THE 2013 WIKD CHIEF ENGINEER MANUAL

BY AL REYNOLDS

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AUDIO
PART 1
FUNDAMENTALS
THE BASICS

· SOUND WAVES

· ELECTRICITY & SOUND WAVES

· SIGNAL TYPES

SOUND WAVES

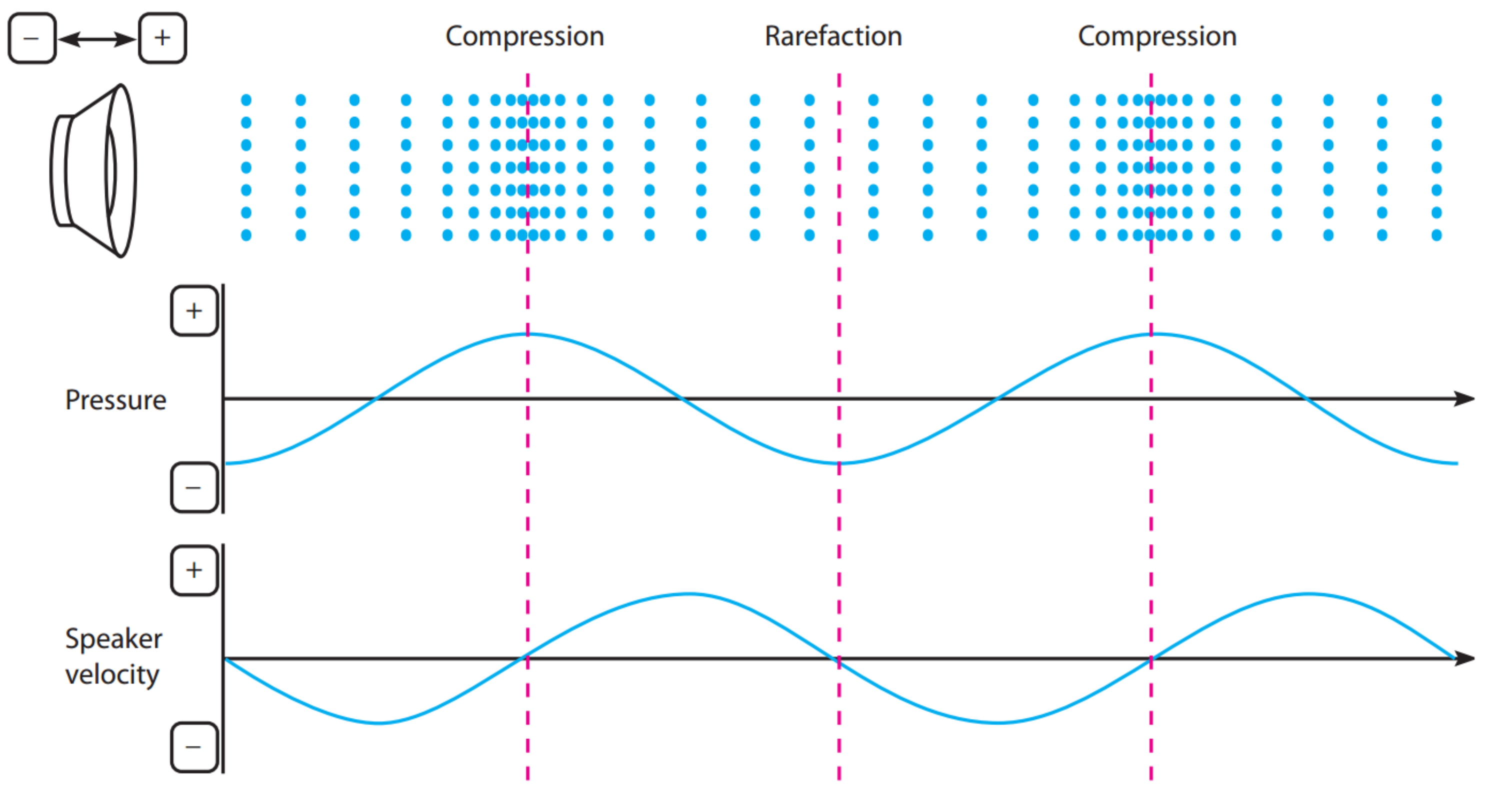
Sound, whether its noise, music, or speech, is the vibration of particles in a medium. To understand this, imagine a speaker.

