The Timewave DSP-59 Audio Filter by Nick Hall-Patch with John Fallows

Commercially available outboard audio filters to enhance reception have been around for many years. There have been a variety of opinions on how useful they may be, which vary from “the best filter is between your ears” to serious studies of how a listener gathers information from an audio signal, followed by filter designs based on these principles (see the references found in the article below on the “CommAudio Processor”).

Presently I am using a homebrew continuously variable frequency (250-1000 Hz) two pole active high pass filter in series with an old Autek QF-2 which features switchable low pass, notch and peak facilities with continuously variable frequency (250 to 2500 Hz) and selectivity (from flat to 50 Hz). They are used between the receiver detector and its audio output amplifier and are a good example of op-amp based analog technology which was first available to the consumer 20 years ago.

More recently, digital filtering techniques have become available to the radio amateur and SWL/DXer in such units as the Watkins-Johnson HF-1000 receiver, and in the Timewave Technology DSP-59 Audio Noise Reduction Filter. Like many other commercial audio filters, the DSP-59 is meant to be used between the speaker output of a receiver or transceiver and the speaker itself, and is powered by external 12 to 16 volts at 1 ampere.

Like earlier filters, the DSP-59 has high and low pass filters, but these are not continuously tunable. The high pass filter has eight selectable cutoff frequencies between 200 and 1500 Hz, while the low pass filter has eight cutoff frequencies between 1.8 and 4.2 kHz. In addition, there is a bandpass function which provides 8 center frequencies between 400 and 2210 Hz with 8 selectable passbands between 50 and 600 Hz, primarily intended for CW and RTTY reception. All of these filters have a very sharp roll-off outside their passbands: 60 dB attenuation within 180 Hz of any of the low pass filter cutoff frequencies, somewhat less attenuation with high pass filters, while the bandpass filters have somewhat better out of band attenuation than the low pass filters.

In digital filters, an incoming analog signal is sampled and converted to its digital equivalent signal strength at fixed and frequent time intervals (at a rate at least twice the highest frequency being sampled). The filters described above are “finite impulse response” (FIR) filters which are essentially computer programs that process the most recent digital information along with other digital information from previous analog to digital conversions. The modified digital signal is then converted back to an analog signal which makes sense to our ears. A primary advantage of this sort of filtering is that extremely sharp roll-offs past the cutoff frequency are possible without the “ringing” sound associated with the phase shifts inevitably associated with sharp analog filters. DSP filters tend to be more effective with faster processing speed, as higher quality filtering algorithms using more steps can be used with higher processing speeds. One reason why more effective DSP filters are coming onto the market is that higher speed DSP chips are becoming available.

The DSP-59 has two functions called “Random Noise Reduction” and “Tone Noise Reduction”. These are quite different from anything found in analog filtering. In the manual, they are explained in the following terms:

> The noise reduction functions of the DSP-59 operate by examining a characteristic of signals and noise called correlation, and dynamically filtering out the undesired signals and noise. The degree of correlation is relative. Random noise such as white noise or static is uncorrelated. Speech is moderately correlated. Repetitive noise such as a heterodyne is highly correlated. The DSP-59 measures correlation and filters out signals and noise that are outside its correlation thresholds. There is little degradation of the desired speech signal. The amount of noise reduction varies according to the correlation characteristics of the noise. Typical noise reduction ranges for 5 dB to 20 dB for random noise and up to 50 dB for heterodynes.

Although the manual doesn’t go into great detail, it seems that in the Random Noise setting, samples of digital data which have no similarity (correlation) to previous samples are eliminated, while in the Tone Noise Reduction setting, digital samples which are identical to previous samples are eliminated. In both random noise and tone noise settings, “aggressiveness” of the noise filter is controlled in eight steps; in the more aggressive setting, it seems that more signal is thrown away in trying to eliminate the interference. These settings also include a variable low pass and a fixed high-pass of 200 or 300 Hz.

So, how does it sound? I compared this unit with my filter set-up described above, and fed each filter with signals from my SONY ICF-2010 and from my homebrew receiver (described in IRCA reprint M39).

Heterodyne reduction: I created a het on a signal on 1490 kHz by tuning a portable radio to around 1055 kHz (so the LO would be at around 1490 kHz) and moving it towards my loop antenna until the het started to distort audio recovery of the desired signal on my homebrew receiver. With both a 500 Hz and a 2000 Hz heterodyne generated using this method, the DSP-59 gave a marginally better sounding signal with no sign of a heterodyne and minimal apparent distortion of the desired signal. This was using the C/LP NRt (tone noise reduction) setting with a correlation setting of 2. The QF-2 gave a somewhat more “woolly” sounding desired signal, but the hets could also be reduced to inaudibility. The best method of heterodyne elimination was detuning the receiver so that the het was outside the IF filter passband, or by using synchronous detection to demodulate the sideband unaffected by the heterodyne.

The low pass filter without noise filtering also worked very well (i.e. very sharp, with minimal ringing) at removing high frequency heterodynes (over 1.8 kHz), but this was only observed using the 2010 in its wide AM position. The IF filters in the homebrew receiver mean that heterodynes much above 2.2 kHz are not a problem, so the low pass filter is not really needed for heterodyne reduction. The audio remaining after the heterodyne was filtered out by the DSP-59 was crisper sounding than when the QF-2 was used to eliminate the same het. This is obviously a function of the steeper roll off possible with the DSP-59, but I suspect that a relative lack of phase shifting within the audio passband may have contributed to this effect.

The high pass filter is specified as not being as sharp as the low-pass, and without the noise reduction in place, it was necessary to place this filter in its 500 Hz position to eliminate a 300 Hz het. It still did a better job of eliminating the het than my 2 pole highpass filter did.

