http://www.w1hkj.com/FldigiHelp/audio_adjust_page.html

Screen shots vary slightly according to Windows version you have.

Receive audio

Setting the correct hardware, operating system, and fldigi received audio levels is not difficult, but it is the one setup procedure most often done incorrectly. The most commonly used sound card devices contain either a 16 or 24 bit analog to digital (a/d) converter. A 16 bit a/d can provide approximately 90 db of signal conversion. For the 16 bit converter, if the peak audio signal that the a/d can handle is +/-1 volt then the minimum discernable signal (1 bit) will be +/-30.5 microvolts. If more than a +/-1 volt signal is applied to the a/d input then either one of two things may occur, (1) the audio is clipped, or (2) the audio is wrapped, large positive signals wrap to large negative signals and vice versa.

The objective in adjusting the Rx audio is to use the full dynamic range of the a/d without incurring overdrive. fldigi provides a number of display controls to assist in setting up the Rx audio. There are two controls that ONLY adjust the visual appearance of the waterfall and DO NOT affect the a/d or the signal decoders. These controls are below and to the left of the waterfall, "Upper signal level (dB)", and "Signal range (dB)".

		ALC: NO.
0	60	

Waterfall level/range controls

Change the waterfall display to the "Scope" view. Do this by pressing the "WF" button twice if it is currently displaying the waterfall. You can also right click once on that button. That button acts as a rotary and is left/right click sensitive. The display should show what looks like an oscilloscope view of the received audio. This is the entire audio signal and not just the signal that is currently decoded.



Scope view

If your sound card a/d (also referred to as codec) is functioning correctly there should be no signal offset and the signal will be centered vertically about the y-axis. Set your transceiver to a portion of the band with digital signals, 14.070 MHz is a good spot. Now adjust the transceiver audio output and the operating system mixer controls so that the peak signals do not exceed the

upper/lower gray lines on the display. The diamond indicator to the left of the fldigi AFC button will be colored as follows:

- BLACK no signal, or insufficient Rx audio
- GREEN signals are in the correct range
- YELLOW signals are exceed 75% of maximum, but are less than 90% of maximum
- RED signals exceed 90% of maximum a/d capability WARNING WILL ROBINSON!

You can then proceed to adjust the waterfall/fft appearance when you have the audio Rx level adjusted correctly.

The three signal views should be similar to this with the level/range controls set to 0 and 60 respectively. The waterfall palette used is "default.pal", and the received signal is about 20 dB above the noise floor.



Windows Audio Properties

A combination of the Windows operating system and certain audio codecs needs to be accommodated. The codecs used by devices like the SignaLink-USB, the TS590S, and recent Icom transceivers are identified by the Windows operating system as "microphone" devices. The Windows mixer controls for that device are then set to insert about 30 dB of gain in the signal path. To compensate most users simply lower the transceiver audio output. The result is very poor decoding. The correct procedure is to force the Windows mixer controls to remove that 30 dB of gain. This procedure is for Windows 10, but is similar on Windows 7, and Windows 8.

Right click on the task bar speaker icon and select the "Recording devices" menu item. This will open the Sound devices dialog:



Sound Devices

This is the dialog on my Windows 10 test computer with a SignaLink-USB connected to a USB hub. The SignaLink-USB shows up as a **Microphone** USB Audio CODEC. Right click on the device entry and select Properties. This opens up the Microphone Properties dialog:

Microphone Properties	×			
General Listen Levels Advanced				
Microphone Change <u>I</u> con				
Controller Information				
USB Audio CODEC Properties				
(Generic USB Audio)				
Jack Information No Jack Information Available				
Device usage: Use this device (enable) ~				
OK Cancel Apply				

Microphone Properties

Select the "Levels" tab and set the level to as close to 0 dB as is available:



Microphone Level

Then select the "Advanced" tab and change the type of converter.

Hicrophone Properties				
General Listen Levels Advanced				
Default Format				
Select the sample rate and bit depth to be used when running in shared mode.				
1 channel, 16 bit, 44100 Hz (CD Quality) $\qquad \qquad \lor$				
Exclusive Mode				
Allow applications to take exclusive control of this device				
Give exclusive mode applications priority				
Microphone Advanced Tab				

Apply and save the changes. The last step is usually needed to insure that the settings are not lost when the operating system is closed and restarted.

Transmit audio

Too often you see an overdriven signals on the digital sub-bands; multiple audio sidebands on PSK, splatter from overdriven MFSK and RTTY. There is absolutely no reason for a transceiver driven by fldigi to exhibit this type of performance. You can set up your computer / transceiver for good solid performance without excessive drive.

The "TUNE" button generates a continuous single frequency audio signal at the exact frequency to which the waterfall cursor has been set. The peak amplitude of this signal is the peak amplitude of every modem signal generated by fldigi. None will exceed this value, even the simultaneous multi-tone modes like Throb. Every modern SSB transmitter uses some automatic level control ALC for preventing overdrive for SSB voice. A little overdrive on a voice channel can be tolerated to a degree. In fact, that is what an analog RF compressor does, overdrive and then subsequent filtering. But you absolutely cannot tolerate that with the digital modes. Here is the way to set up your transceiver for a clean signal. I recommend starting out with a dummy load, but an "off hour" for a band might work just as well if you do not have a dummy load.

- For Windows users
 - Set your sound card output level to the minimum on the Windows mixer
- For Linux users
 - Set your PCM level to about 80%
 - Set your Transmit Level control for minimum output level.
- Set the fldigi transmit attenuator to -3 dB (control to the left of the AFC button)
- Set the waterfall cursor to 1000 Hz
- Enable the "Tune" mode in fldigi ... you do have CAT or PTT set up ...right?
- Make sure your transceiver's speech compression control is OFF

- Slowly bring up the Mixer audio out until your rig's ALC just starts to function (a light blinking or a meter showing this condition).
- Reduce the Mixer audio output until the ALC is disabled.
- You are now transmitting at maximum output power without distortion.
- You can use the fldigi transmit attenuator to make small adjustments in the output power to compensate for variations in the SSB transmit filter.

You can use any level below this and be assured that your output signal will be clean. All digital signals that fldigi generates will be limited to this peak-to-peak voltage. You should always use the minimum power necessary to maintain good comms, remember that even if you are clean at 100 W you signal will be so strong among the QRP signals that it will overpower the AGC on many receivers that are working another digital station within the same SSB bandwidth that you are on. You will appreciate this the first time that you are working a weak PSK DX station and someone blasts through and captures your AGC.

You should try the above adjustments at different audio frequencies. Transceivers that achieve the SSB filtering with crystal or mechanical filters will have a considerable amount of variation across the passband of the filter. This will show up as a varying amount of ALC that is dependent on the audio frequency. Once you are comfortable with the process you can very quickly repeat the "Tune" and set the power for the frequency to which the waterfall is set. The fldigi transmit attenuator control should have sufficient range to compensate for transmit SSB filter variation. You do not want to transmit (or receive) near the cutoff frequencies of the SSB filter. The transceiver filter will introduce both phase and amplitude distortion over the signal bandwidth.