In order to produce sound a speaker pushes a piece of paper, called a diaphragm, in and out repeatedly. As the diaphragm moves forward it pushes the molecules in front of it closer together. As the diaphragm moves backwards, the molecules in front are “pulled back”.

In order for this motion to constitute a recognizable sound it has to happen a lot. How often it happens in a single second is called the frequency. The higher the frequency, the higher the “pitch” of the sound, and vice versa. How loud it is, is determined by the pressure with which the molecules are moved. This is called the amplitude.

Because this happens in such a cyclical nature, the easiest way to visualize it is to use a waveform.

FIGURE 1 - AUDIO WAVE PROPAGATION



When you look at a waveform, the amplitude is represented on the Y axis, while time is represented on the X axis. If the amplitude of the wave is positive, it generally means that the speaker is pushing outwards, while a negative amplitude means it’s receding.

You can determine the frequency by measuring how many cycles pass in a single second. If you imagine a speaker diaphragm at rest, that then moves out and then in before returning to a rest state—that is a complete cycle. On a waveform, a cycle is considered as being from peak to peak, or crest to crest.

A normal human should be able to hear from 20 Hz to 20 KHz. 20 Hz means that a speaker’s diaphragm (starting from rest) moves in and out twenty times every second. 20 KHz means that a speaker’s diaphragm (also starting from rest), moves in and out twenty thousand times every second.

ELECTRICITY & SOUND WAVES

Sound waves can be recorded and then stored in either the analog or digital domains. In the analog domain, sound gets represented using electrical signals. In the digital domain, sound gets represented using ones and zeros.

Conveniently enough, electrical signals work (almost) exactly like sound waves with a few magical properties thrown in. Technically, electricity is the flow of charged particles from areas of high charge to areas of low charge.

Electricity is divided into two major groups: Alternating Current, and Direct Current. In DC circuits electricity flows in one direction only. In AC circuits, the flow of electricity changes direction periodically. For example, the electrical circuits in your home deliver AC power at 60 Hz—that is, the current changes direction 60 times per second.

Audio equipment uses AC circuits to transmit sound waves as electrical signals. The frequency of the sound wave corresponds to the frequency of the AC current, while the amplitude of the sound wave corresponds to the voltage level of the electrical signal. Simple enough right?

MICROPHONES

It's the job of a microphone to convert changes in sound pressure to an electrical signal. For the most part, microphones exist in two broad categories: dynamic microphones, and condenser microphones.

Dynamic microphones are the most common type of microphones. A dynamic microphone consists of two major parts: a diaphragm and a voice coil. The diaphragm is a thin piece of material that vibrates as sound waves are applied to it. As it moves, it causes the voice coil to move through a magnetic field, thereby generating an electric current. This electric current is the audio signal.

Condenser microphones are slightly more complicated to explain, but in general they tend to be more sensitive and responsive to dynamic changes. In addition, they must be powered in order to work—this is why microphone preamps typically have a "+48V" or "Phantom Power" option.

SIGNAL TYPES

MIC LEVEL

The quietest signals of all, these typically refer to the output directly from a microphone (imagine that).

INSTRUMENT LEVEL

These are signals that come directly out of an instrument such as a guitar or electric keyboard. Depending on the type of instrument, the strength of the signal can vary vastly—from very low mic levels, to very strong line level signals.

CONSUMER LINE LEVEL

The majority of consumer audio equipment operates at consumer line level (technically, -10dBV).

PROFESSIONAL LINE LEVEL

The level at which most professional audio equipment operates at (technically +4dBu). It is significantly stronger than consumer type line level signals, which results in a much louder output. Wherever possible, you should seek equipment that operates at this level.

AMPLIFIER OUTPUT

The most powerful type of signal. Typically the last line in the audio chain, used to connect speakers to amplifiers.

In addition to dynamic and condensor microphones, there are also ribbon microphones. However, ribbon microphones are typically only used in recording studios, as they are both exceptionally expensive and exceptionally dainty.

If you run into a ribbon mcirophone, make sure that you do not supply it with Phantom Power, as this will usually damage it.

