Software Defined Audio

It's my intent to try to provide a high-level overview on digital based audio processing as it relates to your received ham radio audio. Some of it will be applicable to processing your mic audio if you so desire as well.

Even if you are not interested in the whole software side of all this, if you already have rack equipment or have been considering the use of an equalizer in your shack or other rack audio equipment, you might find some nice nuggets within this article on using an EQ and or other components.

For ham radio we can use digital audio processing to help us process mic input, process receive audio output and analyze ham radio audio in general. In my last article I covered the latter. Receive audio will be the primary focus here, however, Mic Audio is referenced a few times and covered at a high-level.

Part 1 will focus on a Post Processing Interface DAW Setup

Part 2 will focus on Equalizers, Curves and 101 Audio Tuning

Part 3 will explore Denoisers

Part 4 will explore other FX

Part 5 will touch transmit audio processing

Part 6 will wrap things up!

While this might seem crazy, processing live audio is one of the more intensive tasks we can do on a computer using a DAW and Special Effects or (FX). Most times when I talk about FX now, I will be referring to FX added in via a VST plugin.

What at exactly is a VST plugin? Well, we are going to tell you what a DAW is here in a minute. You might think of a DAW as an iPhone emulator for audio processing, and a VST plugin as an app running on the iPhone dedicated to specific audio process tasks! A second way to think about it is like its a hardware rack where you can plug in things like an Equalizer, tape deck, ect. Keep this analogy in mind as we move forward!

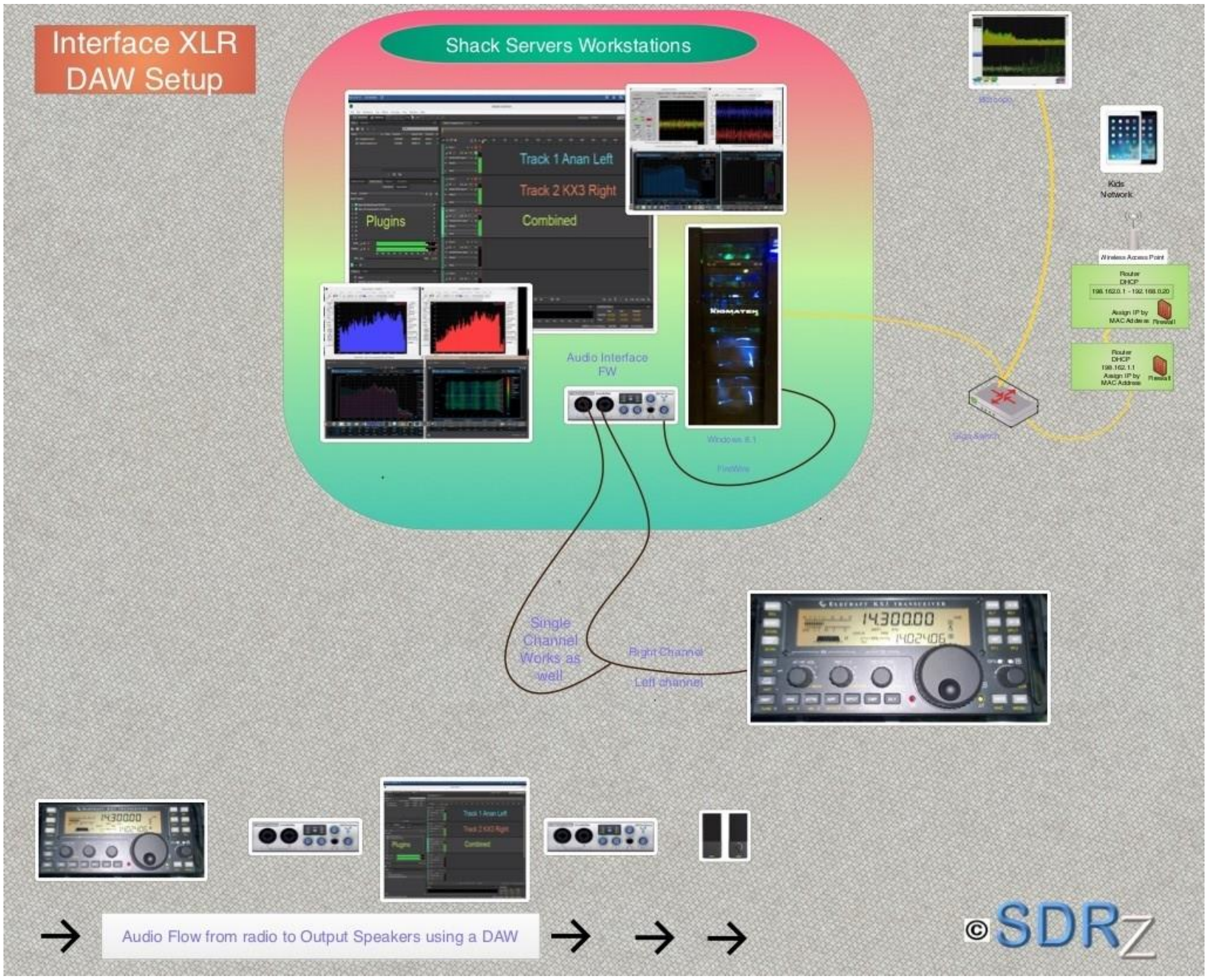
To really kick things off though, we need to talk about the computers you intend to use in this adventure. First off I am going to tell you your boat anchor is not likely going to hold up too well for this type work. I really recommend a modern computer with some horsepower under the hood. Quad Core, 3Ghz, 8-16GB of RAM might be a good place to look. Don't buy one with a consumer line CPU either! While those may seemingly boast 3Ghz speed and the other specs I referenced, they are often really woefully underpowered and hence their seemingly cheap cost.

Look for the article on SDR PC's here on SDRZone if you want to get a good PC for this and running your other software all on one machine. There is good information there you can go use to shop for the components you want in a PC that you can then create a shopping list and go find a consumer PC that fits the criteria. To give you an idea right now on what to expect, it's likely going to cost $1000-$2000 US to buy or build one truly meeting the specs you need to be successful.

Another approach perhaps is to use a second PC dedicated to audio processing. A Mac Mini or something like that might work if all it has to do is process audio and you get a dedicated external interface for it to use to input your audio.

To build our knowledge of Interfaces and DAWs, it helps to understand how audio flows through them.

A DAW is a software package most often used on a PC/MAC/Linux to produce, edit and master audio and or music. There are a variety of DAW software packages out there varying in their capabilities, price and usage. DAW is an acronym for Digital Audio Workstation.



Simple Interface Device DAW Setup

For Ham Radio we only need a simple DAW package. I say this due to the typical audio we deal with ranging from 0-20khz with the lower ends dealing with digital and voice signals and the upper ends possibly dealing with Shortwave Radio reception of music. You can learn a little more about ham audio in the Ham Audio Analysis article here at SDrzone.com if you are interested.

We can adequately sample and process this type of audio at CD quality or 41Khz sampling. Bump it to 48K if you want, however, you're not likely going to gain much!

