Fldigi Sound Level Output Control

To achieve a wide range of levels, while maintaining the quality of the output, we need to use not only digital attenuation built into Fldigi – Scaling the waveforms mathematically – but also the analog attenuation that is built in to the soundcard. The method of scaling the linear output of Fldigi incorporates the following (Per Dave Freese – W1HKJ):

> *'All of the Fldigi internal sound card signal handling is done in floating point (actually a C++ double variable – 16 bits). Every signal that Fldigi produces is designed to be no greater than ± 1 peak-to-peak. This corresponds to the maximum D/A input for the soundcard (or transceiver codec) being used. The TX attenuator is the last amplitude factor applied to the generated modem signal before being sent to the codec.'*
>
> > *Att = 10^ (val / 20) where val is the TX attenuator setting in dB.*
> > <u>*for val = -10, Att is 0.3162277...*</u>
> > *for val = -3, Att is 0.70794...*

As you can see, the peak value for a 10 dB reduction (a 100 X *power* reduction) will reduce the range of the generated audio sample amplitudes from 65,534 possible values to something on the order of 654 possible values (or a reduction in resolution to <u>less than 1%</u> of the original 'source' signal). This may not be apparent when listening to the audio output, but in terms of accurately modulating (and demodulating it on the receiving end) the intended transmitted waveform, this is significant! Although the output waveform still has the possible set of values at 0 dB attenuation (65,534) the output waveform is actually using only a set of only 654 values due to digital amplitude reduction (at -10 dB output attenuation). Further digital attenuation only makes the matter worse as fewer and fewer values are being used!

Digital Attenuation

For example, imagine that your level controls are set such that a full scale digital sound plays out at 0 dB. If we wish to then play out a sound at -10 dB, you can do this in one of two ways. We could digitally attenuate the sound. If the initial (16 bit audio stream) waveform is varying from -32767 to 32767 (16-bit PCM input stream to soundcard From Fldigi), then to get a -10 dB sound level, we'll need to reduce it by a factor of 100. It will then vary from -327 to +327. However, as we're now using a smaller (digital) range, the waveform will be much more limited (due to the restricted digital range) and there will be substantial quantization noise (non-linearity) introduced. This problem is particularly pronounced when we wish to use wider bandwidth signals in conjunction with pure ones, as we'll frequently find that to stop things from distorting (or overloading the TX input), and get the levels right, the tones cover a fairly small range to start with. Again, the 'resolution' of the signal is still 65,534 possible values, but limited to a set of 654 values due to digitally reducing the amplitude.
<u>*Extra digital attenuation then reduces the quality unacceptably.(for wide bandwidth audio!)*</u>

Analog Attenuation

A better alternative is to use the analog attenuation that is built in to the soundcard device. You can tell the soundcard to reduce the level by the appropriate amount, and then there is no need to digitally attenuate the sound by such a large amount that it causes distortion. You're still using the full 16 bit scale, and no extra quantization noise is introduced.
This is accomplished by using the Windows Mixer Controls and the transceiver USB device input attenuation (or gain control).In practice, while the actual levels offered by the soundcard cover a wider dynamic range, they do so in fairly crude steps, so we'll need to use (Sound Control Applet) the analog

attenuation to get close, and then a little extra digital attenuation (fine control) to get exactly the level we require. It is recommended that the Fldigi TX output not be reduced by more than 3 dB from the 0 dB level. In the TS-990S, TS-590SG (and other similarly USB equipped transceivers) this input has very high 'gain' and wide input range (*microphone level sensitivity*) and must be greatly reduced! In the case of the Kenwood radios, it is not uncommon to use input gain values of less than 10 (on the TS-990S) or gain value of '1' in the case of the TS-590SG. Do not be concerned that this control is not 'centered' in the digital middle. The aim here is to ensure a clean undistorted signal is transmitted. This will mean that if your audio chain is properly adjusted, using an un-attenuated output form Fldigi (or any other digital modes software) will never cause ALC action to occur and will result in your TX signal being as linear as possible. Although you could reduce the audio level to control the RF output, the better option is to use your transceiver's 'drive' or 'power' adjustment so that the best linearity is maintained.