AUDIO FUNDAMENTALS
PART 2
TRANSMISSION

- BALANCED VS. UNBALANCED LINES
- SIGNAL GROUNDING
- CONNECTING BALANCED & UNBALANCED EQUIPMENT
- STUDIOHUB+ WIRING
- SPEAKER WIRING

BALANCED VS. UNBALANCED

Regardless of the level at which the equipment operates, it will invariably use either balanced or unbalanced cable to connect to other equipment. In the professional audio world, almost all equipment uses balanced line connections. In short, balanced cables are significantly better at rejecting both electromagnetic interference (EMI) and Radio Frequency Interference (RFI).

Remember how a standard dynamic microphone works? A dynamic microphone consists of two major parts: the diaphragm and the voice coil. The diaphragm is a thin piece of material that vibrates as sound waves are applied to it. As it moves, it causes the voice coil to move through a magnetic field, thereby generating an electric current. This electric current is the audio signal. Tada!

In order to create an electrical circuit between the microphone and the preamp, a loop must be created. This means that at a minimum there must be two lines—typically referred to as a signal line and a ground/shield line. This is called an unbalanced cable.

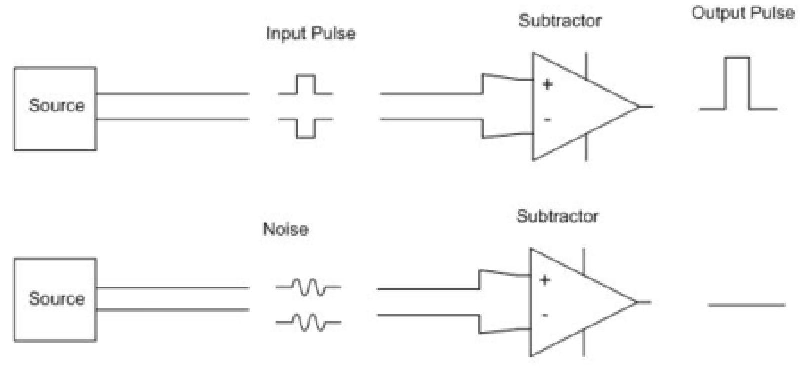
Unbalanced cables are typically coaxial cables (think RCA or TS cables)—the ground/shield line is the outside conductor, and the signal line is surrounded by the shield line. The ground/shield line serves multiple purposes: it completes the electrical circuit (it can be thought of as the return path if the signal line is the send path), and protects the signal line from most electrical interference.

However, in environments with a lot of electromagnetic interference, or in longer cables, the shielding is no longer enough to protect the signal cable from interference—especially if the unbalanced cable is being used for a particularly low level signal (like a microphone signal).

This is where balanced signals come in. A balanced signal requires two signal cables that have identical impedances. Impedance can be thought of as the opposition to the passage of current. Basically, if two cables have identical impedances, it is equally easy to generate a current within the cables. This way, if the cable is subjected to any external EMI, the noise that they would induce would be applied equally to BOTH cables, and with the same polarity.

NOTE: Microphone signals are particularly susceptible to EMI because they need so much amplification—a typical mic level signal can be as low as -60dBu, and a preamp will typically amplify that to around 0dBu. That equates to an output that's roughly 1000 times hotter than the input signal—and just as the original sound has been amplified 100 times, so too has the noise that accompanied that signal.

DIFFERENTIAL SIGNALLING



(OOH A PICTURE!)

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Basically, if two cables have identical impedances, it is equally easy to generate a current within the cables. This way, if the cable is subjected to any external EMI, the noise that they would induce would be applied equally to BOTH cables, and with the same polarity.

In addition, both signal cables carry the signal, but with opposite polarity. That is to say if the "hot" signal cable indicated a voltage of +1.5V, the "cold" signal cable would indicate a voltage of -1.5V.

The trick comes at the end of the cable. The receiving devices looks at the signal on both cables, and then subtracts them from each other. This has two effects—firstly, because any EMI would have induced noise on both cables with the same polarity on each cable, the noise would effectively be nulled out.

Secondly, because the original signals had opposite polarities, taking the difference of the two signals results in a doubling of the original signal. This is called differential signaling.

Differential signaling generally works very well at preventing interference. However, if an EMI source is particularly close to the cable, it can still cause more noise to be induced in one cable than the other. This is common when many cables are laid close together, or in close proximity to power lines. The solution to this is to twist the cables together, thereby ensuring that both cables are as equally susceptible to the EMI source as possible.