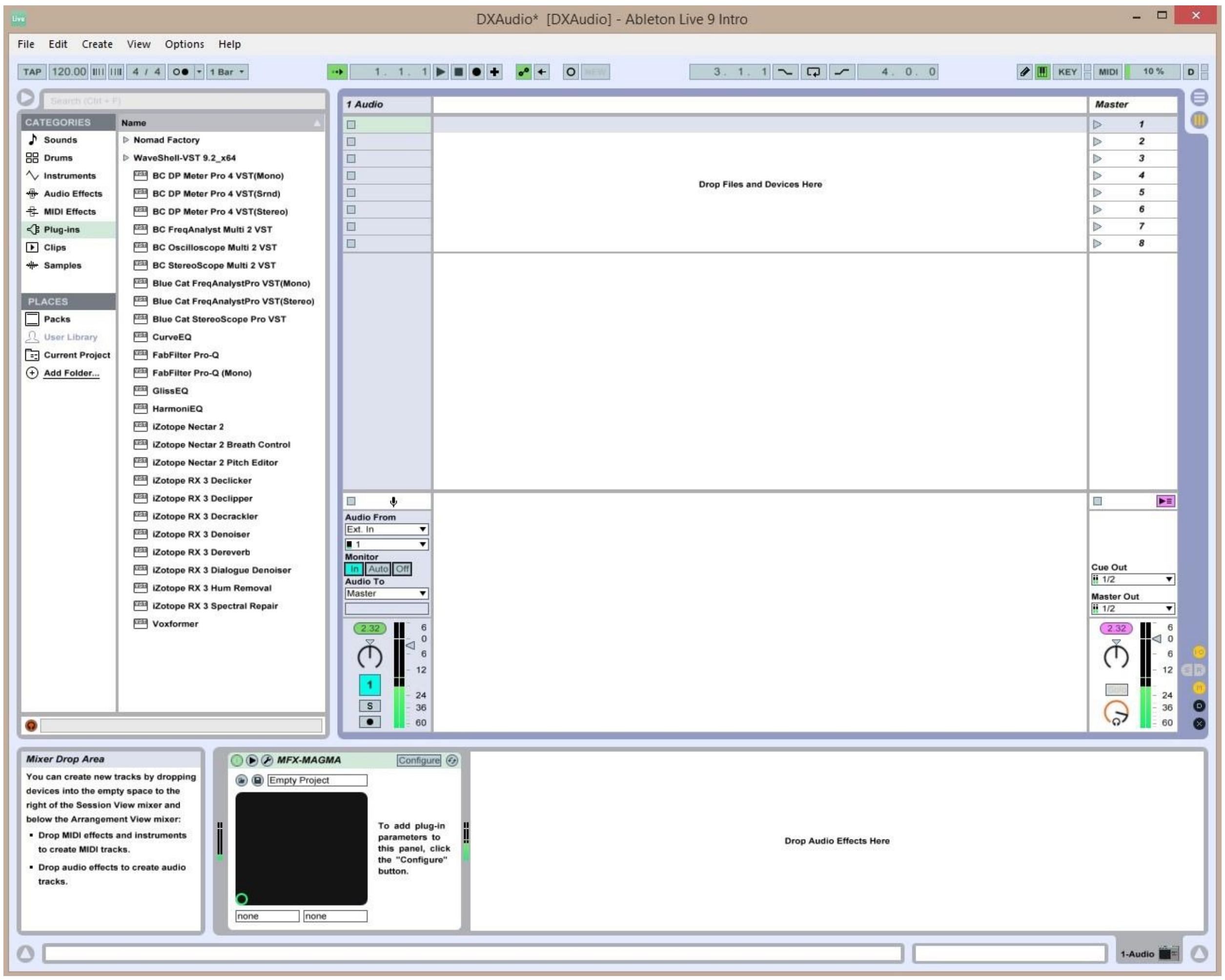
The typical middle aged person can't even hear the full 20khz anyways as we rapidly move to about a 14khz range as we age.

These rates are not to be confused with the filter width we use on SDRs and other ham radios for CW, AM, and SSB as well as other modes.

The human voice typically has a 8khz range from zero to 8khz on an audio spectrum analyzer. Music and some of the higher instruments can get into the upper ranges of an audio spectrum analyzer. To elaborate, we take your 8K of potential human voice range and represent it in a 3K range using SSB. Even though we know a human voice has 8K of range we will then potentially only see 3K of tonal range on the received end unless it was compressed and then expanded again. You can learn more about this in the Ham Audio Analysis article.

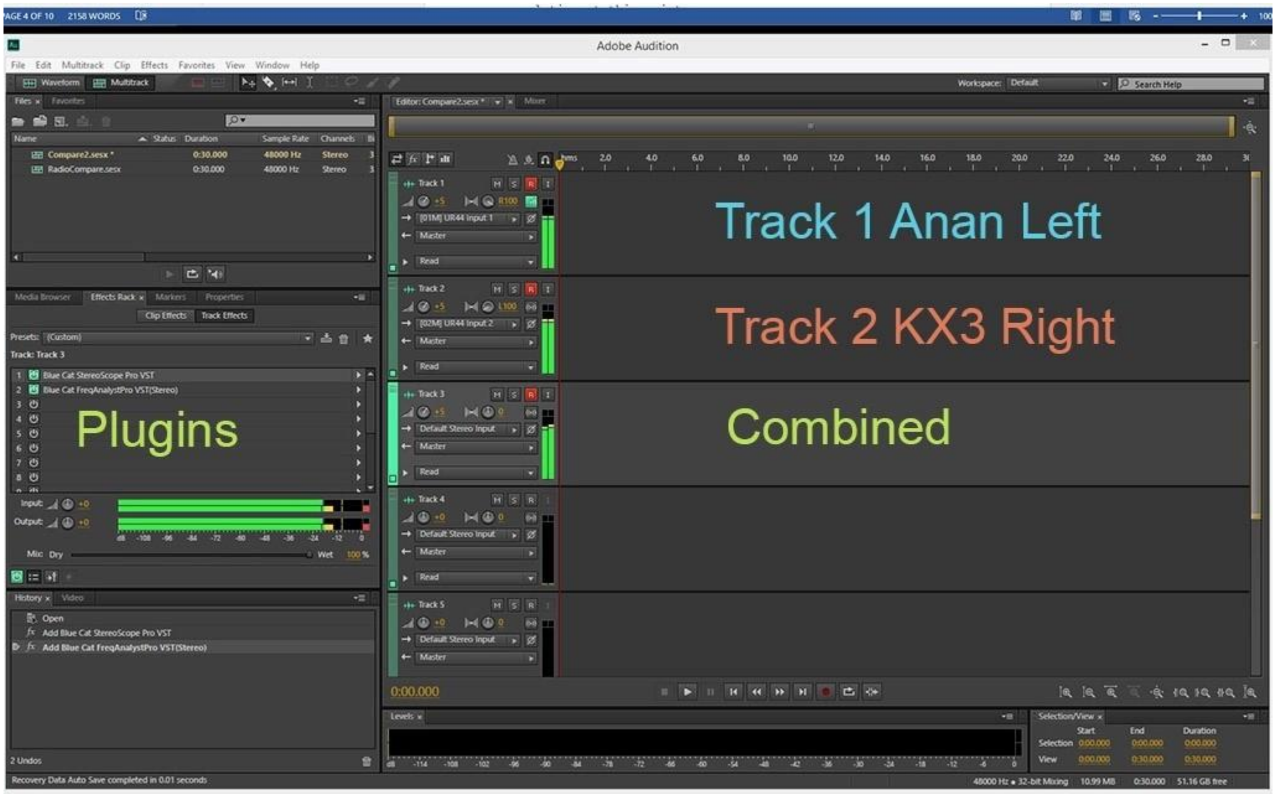
DAWs referred to in this Article are Abelton Live 9, Adobe Audition and Reaper.   Abelton Live 9 Intro runs about $100, Adobe Audition is available via subscription now for $20 a month and Reaper can be had for about $70.

Of the three I will just go out on record and say from my experience that Abelton Live 9 seemed to be the easiest to work with and overcome obstacles. Its biggest drawback right now is its inability to use the new VST 3 plugins.