Noise reduction: I wanted to try this filter on electrical noise, sideband splatter and on static crashes. Using the QF-2 in its low-pass mode along with my high pass filter is my usual method of dealing with these irritations, but the filters just make my listening less fatiguing; they don't make the desired signal more readable. (John Fallows:) I think there is a range where AM signals are made more readable. With a very marginal AM signal, i.e., SNR maybe below 4-5 dB or less, readability is not much improved. However, with AM signals closer to the 10 dB SNR threshold and slightly above, there is an improvement in readability. Once we have good SNR, say of 20 dB plus, there is again not much benefit. The main advantage of the NR mode is for SSB reception, which I believe can make an 80m LSB signal during summer static crashes, for example, sound almost as good as NFM on the local repeater.

Listening to a splatter or noise plagued signal using the DSP-59 in its HP/LP NRr (with the fixed random noise filter) and its C/LP NRtr (with the adjustable random noise filter) settings gave similar results to my present setup. For some reason, the straight low-pass/high pass filter was quite harsh sounding, as if the 1.8 kHz low-pass was not low enough (yet I know it was accurate due to my earlier heterodyne testing); the same harsh sound was noted on the C/LP NRtr setting. These settings were not as pleasant to listen to as my own filters were.

On some types of buzzy electrical noise, the DSP-59 made the signal more readable when the HP/LP NRr or C/LP NRt settings were used, as the noise was virtually eliminated from the audio output. However, the C/LP NRtr setting was better for reducing the intensity of static crashes on MW and on the 75 meter amateur phone band.

Other features: I found the bandpass filter very useful in receiving multiple CW signals, especially when using the ICF-2010 with its relatively broad (for CW reception) narrow IF filter. Signals could be picked out by setting the DSP-59 to a 400 Hz center frequency with a 200 Hz passband (or less), setting the 2010 to LSB/CW and slowly tuning it. The signals just popped up, then disappeared with the next tuning step. The bandpass filter can also be used in this way to DX longwave beacons. By slowly tuning the receiver (with its BFO on) and using a fixed setting of the DSP-59's bandpass filter, one can hear the beacon's sideband (which contains its ID) pop up, then disappear.

Such techniques are not as straightforward on the QF-2/high-pass combo, and the QF-2 is prone to much more ringing than the DSP-59 when its selectivity is tight.

Further comments and conclusions: The DSP-59 has a red LED indicator on the front to tell you if its audio input is overloading the unit. One should keep an eye on this as it is necessary to limit the input amplitude to allow the unit to work properly. Actually, this is also true of analog filters, but they don't usually have the warning indicator.

The bypass setting of this unit does not appear to be a true bypass. Instead, it appears to perform an analog to digital conversion on the signal, and then convert it back to analog without any further processing. Although I didn't find the audio output too much different from the input when using the bypass setting, it may be a cause for concern. If filtering isn't needed, it's probably best to process any signal as little as possible to retain readability.

I found that the DSP-59 made DXing rather more comfortable for me than my present audio filter does, especially when the signal was bothered by certain kinds of electrical noise and splatter. However, it did not dig out any identifications from anything previously unreadable. I'm not sure that any audio filter would do that when one is already using a receiver with reasonable IF selectivity and demodulation options.

The DSP-59 (which was priced in the US$200 range) has been superseded by the DSP-59+ and DSP-599zx (which a couple of commentators found to have a more "natural" sound in SSB listening, but otherwise not much of an improvement over the 59+). The 59 may be still available at some dealers, and will likely appear on the second hand market. The DSP-59+ costs more than the DSP-59 did and uses a new algorithm for het removal, and has many more bandwidths available as well as sharper CW bandwidths. Further information can be obtained about this and the upgrade of the DSP-59 to the 59+ from Timewave Technology Inc., 2401 Pilot Knob Road, St. Paul, MN 55120. (612) 452-5939. Distributors include Ham Radio Outlet, Amateur Equipment Supply and Texas Towers.

Other manufacturers provide DSP audio filters, including JPS (the NIR10 and NIR12 which seem at least competitive to the above; also the simpler NRF-7), MFJ (the 784B) and Quantics (the W9GR DSP filter kit which is mostly radio amateur oriented but has voice filters as well). Radio Shack sold the inexpensive DSP-40, but unfortunately, only its notch filter was really effective.

(John Fallows:) Additional comments on Timewave 59+ (not the 59) based on my use:

- Excellent for ham digital work, including the digital regeneration feature.
- I use a line input/output rather than speaker input, because I want to distribute processed audio to other equipment through a switchbox, i.e., to digital decoder, tape recorder, audio amplifier, etc. The line in/out works fine, except that line output level cannot be adjusted.
- There is an “artificial sound” to the audio when doing the noise reduction, not too bothersome, though.
- I would recommend people go straight to the 59+ as it does have continuously tunable high and low pass filtering, etc.
- With the ROM upgrade, random noise reduction can be adjusted from the front panel; the original model required switching an internal jumper.
- Tone noise reduction provides multiple automatic notch filters - much more useful (and effective) than the traditional AF or even IF based manual notch.
- My main concern with using this unit (and outboard audio DSP in general) is that it takes place OUTSIDE the receiver’s AGC loop. On the 59+, using the AGC tends to introduce pumping, and none of the AGC characteristics are controllable by the user. Dynamic range of these filters is limited, so using AGC is pretty well mandatory, but the receiver AGC and the DSP AGC sometimes work against each other. Hopefully, as DSP is introduced at the IF level in receivers, this issue will get sorted out. (Interestingly, it was the AGC problems which drew the biggest criticism in the original Watkins-Johnson HF1000 release.)