Lastly, while balanced cables do not necessitate the use of a shield wire, many cables still include it. However, whereas in an unbalanced cable the ground/shield wire serves as the return cable, in a balanced cable the ground/shield cable is strictly for shielding. Almost all microphone cable includes a separate ground/shield wire.

NOTE: It is not strictly necessary to use shielded cable for audio transport. Ethernet wiring is often unshielded (It's known as UTP—Unshielded Twisted Pair) and is perfectly acceptable for audio signals. However, for situations where you're likely to have lots of cables next to each other, or very close to power lines, it may be necessary to use shielded cable.

SIGNAL GROUNDING

Every piece of audio equipment should have three "grounds": a signal ground, a chassis ground, and an earth ground. The purpose of a ground can be twofold—it is first and foremost a reference point from which voltages can be measured. It can also be used as the return path in an electrical circuit—in unbalanced cables, for example, the ground/shield wire serves as the return path to complete the electrical circuit.

The primary reason for equipment being grounded is for safety. This is where the earth ground comes into play. Simply put the earth ground is a conductor back to the earth. In the event of a power surge (i.e. a lightning strike), the earth ground would represent the easiest/quickest path for the excess power to be dissipated (electricity will move from areas of high charge to areas of low charge, and the earth, because of its size, typically represents the area with the least charge).

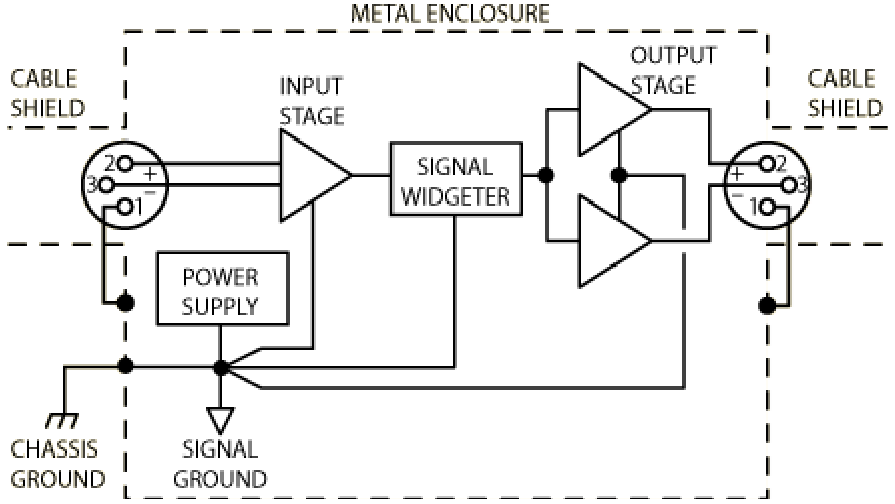
If equipment wasn't grounded and it was subjected to a massive electrical surge, the equipment itself might become electrified—which would be a significant safety hazard.

The chassis ground can be thought of as the enclosure—basically it's the collection of all the conducting materials. If the equipment uses a properly wired 3 conductor power cord, then the chassis ground and the earth ground will be linked. This goes back to the above section—by connecting the chassis ground to the earth ground, you ensure that the enclosure has the same potential as the earth ground, which helps prevent the equipment from becoming electrified.

Last comes the signal ground. Its only purpose is to serve as the 0V reference point for the actual audio circuitry within the unit. As such it's important that it remains as immune as possible to EMI and RFI.

PROPERLY DESIGNED AUDIO EQUIPMENT

(SHOULD LOOK LIKE THIS)



(ISH)

BALANCED & UNBALANCED INTERCONNECTION

First off, if you're working for WIKD you shouldn't really have to worry about this—nearly all of our equipment uses balanced connection points, and all of the wiring is UTP CAT6 Ethernet. In fact the only two unbalanced audio sources are the DJ inputs, and as you'll notice....they have a very quiet buzz (they're not wired properly right now).

However, at one point or another you will probably have to connect the two together and you should know how. The best, and easiest way is to use a DI (Direct Inject) Box. Originally designed to convert the unbalanced instrument level signals from instruments to a balanced line or mic level signal that could be run long distances and connected to a mixer, you can use them for converting most any unbalanced signal to a balanced signal.