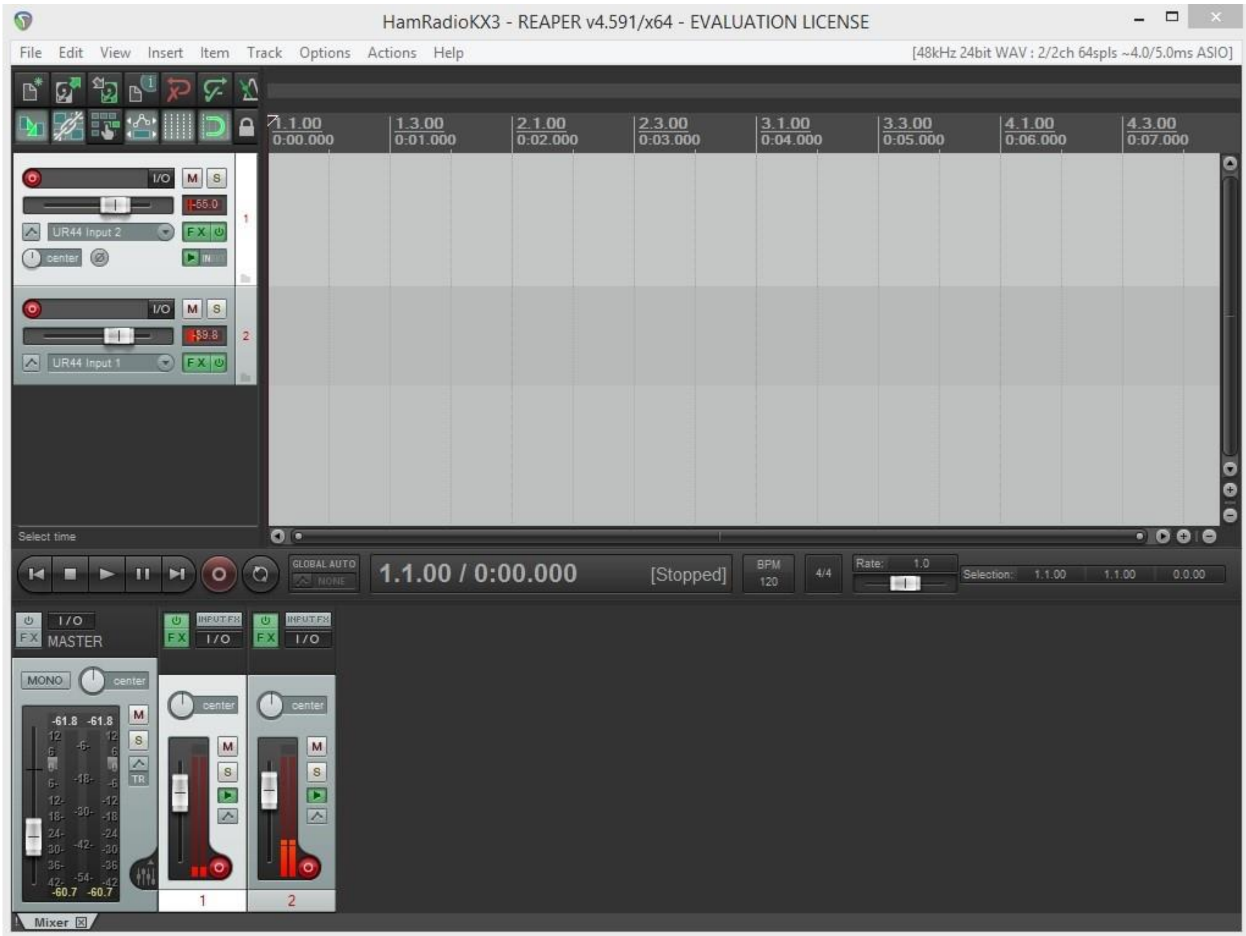
Abelton Live 9 Intro

Adobe Audition would be nice, however, I was not able to overcome a hardware issue in configuring it that Abelton allowed me to address through Driver Compensation Correction settings. It does support VST 3 and has a number of built in plugins in the DAW itself. If you find the right hardware to use with this, extremely fast then you might be ok. That said though, $20 a month for Ham use is simply too much in my humble opinion!



Adobe Audition CS6

While I liked Abelton most, Reaper might be the hams answer for a tool as it is the most affordable and supports driver error compensation as well as VST 3 plugins. Its short coming for me at first was that it seemed to force all the plugins into one large window rather than letting them free float and drag them wherever one wants on their desktop. Rather annoying actually for someone like me. This really isn't true and since having written the original article, I now use Reaper for all my receive audio.



Reaper 64bit DAW

Ultimately you can try these DAWs as well as numerous others ones to see which you like most!

Look for a few things as you shop for your DAW. Hardware it can support, Driver Error Compensation, full range of VST plugin support, ease of use in the interface, how its supported and overall user feedback.

Once you choose a DAW you next need to decide what interface you want to use.

When choosing your DAW, keep in mind that you are also choosing whether you are going to be using 32 or 64bit based VST plugins in relation to your computing architecture and operating system. This is relevant in the VST plugin world as many are 32bit, newer ones offer 64 bit versions and it may have some impact on the number of plugins you can find and use. 64 bit and VST 3 Plugins seem to be the way things are moving now with the advent of more proliferation of 64 bit operating systems running out there. To be direct, I really recommend using a 64 bit operating system and matching DAW and plugins.

Hardware

An interface can be as easy as using your soundcard in your computer. You may or may not find this adequate based on the sample rates it supports, its speed performance in processing audio and its compatibility with your DAW.

There are many dedicated interfaces one can purchase as well. There are Thunderbolt, USB and FW flavors available. Depending on the interface, the quality of its drivers, the PC they will perform differently. I cannot speak to a Thunderbolt flavor of these, however, I can on the USB verses Firewire flavors where the Firewire is faster and has lower latency than USB 2. Its not really surprising, however, FireWire is going away soon and USB 3.0 Interfaces are just not here yet.

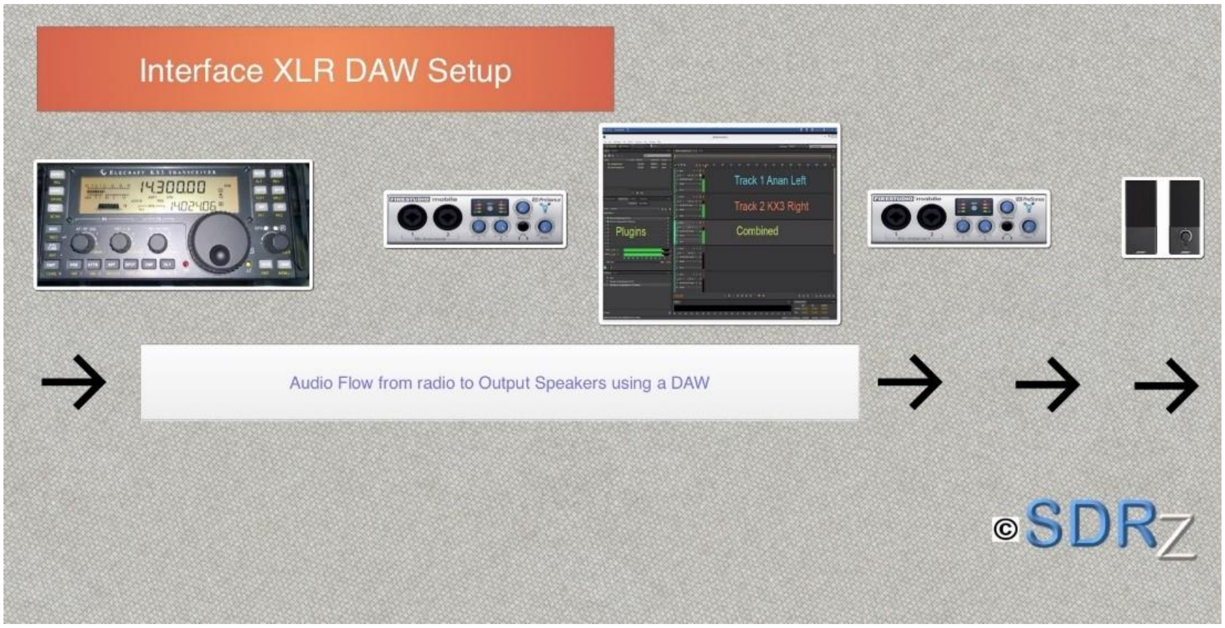
This is where Driver Error Compensation features in the DAW become important as you are likely to end up with a slower device if using an external one and will need to correct for it in the DAW to avoid hearing an echo when you process your live audio in real time.

This is probably a good time to mention that contesters may not like dealing with processed audio as even correcting for the echo will introduce some overall latency that may not be desirable in contest situations. The upshot to this is that you can simply bypass the DAW when contesting and just transmit and or listen to unprocessed audio if you use an external device like I currently use.

The radios are hooked into the Interface either via 1/8 minijacks, ¼ in jacks, or via XLR cables directly out of the phones jack on the radios. The speakers can be hooked up directly to the interface speaker jack as well. In my case I use the Phones out jack on the interface directly to speakers and the volume input levels both into the Interface and Out of the interface are controlled by the interface box and knobs on it.

NI0Z Interfaces – Steinberg and PerSonus

The computer software picks up the audio via a USB or Firewire interface, sucks it into the computer where the software interacts with it and sends it back to the Interface for output.



Simple Flow of Audio from Radio to Speakers

This trip to the computer, through the FX processing and back out to the phones jack takes time. In my initial setups I was experiencing latency which created an echo. It is said that anything more than 6ms delay will cause noticeable echo. I was hearing both the input and output into and out of the audio interface. The Driver Error Compensation settings allowed me to address this. If you are willing to feed your output back into your input on your soundcard some of the DAWs can test your system and provide the correct setting automagically.

