulator.

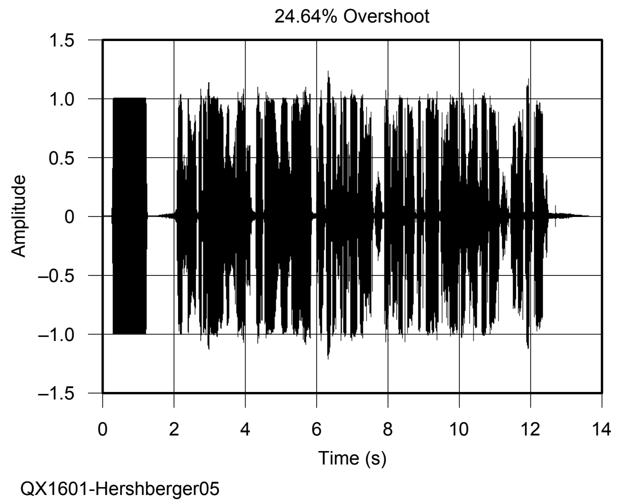

Figure 5 — An externally processed CESSB signal, reproduced by a nonlinear phase phasing method SSB modulator.

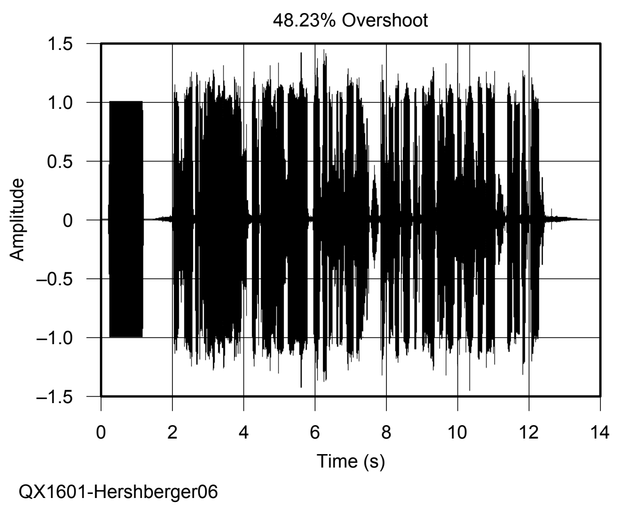

Figure 6 — A peak limited audio signal (not CESSB) applied to a nonlinear phase filter SSB modulator.

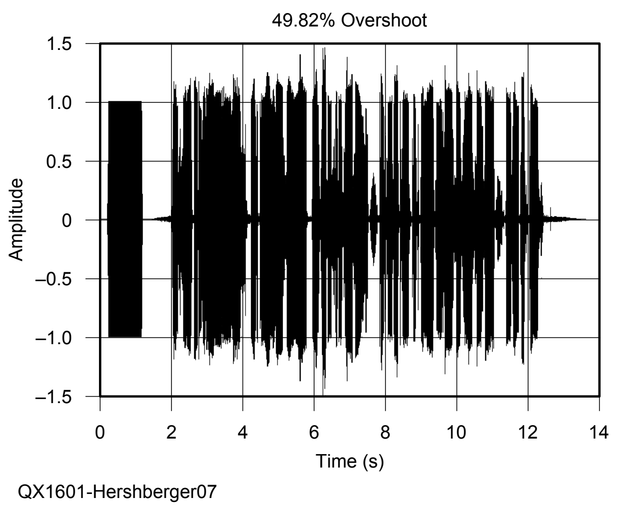

Figure 7 — A peak limited audio signal (not CESSB) applied to a nonlinear phase, phase-difference network SSB modulator.

uses the coefficient set II given by Theodor Prosch, DL8PT, in Table 1 of his Sep/Oct 2012 *QEX* article.[4]

A Hilbert Transform filter, referred to a compensating delay line, has a $d\phi/d\omega$ characteristic (phase slope) of zero. The phase shift remains at 90° for all frequencies. So, the group delay of a Hilbert Transform is also zero when referred to a compensating delay line. The compensating delay and the Hilbert Transform filter constitute a pair of phase difference networks. Their phase difference is 90° for all frequencies for which the Hilbert Transform filter is designed. Yet, there is no time delay variation versus frequency for either path.

But traditional analog or digital IIR all-pass filter phase difference networks do have time delay variations versus frequency and that is what makes a "phasing" type SSB modulator unsuitable for CESSB. The all-pass network pair has the following phase shifts:

$\Phi(\omega) + \pi / 2$, and $\Phi(\omega)$

So, it is the $\Phi(\omega)$ phase function that introduces phase distortion and causes overshoot in a phasing-type SSB modulator. Theodor (DL8PT) Prosch's Figure 4 shows the $\Phi(\omega)$ phase function. (See Note 4.) In a true Hilbert Transform modulator, the $\Phi(\omega)$ function is zero, however, a Hilbert Transform modulator requires more computation than a phase-difference network "phasing" type SSB modulator.

The minimum-phase, elliptic type band-pass filter does not work for CESSB because of its group delay variations. The same is true for the phase-difference network SSB modulator. It also has group delay variations.

Using CESSB Processing With Older Analog Radios

While the examples of Figure 4 and Figure 5 show some overshoot when used with CESSB-processed input audio, the overshoot is considerably worse with ordinary peak-limited audio. The same nonlinear phase elliptic filter SSB modulator, when driven from the peak limited audio (no CESSB audio processing) produces the RF envelope shown in Figure 6.