Radial and Whirlwind all make excellent devices specifically for this purpose.

If you can't afford to spring for a DI Box, then you can use a special cable assembly. In addition, because almost the entire station use the StudioHub+ wiring standard, you can use this handy guide for making interconnections.

SPEAKER WIRING

Speaker wiring is, by and large, a whole lot easier than most other things. It begins with determining whether you're using an Active or Passive speaker. Active speakers are those that contain an amplifier within the speaker enclosure—the studio monitors and subwoofer in the studio are examples of active speakers. Active speakers are particularly convenient because you only need to connect a power cable and an audio cable—the subwoofer, for example, accepts balanced XLR inputs.

Passive speakers are more complicated, because you have to use specialized speaker cable to connect the speakers to the amplifier. Because the output from an amplifier is so much stronger (relative to the input), you must use thicker cable that is capable of supporting such levels.

All speakers use differential signaling—that is, they require both a positive and negative polarity signal in order to physically move the driver (a loudspeaker is actually a reversed dynamic microphone—electricity is run through a voice coil, which generates a magnetic field. Variations in the signal cause variations in the magnetic field which forces the diaphragm of the speaker to move in and out. And just like that you get sound!)

In connecting amplifiers and speakers you will likely only run into three different types of connectors: Banana Plugs, TS connectors, and SpeakOns.

NOTE: It's now illegal in Europe to produce equipment which uses Banana plugs, so the likelihood of you finding new equipment which uses them is slim. TS connectors are limited in the amount of power they can handle, generally have no locking mechanism, and can short a live amplifier if they're unplugged. For these reasons you will generally only find them on low end speakers.

Speakons may be the greatest connector ever made. They're easy to use, durable, and don't suffer any of the problems of TS connectors. (Nor are they illegal!) They are made by Neutrik, and just about any professional loudspeaker will use them. They come in 3 different configurations: 2 Pole, 4 Pole, and 8 Pole. A 2 Pole connector has 2 conductors—one for the positive polarity signal, and one for the negative polarity signal. Likewise a 4 Pole connector has 4 conductors, and can therefore carry 2 separate audio signals. An 8 Pole connector can carry 4 separate audio signals.

The reason for supporting multiple audio signals over a single cable is simple—most loudspeakers contain more than one speaker, or driver. Sending multiple signals over a single cable allows you to amplify the individual speakers.

OOH... BIAMPING

For example, the JRX's WIKD Entertainment uses contain a "tweeter" to produce the high frequencies (1.8KHz to 16KHz), while a 12" "woofer" produces the lower frequencies (60Hz to 1.8KHz). There are two possible ways of connecting the JRX to an amplifier. The first, and the way it's set up for now, uses a 2 Pole conductor to provide a single audio signal. Within the speaker, there is a specialized equipment, called a crossover, which splits the incoming single based on frequency—thereby making sure that the upper range of the signal goes to the tweeter, and the lower range goes to the woofer.

The alternative, is to bypass the speakers internal crossover and instead to give each driver its own amplifier. To do so you would insert a crossover between the mixing console and amplifiers—the idea is the same as before—to split the signal into high and low parts, and then to send them to their respective amplifiers. This is called bi-amping. The advantage of bi (or tri, or quad) amping is that it allows you considerably more control over the sound of the speaker, allows for the speakers to be a little bit louder, and if you do everything right, sound better.

The downside of bi/tri/quad amping is that it is considerably more expensive. Instead of just buying an amplifier and 2 Pole speaker cable, you now have to buy (at a minimum), a crossover, an amplifier for each range of frequencies you wish to drive, and the appropriate cable—all of which gets expensive quick. As such, bi/tri/quad amping is the domain of really high end audio systems.

For the most part, you won't have to worry about this. Both of the JRX systems WIKD Entertainment uses are perfectly happy without being bi-amped.

(ARE LAB GRUPPEN'S THE MACDADDY OF ALL AMPLIFIERS? YEAH PRETTY MUCH.)