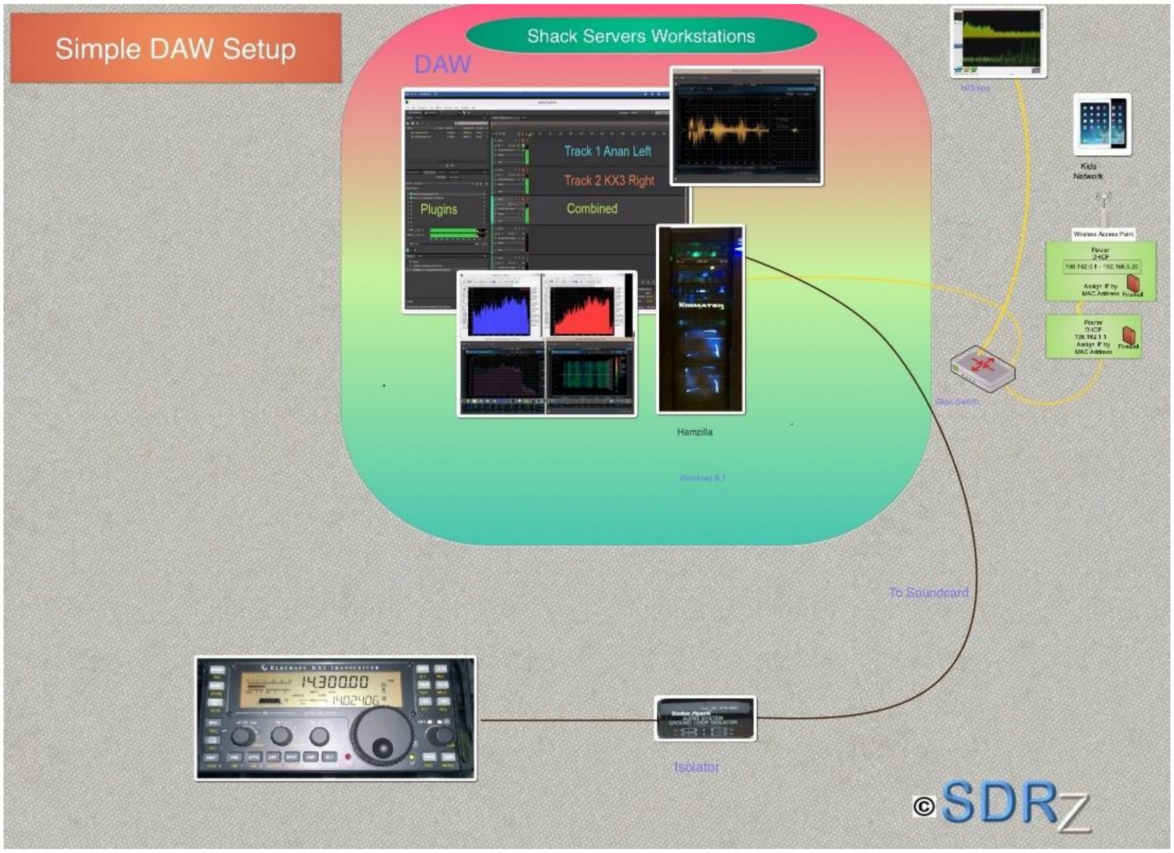
In shopping for an external interface you can look for the following characteristics if you want room to grow and think you might use the interface for something more than ham radio processing.

Greater than 110DB Dynamic Range, 192K Sampling Rates & 24 Bit Audio. If you don't mind dating your life expectancy a FireWire interface may help.

Since we can process our Mic Audio in much the same way its worth me pointing out that I found all said and done that the Mic Interface needed a faster device than the output processing. Just keep that in mind and buy your gear somewhere that you can return it if it doesn't live up to your expectations. Understand an article like this and doing more of your own research can help you make more informed decisions about purchasing an audio interface.

Some last thoughts for now on the hardware side of things. If you want to process input from multiple radios without having to switch cables all the time then an external interface with multiple inputs might be more useful. Just remember if you want more than 2-4 inputs you need to make sure whatever DAW you get supports the number you desire. They make devices that can take 16 or more XLR inputs if you really want to go there. Abelton Live Intro for example only supports 4. Just keep that in mind.

I get my gear from Guitar Center and Sweetwater if you're interested in my sources. Ask for Bart at Sweetwater and tell him I sent you if you want to order online!



Simple Soundcard Interface DAW Setup

Honestly if one wants to just play and see if any of this really appeals to them, they can simply use a sound card in the PC as shown above. If you do not have interference issues the isolator pictured may not be needed. A ground Loop Isolator though can save you a lot of hassles and they can be cheaply made on ones own or commercially purchased for about $20 at radioshack or off car radio audio sites.

So as we move forward, the trick becomes to find the right plugins that maintain that balance of desired results with desired processing power. A plugin can be thought of as a virtual piece of Rack Audio Hardware. There have been literally 10s of thousands of rack audio components for audio processing and engineering. A studio typically consisted of scores of these to address sound engineering for recording artists needs.

Again to note: when you chose your DAW you also chose whether it was going to be 32 or 64bit based in relation to your computing architecture and operating system. You can bridge 32bit plugins with great success to 64 bit DAWs using a free tool called Jbridge which you can find via google.

Today we can buy many of those hardware rack components emulated in VST plugins which you can access and use in your DAW. You can find and use literally 10’s of thousands of plugins now as well today. Also of note, you may find that your sound card if you purchased a separate one comes with its own lightweight DAW that supports VST plugins. This may be a good way to go as it should have been optimized for your card already. Many of the interfaces I have shown also come with their own DAW like CUbase and others.

Using a DAW you can reduce the need for all that physical hardware and save space at the cost of computing power. Really Software Defined Radio in large part has followed this model to some degree by taking analogue signals and converting them to digital signals and moving filtering and processing into the digital domain.

Part 2 - Equalizers, Curves and 101 Audio Tuning

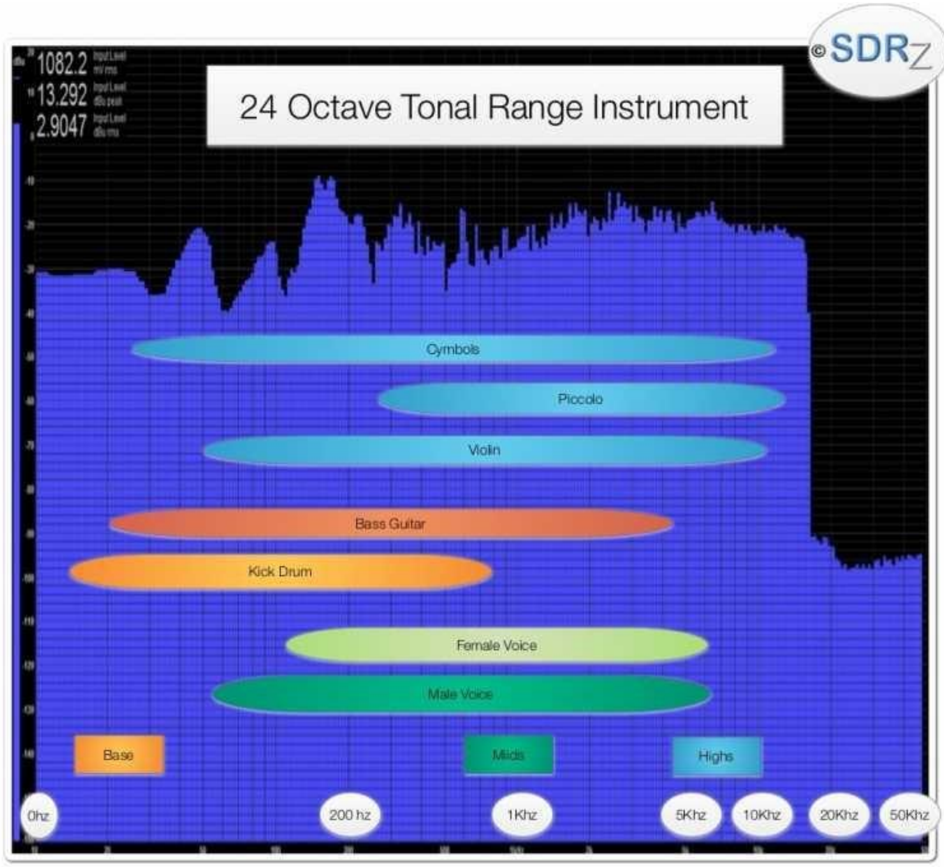
We need to understand a little bit about an EQ and what it can do for us before we consider actually getting one and applying it.

If you’re an audiophile or audio engineer then you can skip as much of this as you want, or jump to my choice of plugins.

In the Audio Spectrum Analyzer view below we see a 24 octave view of some FM Music from an 8 bit RTA Dongle.

The Bass characteristics are located very much on the lower left end of the spectrum. The Mids are represented in the lower center and the Highs start to pick-up in the right portion above 2Khz.

Interestingly in my research I came across some articles on tuning hearing aids and reading audiograms. To some extent as we age each of us starts incurring hearing loss and or impairment. Its worth noting that an EQ not only can obviously make your audio sound better, it can also actually allow you to hear it better if you keep that in mind as your apply a curve. We'll talk more about curves in a minute.



24 Octave Spectrum View and Common Instrument Mapping

In most simplistic terms an EQ will allow us to raise and lower these components either broadly or in greater detail depending on the equalizer, the number of bands and range of octaves that it addresses.

Specifically, equalizers adjust the amplitude of audio signals at particular audio frequencies using filters. Depending on the capability of the EQ this can range from broad controls to very specific control over any frequency in the audio spectrum.

Also perhaps of interest is that people hear better with visual cues. To grossly extrapolate, its entirely possible that using a panadaptor or spectral analyzer view while listening may actually improve your ability to hear a DX.

