IRCA TECHNICAL COLUMN
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Using a PC sound card for SSB/CW/AM demodulation

If you've been in the market for a receiver in recent years, or have just been dreaming about it, you're aware that a number of the more expensive units extol the virtues of Digital Signal Processing (DSP). This is simply the use of computer algorithms to perform functions, such as bandpass filtering, which have formerly been taken care of by analog circuits.   In such receivers, the analog signal is sampled at some point in the signal path, usually later on in the IF strip, and converted to digital data, whereupon a dedicated DSP microprocessor performs various computations on the data which can filter it and demodulate it to baseband (what we in the real world would regard as equivalent to an audio signal). Then the processed data is converted back to audio, so that we can actually listen to it. The advantage of DSP is that simply by changing variables in a program, you can obtain a greater number of functions, such as different filter bandwidths, than you could ever afford to do using hardware. A programmer employed by the manufacturer figures out ways that you can control those variables without your knowing anything about DSP.

The TenTec RX-320 is an example of a receiver which uses DSP to perform many of its functions, and it isn't too expensive, but some of the stand alone receivers with DSP are quite pricey. However, you don't have to break the bank to experiment with DSP, though you will need some technical skills, and some familiarity with the generic mixer and volume controls for your PC sound card.

Until fairly recently, dedicated DSP microprocessors were the only way to perform real-time DSP on any but the slowest digitized signals. Ordinary computers would not have been fast enough, especially as the PC processor is not optimized for the "multiply and accumulate" instructions needed for DSP. However, even the cheapest new computer boasts clock speeds well in excess of 1 GHz, so, in fact, a "normal" PC can easily perform near real time DSP instructions on data which is derived from well above audio frequencies. In addition, the standard computer is equipped with a sound card that includes an analog to digital converter for incoming signals of up to 22 kHz and a digital to analog converter with a similar frequency range.   Combine that with a reasonably fast
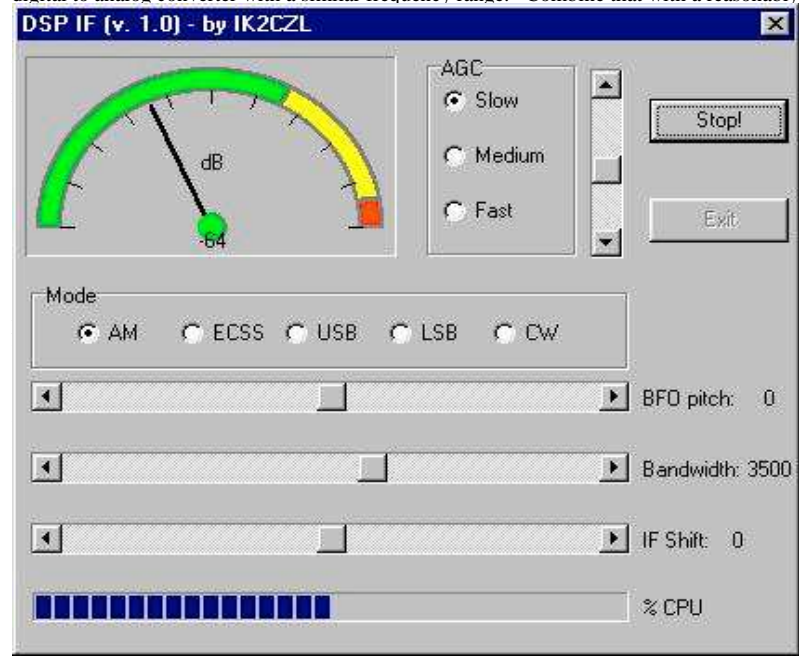
get inside your receiver and create a buffered 100 mV output from its IF stage.

Figure 1 shows the software's control panel. As Vittorio describes on his web site, the software provides:

- a level meter for measuring the IF signal level with 1 dB accuracy and 100 dB of range.
- an AGC control with three different time constants
- a volume control
- two buttons for starting/stopping/exiting the program
- a mode selector: AM is demodulated using a true square law detector, while the SSB and CW modes use a product detector. ECSS is "enhanced carrier selectable sideband", for superior demodulation of AM signals. When ECSS mode is selected, a self-synchronized notch is also inserted, which takes care of removing any interfering carrier(s) present on the frequency you are receiving.
- a BFO control allows adjustment of the BFO frequency with a resolution of 1 Hz
- a bandwidth control changes the selectivity of a group of cascaded filters from a minimum of 10 to a maximum of 6000 Hz.   This bandwidth is continually tunable; these are not discrete steps.
- an IF Shift control allows the user to detune the filter center frequency, and enhance signals present in one or other of the sidebands.
- an indicator shows you how much CPU time your PC is using for DSP functions

 To hear the output audio from this program, you use headphones or amplified speakers connected to the audio output jack of your sound card.

Figure 1 -- the IFDSP control panel

y months ago, I was fortunate enough to get together with him and try out his software using the 455 kHz outputs from an AOR AR-7030 receiver, as well as from my homebrew receiver.   Although I have mentioned fast PCs in connection with running this software, Vittorio runs it on a laptop with a 166MHz Pentium and Windows 95; it runs even more easily on faster PCs.   Although he prefers to run the IF signal to the line input of the sound card, we were able to make it run successfully on the microphone input of a laptop which lacked a line input. One just has to be careful not to overdrive the sound card input with too much signal, and to be assured that there is no DC voltage provided by the microphone input, as that could damage the converter.

Although this software is a work in progress, the results were quite impressive.   We were able to get at least as good quality audio from the software as the receivers alone provided, and on signals that were bothered by noise or interference, we were able to use the flexible controls to occasionally tease a bit more readability out than that of which the receiver alone was capable.

If you don't have IF shift or passband tuning on your receiver, find that you would like more IF bandwidths than your receiver provides, or an accurate signal strength meter, this may be a way for you to inexpensively add these functions to your present receiving setup.   Vittorio is interested in improving the "IFDSP", as he points out that he is only using a fraction of the PC processing power available to him to provide the present features.   Although he is planning on improving the shape factors of the present IF filters, the really interesting future development will be in providing superior demodulation schemes to that which we are used to in our present receivers. This may allow us to hear signals that are presently unreadable due to noise and interference.