AUDIO FUNDAMENTALS

PART 3

DIGITAL AUDIO

· SAMPLE RATE

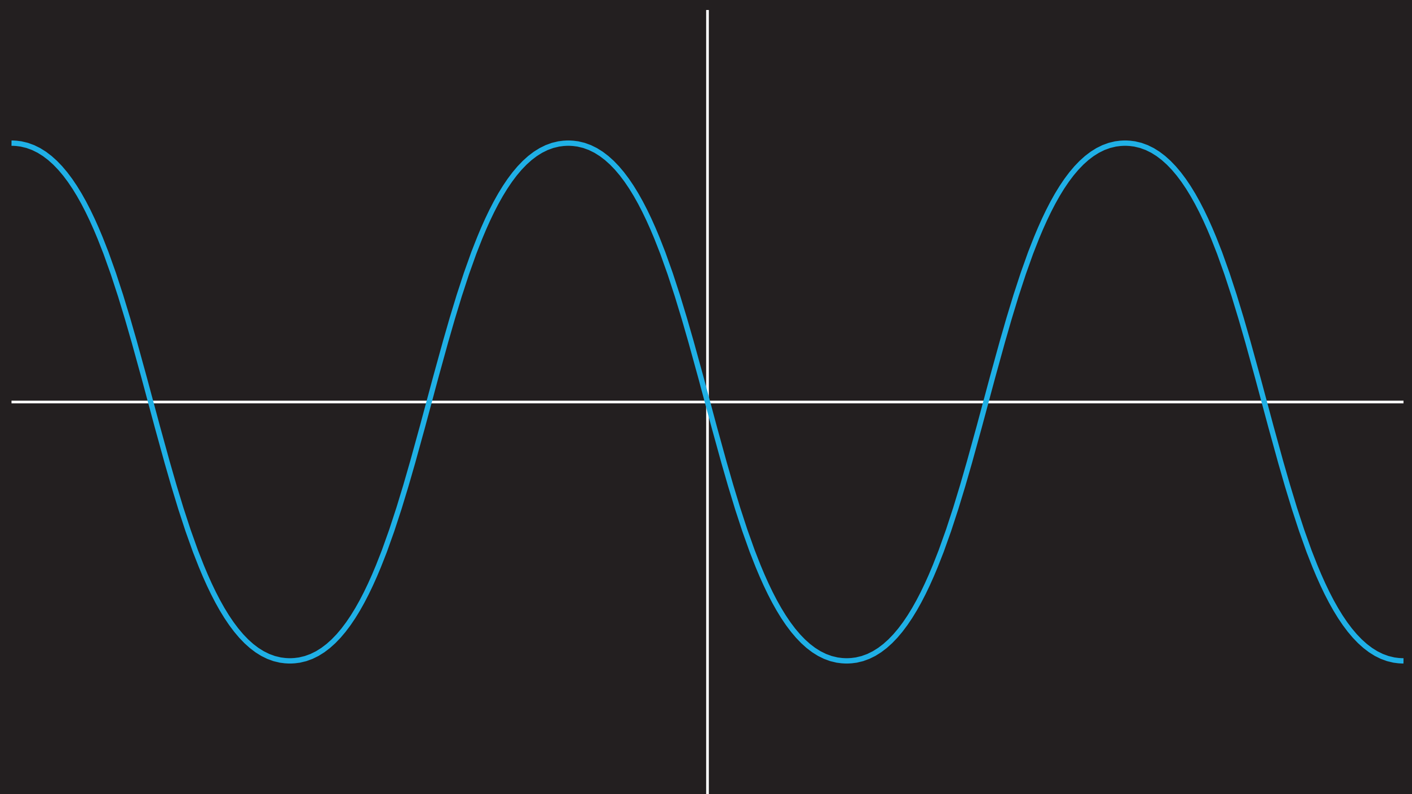
· WORD LENGTH

· DYANMIC RANGE

· AUDIO CODECS

DIGITAL AUDIO

The basics of digital audio are surprisingly easy to understand, but you have to start by understanding analog audio. When you make a sound you create vibrations in the air. If you put a microphone in front of your mouth, those vibrations cause the diaphragm of the microphone to forwards and backwards. The movement of the diaphragm generates an electrical current and all of a sudden you have an audio signal which might look something like this:



(IF YOUR MOUTH IS A TUNING FORK THAT IS...)

When the diaphragm is moved backwards (or away from you), you get negative voltages. When the diaphragm is moved forwards, you get positive voltages. The louder you are, the greater the amplitude of the signal is. The higher pitched you are, the higher the frequency of the waves—and vice versa.

Good? Good.