Audio Spectrum Analyzers

This is a late add to the article; I hadn't anticpated finding additional utility so quickly in Spectrum Analsys Displays.

Below we are looking at the Anan (Left) side by side with the KX3 (Right). I have been using these equalizers on top to see if I can get the two radios to sound the same.

On the bottom two quads I have Nugen Visualizer running and Blue Cat Frequency Analyzer Pro monitoring the output a few different ways. What these do is allow me to see the new shape that the EQs have applied so I can in real time adjust the curves using FabFilter Pro Q on each radios input and then monitor it on the combined output in semi stereo (meaning I am able to use stereo analysis tools to compare the two channels).

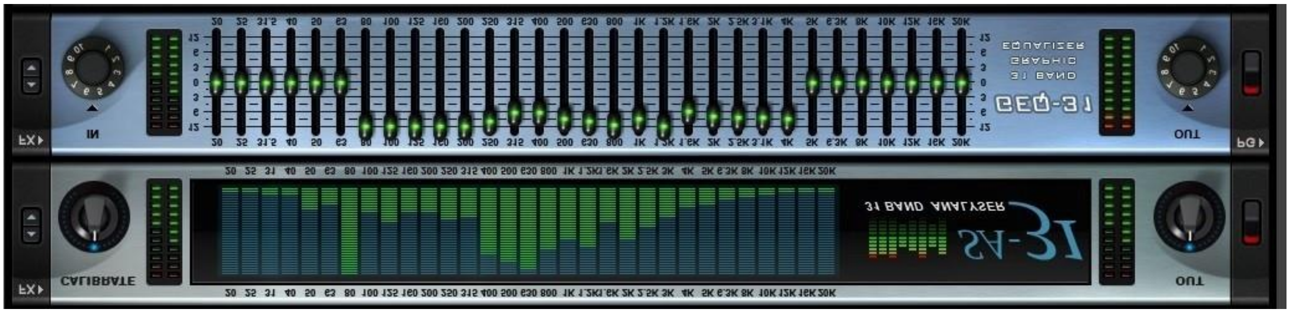
What you see next is PowerSDR on the Anan and then a Meter VST program allowing me to see if both radios are inputting at the same levels. The Anan is top and the KX3 is bottom.

As we move to the bottom display StereoScope View, we are able to see the two channels, Anan top and KX3 bottom side by side and the tonal intensity of their spectrums. As one adjust the AGC they can see what happens to the output on these screens. Remember, to the radios these are all outputs. If we adjust the AGC too aggressively (see next picture) we can see it on the StereoScope. By making sure we don’t create too much red, we can help assure we are setting the AGC correctly and fine tune our audio output for the best listening experience.

Exaggerated Example of AGC set far too aggressive on Anan Top vs KX3 Bottom

I think now that these are valuable tools in allowing the new wave SDA (Software Defined Audio) user new levels of controls for both hams and shortwave listeners. From a testing perspective as well, it helps make sure that the two radios are set equally if that is one of ones goals for testing. The other being that you can see if they are set optimally. These tools will be invaluable in setting your EQ as well. By the way, I have moved to Repear in these views for my output and it is working rather well and definitely recommendable to hams for this type work!

In one more layer lower, we can in ham terms begin to notch out problem audio as well if we have discreet detailed control over our audio. For example, if the bass components are causing your speaker cones to rattle, we can turn the bass down using our EQ until it meets our expectations. We can also address hiss to some degree by tuning the higher octaves of our output.



Magma Spectrum Analyzer and 31 Band Graphic EQ

Our goal of using an EQ is to improve the sound of the audio to our liking or to emulate various sound qualities. Ultimately the result of you adjusting an EQ results in a curve.

This article might be of interest for additional reading if you want to learn more about EQ adjustments and applying your own strategy and curves. I very much like its approach as a strategy rather than defining absolutes. http://theproaudiofiles.com/voice-processing-eq-cuts-boosts/

There are two primary types of equalizers. The following excerpt is pulled from Wikipedia and slightly modified for the purposes of this article.

Graphic Equalizer

In the *graphic equalizer*, the input signal is sent to a bank of filters. Each filter passes the portion of the signal present in its own frequency range or *band*. The amplitude passed by each filter is adjusted using a slide control to boost or cut frequency components passed by that filter.

The number of frequency channels (and therefore each one's bandwidth) affects the cost of production and may be matched to the requirements of the intended application. A professional live sound reinforcement typically has some 25 to 31 bands, for more precise control of feedback problems and equalization of room modes.

Rather than physical knobs we have graphic controls now to represent the controls and the more sophisticated the equalizer the more computing requirements it may require to process your audio.

Parametric Equalizer

Parametric equalizers are multi-band variable equalizers which allow users to control the three primary parameters: amplitude, center frequency and bandwidth. The amplitude of each band can be controlled, and the center frequency can be shifted, and bandwidth ("Q") can be widened or narrowed. Parametric equalizers are capable of making much more precise adjustments to sound than other equalizers, and are commonly used in sound recording and live sound reinforcement. Parametric equalizers are also sold as standalone outboard gear units.

A variant of the parametric equalizer is the semi-parametric equalizer, also known as a swappable filter. It allows users to control the amplitude and frequency, but uses a pre-set bandwidth of the center frequency. In some cases, semi-parametric equalizers allow the user to select between a wide and a narrow preset bandwidth.

Plugins

One great way in addition to googling to learn more about EQs is to actually play with some and this is the great thing about VST Plugins in a DAW. Many have free trials you can play with and test. There are a plethora of free ones as well and one doesn't have to spend a dime if they don't want. I will put it this way, in the older world of Rack Audio, for much of what the average ham would want to spend most of it can be had for free assuming one already has the interface hardware and DAW to use it.

EQs

In my exploration of EQs two surfaced of interest to me. Fabfilter Pro Q and Voxengo GlissEQ. Both of these combine Spectral Analyzers in the plugin, they cost $200 and $99 respectively and seem to function in similar fashion as far as being used on a radio. IE, the one for more money delivers more features and may result in better performance in more demanding situations.

For starters, the FabFilter interface is more elegant and polished, however, the Voxengo seems to have more presets. They both seem to be a hybrid combination of graphic and parametric EQs allowing you to tap into the best of both worlds. Having the built-in Spectrum Pre and Post Processed views is also very nice as you can visually see the adjustments and effects they are having on your spectrum analyzer view.

I was amazed at how much I could improve audio playing with various curves. I believe being able to make a library of your own easily accessible and applicable curves is a key feature we want to consider in our quest for an Equalizer.

Voxengo GlissEQ Picture

Where FabFilter Pro earns its higher price is in its ability to click on a headphones icon on any curve you apply and screen the sound you hear to specifically only hear that portion of the audio and the effect your curve is having.

FabFilter Pro Q

This is fabulous for finding and addressing problem areas. Find that audio that's causing your speakers to crackle and pull it down in a notchable type paradigm until it disappears.

FabFilter Pro Q – Clicking on Headphone Isolates Audio in Selected Curve

You must understand now though that every adjustment can result in both correcting an existing problem and then creating a new one! Pulling down your mids for example to get rid of an issue you are hearing may then result in flat less legible audio.

For this reason, I have found that using a second EQ on the final output of the audio to brighten and readjust audio can help. You just have to be careful it does not also create new problems or resurface your old ones. There are a variety of plugins out there, non EQ that can be used to improve audio. More on a few of those later!

One other note here, don't underestimate the ability of a simple graphic EQ to work wonders on your sound, notch out problems and add depth to your input and output audio!

Here is a great link to explore 30 top VST EQs.

http://getthatprosound.com/30-of-the-best-eq-plugins-in-the-world/

I hope you enjoyed this small intro to EQs and have fun playing with some of this yourself!

Part 3 – Denoisers

What is a denoiser, LOL? Its techno garble for Noise Reduction, something as hams that we have had on radios now for a few years. Some of these reducers have been ok at best with just a few having serious merit in dealing with band noise reduction. We don't want to confuse this with Auto Notch filters that do a really nice job nowadays in getting rid of continues hums or frequency spikes.

I am going to be a little lazy here and ask you to read some links on the topic rather than try to explain everything here. At a high-level though you need to know that these plugins were not really made to deal with our kind of Ham Noise. To clean up the noise we have would take a very industrial strength solution. I bring this up because we should trim expectations as to what we can accomplish.

Also of note is the computing requirements on some of these. There were times when the processing sucked up enough resources even on Hamzilla to cause pops and cracks in the output sound which result when our collective system cannot process the load fast enough.

Again, real-time audio processing with limited latency can be computing intensive tasks and may not work ideally on entry level computers!