With CESSB audio processing, overshoot is 24.64% instead of 48.23%. Compare Figure 6 to Figure 4. So, even though there is overshoot, there is still some advantage obtained by using a CESSB processor in front of a conventional nonlinear phase filter-type SSB transmitter. With this example, RF power output would be about 1.5 dB greater.

Now let's look at the nonlinear phase, phase-difference network modulator. With conventionally processed audio instead of CESSB audio, Figure 7 shows the RF envelope.

Again, CESSB processing reduces the overshoot from 49.82% to 24.64%. Compare Figure 7 to Figure 5. So even though older nonlinear phase transmitters do not produce true CESSB output from a CESSB audio input, they do benefit from CESSB processing.

Phase equalization (in DSP) of the particular crystal filter, mechanical filter, or phase difference network could certainly reduce the overshoot of these older types of SSB modulators.

Is Your Rig "CESSB-Ready?"

If your rig is a FlexRadio 6000 series, it already has CESSB built-in.

If your transmitter is older or nonlinear phase, it can probably partially benefit from CESSB audio processing.

If you have a modern DSP based transmitter, it might already be fully "CESSB-ready." To find out, you just need to connect a CESSB processor to its audio input and then look at the RF envelope on an oscilloscope.

As of this writing, there are no external CESSB processors available in hardware, but there is still a way to test your rig. The WAV files used to generate the figures in this article are available from the ARRL *QEX* files website. (See Note 3.) All you have to do is play the WAV file (CESSB-ready-test-audio.wav) into your rig and look at the RF envelope coming out. Here are some suggestions:

1) Turn off any equalizers, audio compressors, or other audio processors.

2) If possible, turn off ALC.

3) Run the transmitter power down to about 25% of normal by reducing audio (mic) gain, so you can see any overshoots.

4) If your transmitter has adjustable transmit bandwidth, increase it to about 3.5 kHz or more.

5) Use a dummy load! The audio test files contain my call sign, and you wouldn't want to misidentify your station!

The WAV file contains the reference tone as a maximum PEP reference. If all of the speech peaks stay at or below the reference tone amplitude and look like Figures 1 to 3, congratulations, your rig is CESSB-ready! If the voice peaks visibly exceed the reference tone, and look like Figures 4 through 7, then your rig is not CESSB-ready, but it still may benefit from the use of a CESSB audio processor.

You may also wish to test with the peak-limited-audio.wav file. This file does *not* contain CESSB processing. It only contains simple audio peak limiting. This file will cause SSB modulator overshoot.

The file externalcessb.m is the GNU *Octave* script. Externalcessbmc.m is an edited script that is compatible with Matlab®. Both scripts will create many plots of SSB envelopes, spectra, and filter characteristics.

Conclusions

Although the most convenient way to generate CESSB may be to build it into each radio, CESSB processing can be done with an external box, and radio manufacturers could make radios that are "CESSB-ready." If you just plug in a mic, you don't get CESSB. You get plain old SSB. If you have an external CESSB audio processor, however, then you will get CESSB from a radio that is "CESSB-Ready." Some of the modern DSP rigs might already be "CESSB-ready." Many older analog SSB modulators are not going to preserve CESSB, since they are not linear phase.

If radios that are "CESSB-ready" are made, along with external CESSB processors, then hams will have the option to "mix and match" processors and transmitters. As speech processing algorithms improve, the external CESSB processor can be replaced or upgraded, and the same radio can continue to be used.

The CESSB processor-to-radio interface is a single audio signal. The audio signal path needs to be flat amplitude and linear phase. The SSB modulator also needs to be flat amplitude and linear phase.

Although nonlinear phase transmitters cannot fully preserve the CESSB signal, they do obtain a partial benefit from external CESSB processing.

Dave Hershberger, W9GR, was first licensed in 1965 at age 14 as WN9QCH. He is an ARRL Life Member. Dave holds a bachelor's degree in math from Goshen College and bachelor's and master's degrees in electrical engineering from the University of Illinois. He has been awarded 19 US patents. Dave is Senior Scientist at Continental Electronics. His recent projects include two ATSC digital television broadcast exciters with adaptive linear and nonlinear precorrection, a DSP based FM/HD Radio® exciter with adaptive precorrection, new high power uplink transmitters for the JPL/NASA Deep Space Network, and a 2.4 megawatt VLF transmitter system.

Notes

[1]David L. Hershberger, W9GR, "Controlled Envelope Single Sideband," *QEX*, Nov/Dec 2014, pp 3 – 13. You can download a copy of this article at: **www.arrl.org/files/file/QEX_Next_Issue/2014/Nov-Dec_2014/Hershberger_QEX_11_14.pdf**

[2]There is more information about GNU *Octave* on the *Octave* home page at **www.gnu.org/software/octave**. You can also download the latest version of GNU *Octave* from that website.

[3]The GNU *Octave* files and WAV files are available for download from the ARRL *QEX* files website. Go to **www.arrl.org/qexfiles** and look for the file **1x16_Hershberger.zip.**

[4]Theodore A Prosch, DL8PT, "A Minimalist Approximation of the Hilbert Transform," *QEX*, Sep/Oct 2012, pp 25 – 31.