An Analog to Digital Converter works by repeatedly measuring the voltage of an analog signal as quickly as possible. The faster it can measure the voltage, and the more accurately it can describe the voltage, the better it can approximate the original signal. The rate at which is samples the voltage is called the "Sample Rate". How accurately it can record the measured voltage is defined by the Word Length.

SAMPLE R A T E

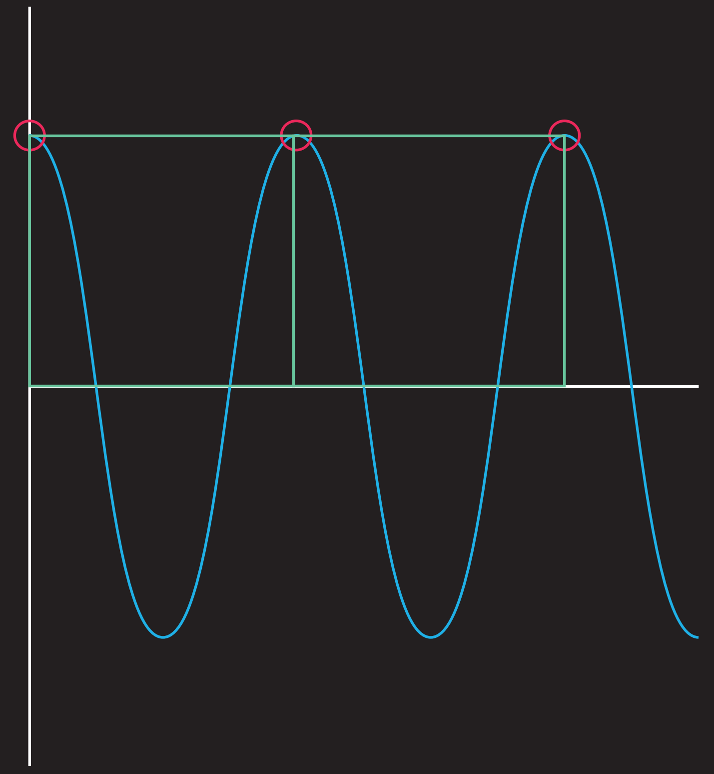
The average human can hear a range of frequencies from 20Hz to 20KHz. That equates to a bandwidth of roughly 20KHz. Many years ago, when the scientists of Bell Labs were first converting analog signals into the digital domain, they discovered that in order to accurately recreate an analog signal from a collection of digital samples, the sample rate had to be at least twice the highest frequency present in the original signal.

In plainer terms, it means that in order to faithfully reproduce signals from 20Hz to 20KHz, you had to sample the analog signal at least 40,000 times per second. This was later standardized to a sample rate of 44.1KHz—and that's the standard audio CD's and WIKD use. There are several other standard sample rates with which you should be familiar with:

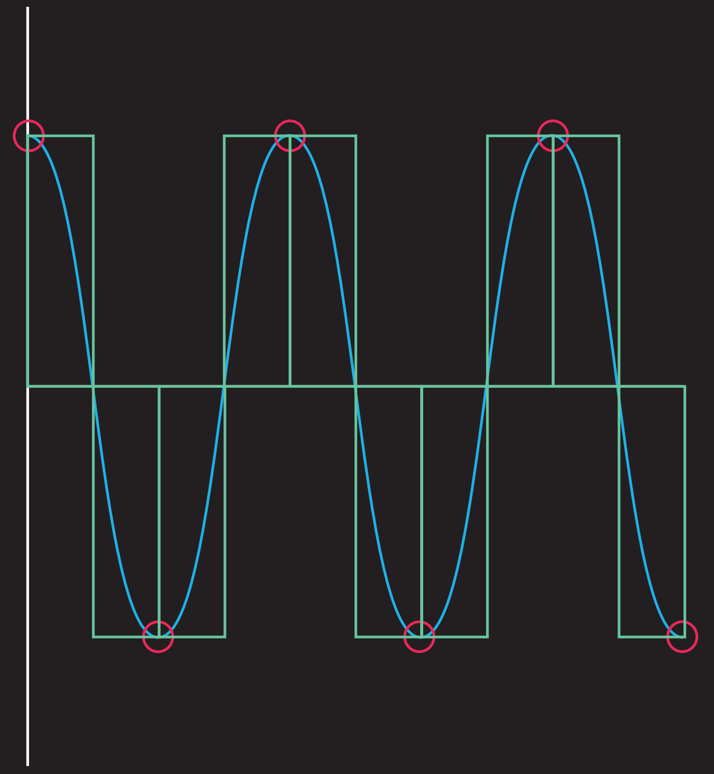
SAMPLE RATE	USES
8 KHz	The standard for Plain Old Telephone Service (POTS), because it just barely covers the frequency range neccessary to understand human speech (300Hz-3400Hz)
16 KHz	The standard used for most high end VoIP phones.
32 KHz	Most commonly used with Microwave STL's (Studio-Transmitter Links).
44.1 KHz	The standard for CD Audio. WIKD uses this for all audio within the studio.
48 KHz	The most commonly used standard for professional audio and video equipment. The Omnia One in the Transmitter Shed upconverts all audio from 44.1KHz to 48KHz before connecting to the Nautel.
96 KHz	The standard for DVD-Audio and some Blu-Ray discs. Available in high quality audio equipment.
192 KHz	The highest sample rate you will likely come across—found on some Blu-Ray discs, and high end audio equipment.