Here is the detailed link followed by my cliff notes:

Gating

I use a noise gate on my Mic Audio to help eliminate fan noise from my radios, computers, ect. In most simplistic terms the gate lets you set a level where it cuts off sound in the range of audio you want, IE hiss, fan noise ect. This is great when you're not talking into the mic but tends to break down when you are talking as the noise slips through when your voice opens the gate. The reason its effective is that the gate closing again when you pause, key blank air, ect. It's an optical illusion of sorts as usually your voice becomes the focus and the distant operator on the other end doesn't notice the background noise barring it being such that it draws their attention, like a phone rigging or a leaf blower! Smile! In receive audio, to be very direct, its not going to help us!

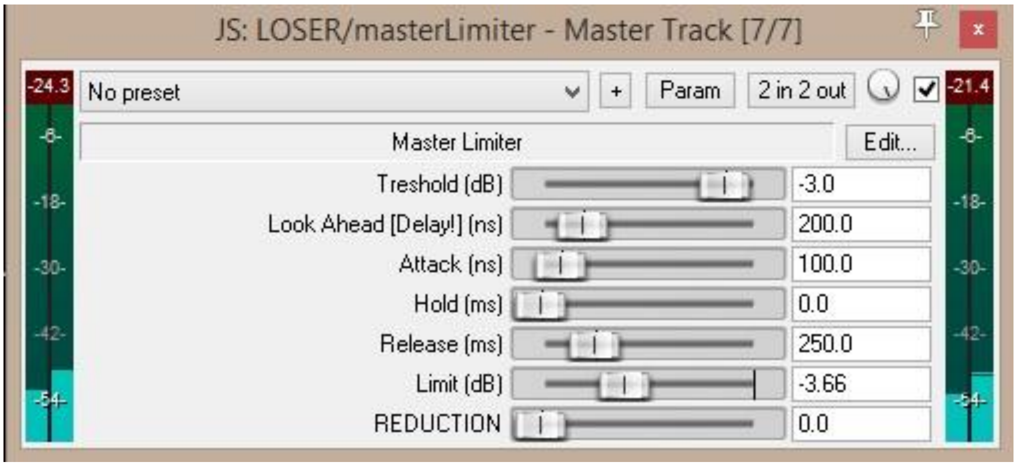
Expanders

Expanders are often used in conjunction with gates. When an audio signal falls below a set threshold, it's its gain is progressively reduced. Again, useful on the transmit side to help smartly attenuate background noise and probably not what you want on the receive side when DXing thus having it cut your receive off if the sender drops down lower. You can learn more about expanders in the article link if you are interested.

Limiters

We can deploy limiters to prevent loss of audio. In most simplistic terms we lose audio if it goes over a certain level on our output. If multiple spikes in our audio occur this can easily still create loss by compounding the overall net spike. A limiter can watch and attenuate these spikes to keep your output in a contained and lossless range. IE, it can prevent your audio from clipping and thus experiencing loss.

Here I am using a limiter to help make a DX easier. In this real scenario GP0STH from GU was weak and low, turning up my volume allowed me to copy him. The side effect though was US based stations were extremely loud to the point of discomfort. Deploying a simple limiter allowed me to knock the Pile Up stations down to a much more palatable volume level where I could comfortably work the QSO.



Notching/Filtering

This is a loose term here made up by me to describe a more predictable range of problems that some noise reduction techniques can more easily address. They are good for **Hums, Buzz, Clicks and Pops** as these are more predictable in their effect on audio and thus more easily detected and so to speak, notched out. Suffice to say that some of our VST Plugins are good at this because these problems readily occur in broadcast audio which is more like the situations we face in operating our radios on the receive side.

A simple EQ can easily act as a notch filter by allowing you to select the offending audio and reduce it. Instead of working the signal, you are simply working the audio and applying notching late in the receive chain.

Bumps, Bangs, Coughs and Door Slams are now also more readily dealt with in the higher end noise reduction engines as well. They follow similar logic toHums, Buzz, Clicks and Pops so I will let the article speak to this other than to Joke and say that if you get the right engine you can make your sick coughing ham living in a noisy house more pleasurable to talk with.

Again, I am greatly summarizing the article link here and its really interesting to read the article, I recommend it.

Fader Based Noise Reduction

These are more manual methods of using controls to manually and broadly lower noise. You might even think of it as an EQ of sorts allowing you to selectively fade areas of noise. Yes, an EQ as we discussed earlier can help reduce noise and notch problems. The plugins out there have been optimized to deal with noise and the fading controls are backed by low level algorithms that assist in dealing with noise. The rest of the Genie will stay in the bottle for this discussion and some of these companies are very secretive about their algorithms.

These types of noise reducers are overall great for broadcast Noise Reduction as they are not CPU demanding and still fairly effective.

In my exploration the description made them seem ideal for our Ham Radio Processing, more on that later!

What I observed overall with these tools is that they had a very broad impact on an audio spectrum view seemingly attenuating across the entire range and lowering the output levels.

Interestingly if we reduce volume we reduce the noise we hear with it!

Graphic Noise Reducers

These type reducers tend to be learning based in that they allow you to sample noise and reduce it from the output. They can be extremely powerful in controlled situations where you can take your time to profile and then apply and subtract noise.

They can be great for all our standard types of noise we listed in the beginning of Part 2. Their drawback are that they can be extremely computing and memory intensive thus demanding a higher performance computer.

They also can be wrongly used to actually create new unwanted distortion in the form of warbling like sounds or other undesired output results.

As you might guess, they are seemingly not well suited for real time audio processing due to their computing demands.

Plugins

In summary in my trials of plugins I found the Graphic Noise reducers to be more effective in real time noise reduction that the Fader based ones.

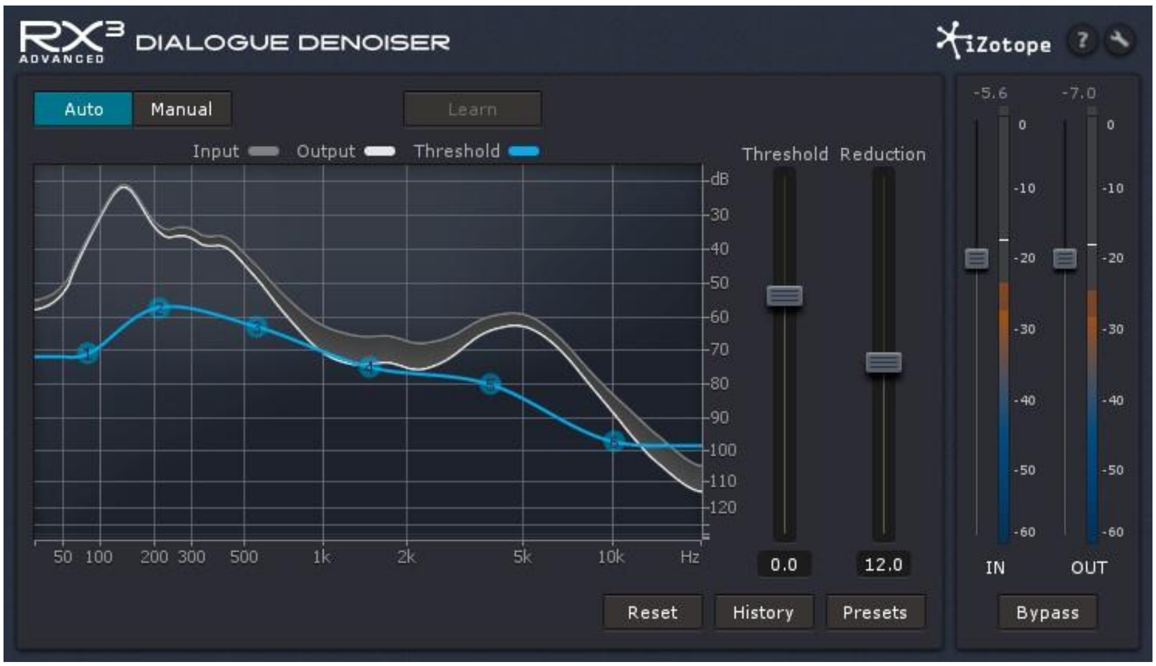
I played with the following plugins: WNS & W43 Fader based and iZotope RX3 on the Graphical based side of the equation.

W43 seems to have minimal effect on reducing ham radio noise

WNS had a moderate effect on reducing noise

The Fader based ones seemed to have minimal impact and of the two I liked the WNS better which happens to also be the cheaper one at $99. The learning suggest functions seemed to yield the best overall results and were easy to trigger on each new QSO I listened to and processed the audio.

iZotope RX3 was able at times to make seemingly significant reductions, however, often it was accompanied with random crackles and or pops sporadically popping up in the output. As you can guess, this varied by the quality level I set and the demands on the computer. Sometimes the result could be horrible and required tweaking and or multiple passes at successfully reapplying it.



iZotope RX3

I have spent a considerable amount of time talking about Noise Reduction here and in the end I am not sure how much I will end up using it in my final processed output. I may leave it in the mix and trigger it when I feel it’s worthwhile.