Analysis

Although in the purist interpretation of the above explanations we would like to be able to use the highest 'resolution' audio, the reality of actual implementation in use is determined by the components and their individual adjustability in the audio transmit chain. Some newer digital only interfaces (built-in USB audio or 'virtual' sound card devices) may only allow for 'digital' adjustment of levels to the transmitter. These devices will have limited ability to reduce the audio level without introducing a reduction in 'resolution' of the source signal.

Some newer equipment with the integrated USB capability may have a range of adjustment from an arbitrary 1-to 100 scale which may be either linear or logarithmic so that adjustment covers a very wide range of input capability from microphone level (say -40 dBm to -20 dBm – line levels) to well above the 'standard' line-level input level (0 dBm into 600 ohm load) or beyond.

Fldigi does provide for a wide range of attenuation from 0 dB to -30 dB. This range of levels is far more than needed to cover almost any situation. I have tested the resulting transmit audio waveform with the level at both 0 dB attenuation and at -20 dB of attenuation with no discernable artifacts from significant digital calculation attenuation, but we should be careful to fully qualify this statement!

Although the digitally calculated and attenuated audio level does contain all of the possible resolution of the 65,534 values (16-bit) when the signal is so attenuated that its maximum value is less than 1/100th of the calculated 'full maximum' (this is 'only -10dB) we are left with a signal of a maximum resolution of ±327 digital increments. For an audio bandwidth of our transmitters of about 3 KHz this may never be an issue. Therefore we can use either method of attenuation (analog or digital) without noticeable effects. If however we choose to use less than 3 dB of attenuation, we still retain about 32,767 possible values of resolution. Obviously this is the more desirable situation.

Linearity Issues

From a standpoint of attempting to retain the highest 'resolution' audio signal, we should attempt to use the least amount of digital attenuation (keep Fldigi TX attenuator as close to 0 as possible). Practically we wish to keep our digitally controlled (Voltage Controlled) analog amplifiers in the sound card as linear as possible also. This is a delicate balancing act that each operator must adjust based on what control is available and where to apply such control in the audio chain on its way to the modulator. Capability-wise, the 16-bit resolution will provide fidelity to beyond 20 KHz with excellent linearity, with a limited voice bandwidth, 654 values should also provide excellent results within the 3KHz modulator filtering (low-pass filtering) we are permitted to use (by strict definition – 2800 Hz).

Use what you have available, keep the levels tightly controlled so that your signal is linear (no ALC or limiting action, definitely no compression or TX equalization) and your resulting signal will be easy to decode and you will not interfere with the next signal up or down the waterfall.

When using PSK31, you should enquire if a strong station reports you as ‘599’ (or ‘59’ if using RSQ) and reports that your received IMD is above -20 dB! Fldigi and other programs cannot calculate the IMD of the received signal on random text, but rather require the ‘PSK IDLE’ pattern to accurately measure the IMD. To transmit the ‘PSK IDLE’ signal from Fldigi, simply use the ‘PTT” button or your <TX> macro. Remember to return to the RX condition by the use of the <RX> macro tag, hitting the ESC character twice, or clicking the PTT button again.

Try this measurement with a local station and adjust your audio levels as above for NO ALC action and you will find that even with a less than ideal antenna, you can make quite a few contacts at reduced TX power! If your RX PSK IDLE at your receiving site is at or near -30 dB you are probably at the best your station can do! Your other hams on the band will be glad you don’t have a ‘wide’ signal, and your signals will be much easier to decode.

Don’t forget the Receive Gain

The same is true of your RX levels. If the level into the sound card exceeds the linear range of the analog to digital converter (ADC), the algorithms in the receive chain will have a hard time decoding the signal. It will also make it difficult to decode the lower level signals being ‘swamped’ by the ‘louder’ overloading signal in the passband. Any signal driving the receiver circuits to the point of overloading will cause problems. It is best to reduce the level just a bit than to have the ‘perfect’ level close to maximum gain.