This begs an interesting point again about contesting and certain operating paradigms you may not feel you have time to tweak and fiddle with audio processing when making QSO’s and so ones final design for their DAW for Ham Radio Receive Audio processing may put a lot of consideration into the fiddle factors. I am a fiddler and honestly sometimes rather than fiddle than communicate. To each their own, however, just as Hams have incorporated rack based audio into their shacks there will be a happy medium here where the fiddle factor can be limited and one can vastly improve the sound and copyability of their received signals.

Thinking in terms of whether I would invest in these, if I did, it would likely be in WNS for $99 as its hassle free for the most part, low CPU, and has a moderate effectiveness in reducing ham noise.

A useful Link in exploring Noise Reduction Plugins

http://www.sonicscoop.com/2013/05/30/the-best-noise-reduction-plugins-on-the-market/

Part 4 – Other FX

In part 4 here we will talk about some other FX that one can leverage out there. I will try to stick to those that apply to Ham Radio Audio.

When I first started off down this road 2 years ago I began with Rack Audio. Ironically I found the rack components, particularly the equalizer and FX unit difficult to work with as the controls were embed in menus accessed manually in tiny screens and referenced in large manuals! That can be a fun adventure on its own, however, I soon wanted something different and dreamed about processing my audio on the computer along with my preferred style choice of Software Defined Radio. After all, why not have software defined sound to go with my software defined radio!

At the time dedicated audio hardware processors were big and one could buy a PCI Card or FireWire device you could add on to your computer and they often had plugin bundles that came with them. They had onboard FX Processors to do the heavy lifting and depending on the number or quality of the processors one could use X number of plugin FX in a live given processing situation.

I am really glad I resisted the desire to buy into that because I feel they are already largely extinct and being more and more replaced by higher end PC's running DAWs and Plugins on the PC.

My fear was the computing hardware I would need and the certainty that I would need to dedicate a PC to audio processing. I had a stealthy PC at the time Circa 2008-2010 standards and so I bought a Presonus Firewire Interface, and a cheap plugin pack from Nomad Factory with 64 FX and Abelton Live to run them.

Plugins

In having used Rack Audio and having played with the Nomad Factory Plugins I felt this was a semi affordable way to get into processing my Mic Audio which was my focus at that point and time.

It had a spectrum Analyzer, an EQ to adjust my voice as I wanted it and some extra special FX plugins I liked.

Claritone

One was called Claritone which was a voice fullness component that made my voice sound fuller and richer despite the 3K SSB range limit we operate in.

From the manual: Claritone is an all-purpose enhancer that can fatten your bass, strengthen a chorus or add clarity and sparkle to a vocal performance. Claritone makes it easy to get musical results with the quick turn of a knob.

Spring Reverb

The other was a reverb I used to add the tiniest amount of reverb (echo) affect to brighten my voice as well and add presence. On a scale of adding this effect from 0-100 I would say I added .3 percent of reverb.

Spectra Gate

Eventually I learned about the Spectra Gate in it that allowed me to greatly rid myself of background noise from my big Ameritron Amp, Anan Fan and Computer Fans. This was far superior to the DE Noise Gate in PowerSDR.

From the manual: This plug-in ideally suited to drums and other percussive material as well as guitars and vocals. It offers a super-fast response time, stable triggering and freedom from chatter. Spectra Gate helps remove noise and hiss, but can also be used as a special effect

Yellow Cab

Another Plugin I like is called Yellow Cabinet and it can add various broadcast sound qualities to vocals. It includes Mic Emulations and broadcast tone emulations to dynamically change or brighten in an overall goal towards emulating famous sounding audio.

From the manual: As its name implies, Yellow Cab will always take you where you want to go, quickly and reliably. This cab isn't just one color though: It packs a punch with 25+ cabinet emulations to choose from. Choose from an assortment of different cab configurations to get the tone of your dreams. As a courtesy, we through in different Microphones and in our favorite positions.

This subject could get quite exhaustive and as you might imagine you can use these plugins for both input and output. Here is a Link to the Nomad Factory Plugin Manual and I offer it not to advertise for them, however, if you skip down to the plugin descriptions you can learn a lot by reading them all and learn even more about audio processing.

http://www.timespace.com/pub/MAGMA_User_Manual_v2.pdf

At $99 this buy far was still my best buy to date and thinking about it I could as a ham fully embrace and live within this set of plugins without ever adding any more. In fact, just writing this brought my attention for my need to go play with some more of these! Smile!

Part 5 – Transmit Audio

This part will be very short! The main reason being that we have already touched on most of the plugins you might use for transmit audio processing, often in cases making direct references to doing exactly that!

What you do need to know is that on an integration level there could be massive issues in trying to operate your station with both transmit and receive functions covered in a single DAW.

The reason is that if your feeding mic audio into an input and also feeding receive audio into an input, without manually switching between the two somehow while you're transmitting and receiving the two may not play well together! And we certainly don't want the hassle of manually flipping back and forth!

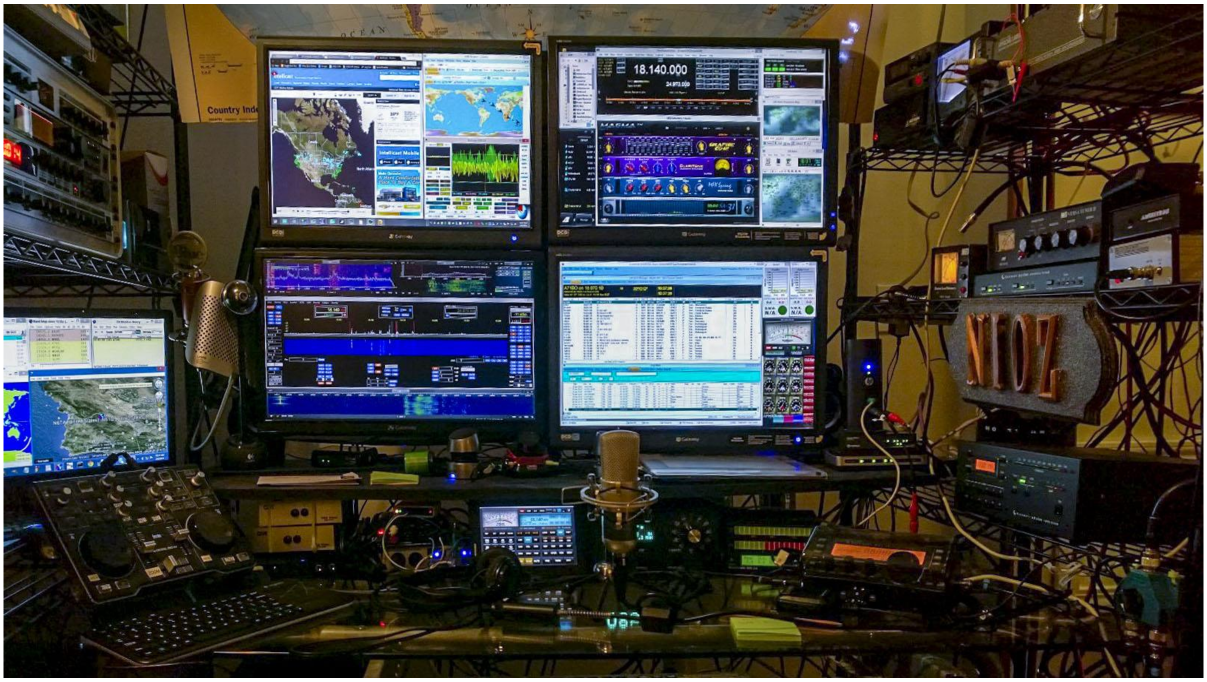
The way I solved this was to run two interfaces and two DAWs. Sounds crazy but it works great and there are two distinct circuits if you will for transmit and receive audio. This method requires zero switching! The downside is that running another DAW has computing overhead!

As you can see in this next screenshot of my desktop, its possible to run everything on a single computer with 2 DAWS, 2 Interfaces, SDRs and a plethora of software.

Reflects a full blown software defined ham station with extensive integration

On the top left we see the Microphone Instance of Abelton running with the Personus Firewire Interface and Magma Plugins and on the top left we see the Receive Audio Plugins running a BlueCat Analyzer, RX3 Noise Reduction, GlissEQ and BlueCat Meters Pro. A dual mono to stereo adapter doubles the Anan audio output which is fed into Input 1 and Input 2 on the receive audio interface and DAW where I create seudo stereo. Just to give you an idea of how much load a good PC can lift, Studio 1 is running, PowerSDR is running and not shown, PSTRotator is running, Ham Radio Deluxe is used to Sync Studio 1 to PowerSDR. You also have, Watt Meters, propagation forecasting software and CommCat logger and spotter running. World Clocks, DDUtil, Bobs Meters and TrueRTA Spectrum Analyzer and scope on the bottom screen. Hamzilla is running about 50% CPU handling all this with VAC and Virtual Serial Ports running underneath. Even with all of this running Hamzilla can pass a DPC test and report green. PS, this is an insane example and something I was playing with, I dont actually operate like this on a regular basis.

It might be hard to fathom how this all runs as far as a station instead of the screenshot. Here is a shack shot showing quad displays with a small USB monitor on *the bottom to better visualize all the integration. Neatining up cables by the way is the next project! Actually, much bigger changes to make it all more comfortable and neater are in the works!*



Depending on what microphone you are using, you may want a semiprofessional interface with phantom Power and XLR inputs. Using this you can use Condenser Mics you can purchase at music stores and XLR cables to hook them to your interface.

I use a Blue Spark mic for example and another lesser known name and they both work great!

I only use the Nomad Factory Plugins now for mic audio and they have served me well since I started using them. Its not that I am claiming this as the best or anything like that, they have just simply met my own needs to adjust my transmit audio.

iZotpe make a plugin called Nectar that I have been looking at for some time, however, I just can part with my money for it. I plan on playign with the trial this weekend to see if I change my mind.

MAGMA RACK

Latency can be a real issue on input audio, be prepared to find and use a faster interface for this type processing. The original USB based interface I purchased had to go back in favor of a firewire interface. It may be that Driver Error Compensation may allow a slower interface and if I find time I will try this with my USB interface.

Part 6 - Wrap-up!

I hope, you have found this latest article useful! As I always tell you, I write these as much for me as I do you. Why, because it makes me learn better so when I share I don't embarrass myself too much!

I have left us short here so that you can explore on your own and honestly, I want to continue to explore and learn. There are all kinds of filters to explore and learn about! Audio knowledge is rather expansive in this day and age with all me know and have learned over the years.

If you have rack audio, like it and feel you are accomplishing all your goals, there is merit in keeping it! The reason I say this is any time we choose to move from appliance like hardware to the computer we are subject to many other complexities and issues running with other software and as I have experienced myself, the operating systems themselves and the numerous patches that come out for them!

On the other hand, if you are looking for ways to turn that rack investment into a new platform, selling the hardware and taking the money to get an interface and or DAW could be a nice move if you already have the computing hardware to go with it. It will free up the space your rack is sitting in and open up numerous other possibilities.

You should know that Shortwave Listening is also greatly enhanced by all this, especially if listening to music! I was able to make some local AM radio sound as good as FM Stereo almost using these tools.

If you have a lower frequency SDR then you can also leverage this for FM communications as well.

The point is there is a new world of possibilities here and you never know how a Ham might take a plugin designed for one purpose and put it to use for another! EQ into Notch Filter, wink wink!

Have fun and you can find the forum thread with continuing thoughts and notes in the ZForums section!

https://sdrzone.com/index.php?option=com_kunena&view=topic&catid=6&id=200&Itemid=146#656

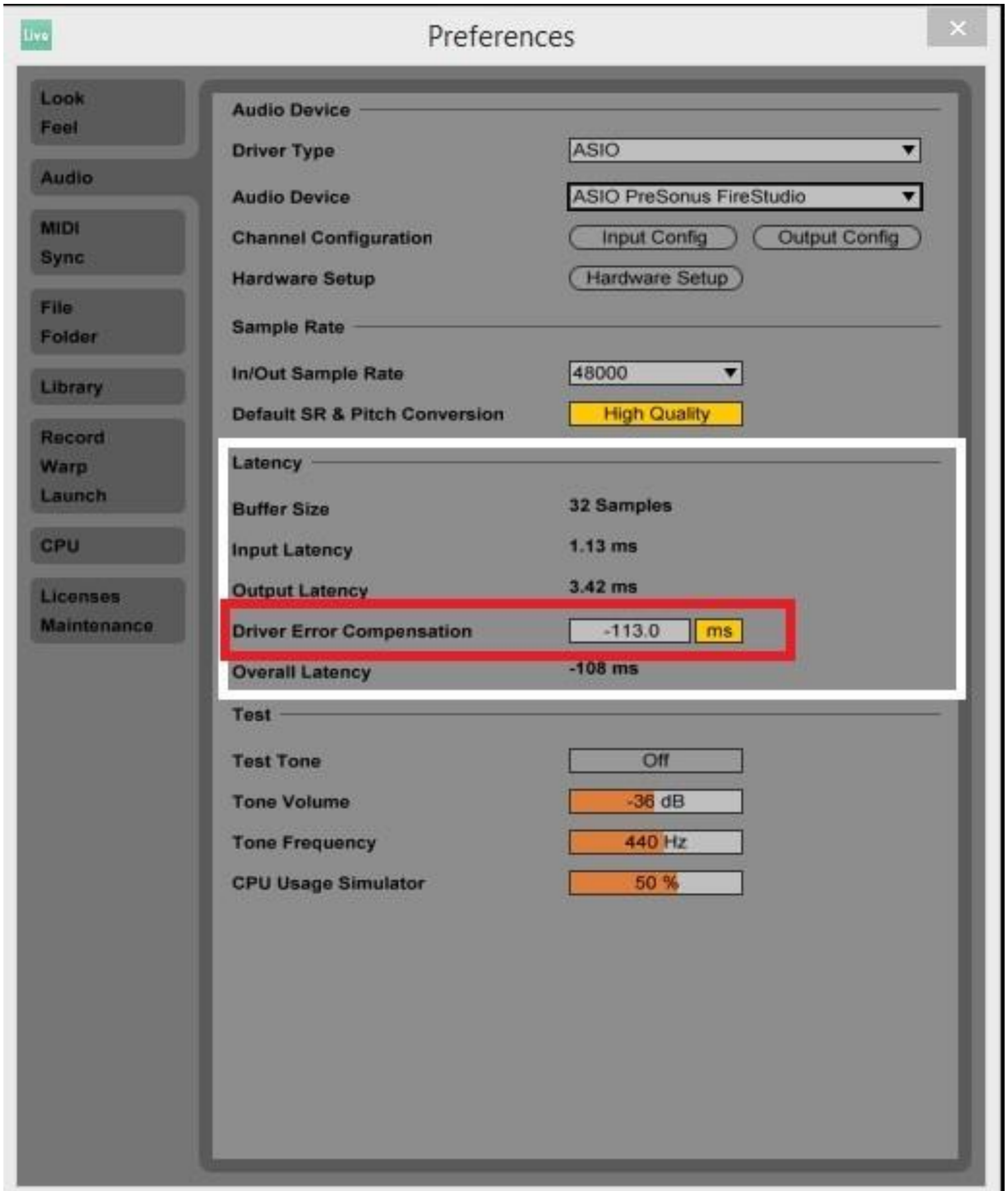
Last but not least are a few notes I had captured and left in just for fun and your reference!

Additional NOTES

It took a bit of work to get the interfaces and DAWs configured and was not as straight forward as I would have liked. I had to abandon Adobe Audition and move into running a second copy of Ableton Live Intro 9.

The reason I had to move to Abelton may be flawed or simply my inability to figure out how to deal with the latency I was experiencing.

Abelton allowed me to slow down the input audio to match the latency experienced at the output to get rid of the echo. I know, that sounds crazy because you would think that it would just slow the output down as well. Think of it as an offset where you set negative input latency. I am not going to attempt to explain this more as I simply can’t at this point, however, it worked!



Driver Compensation Error Correction Picture

The net result of all this magic is I can now use and hear the result correctly of the processed receive audio coming from my Anan or KX3 and adjust it in real time on the PC. This has some limits, even on Hamzilla because some of this software was really meant for processing recorded audio and being able to take its time and use lots of CPU to do its work. Realtime audio breaks down, if you can't process fast enough you can get crackles, pops, ect.

For my own station I have landed with the Magma Plugins for Transmit and see no reason yet to change them. The upshot of controlling your audio is that you can enrich it greatly and still stay within the 200-3200hxz range virtually creating audio that sounds

8K wide to the listener. Many can't believe how good audio can sound if some care is put into it.

On the receive end which is a newer domain for me, I am very much interested in FabFilter Pro Q, WNS for Noise Reduction and am looking at some other plugins.

I happen to like the BlueCat plugins I purchased and am looking forward to playing with those more on both ends of my audio, in and out. Since I own Abelton I simply launch two copies of it to manage my in and out audio.

Voxengo and Nugen both have plugins I am wanting to tryout. Flux does as well. There are just so many to explore!