THE NYQUIST SAMPLING THEOROM

Let's say you generate a sine wave with a frequency of 20Hz, and a peak voltage of 1.0V. According to the Shannon-Nyquist theorem you must sample this signal at 40Hz in order to accurately reproduce it.. If you sampled the signal at 20Hz, then your samples will have the same period as the signal---every measurement would be the same! In order to accurately sample the signal you should be able to capture both the peak points within a single cycle. And that's why, in order to accurately sample a 20Hz sine wave, you would have to sample it at 40Hz.



(SAMPLING AT 20Hz)



(SAMPLING AT 40Hz)

WORD LENGTH

Think about the 20Hz sine wave from the previous example. It has peak voltages of 1.0V. If you were to sample that signal at 60Hz, you would record three unique values: -1.0V, 0V, and +1.0V. In order to represent that in binary you would need to use 2 bits, or two binary digits. The leading bit would indicate the polarity of the signal—a "0" would indicate negative polarity, while the second bit would indicate the value—"0" for 0V, and "1" for 1.0V.

However, most audio signals are considerably more complex than a simple sine wave. As such 2 bits isn't really enough to accurately describe audio—in fact the only thing you would be able to produce is a square wave!

The solution then, is to use more bits—the more bits we use the more unique voltages we can measure. The number of bits used is called the word length. There are three common word lengths: 8 bits, 16 bits, and 24 bits.

A word length of 8 bits yields 128 unique values. A word length of 16 bits yields over 32,000 unique values, and a word length of 24 bits yields over 8 million unique values. The math works like this:

For every bit there are two possible values: "0" or "1". The number of possible combinations is determined by raising 2 to the exponent "n", where "n" is your word length minus one (remember that you have to reserve one of the bits to indicate the polarity of the voltage).

The difference in quality achieved by increasing the number of bits is significant. 8 bits is considered to be the smallest possible word length you can use and still understand human speech. In order to generate an acceptable music recording you have to jump up to 16 bits—the standard for CD Audio. Once you make the jump to 24 bits and above, you are able to negate the majority of the problems caused by converting from analog to digital to analog. 24 bit audio is currently considered to be the standard for very high quality audio.

DYNAMIC RANGE

Word Length really comes into play when determining the dynamic range of a signal: that is, the difference between the loudest and softest points. In the same way that most humans can hear sounds ranging in frequency from about 20Hz to 20Khz, we can hear a dynamic range of about 140dB.

The maximum dynamic range of a digital audio system is a direct result of the word length. A 16 bit signal has a theoretical maximum dynamic range of about 96dB, and a 24 bit signal has a theoretical maximum dynamic range of about 144dB.

The actual dynamic range, however, is also a function of the all of the associated I/O and processing circuitry. If the input circuit can only handle a dynamic range of 60dB, then it simply doesn't matter how large your word is—your maximum dynamic range will be 60dB.

AUDIO CODECS

In the same way that analog audio can be stored and transferred in a variety of different ways, so to can digital audio. The way in which the digital audio is stored is determined by the audio codec. There are three different types of audio codecs: Uncompressed, Lossless, and Lossy.

UNCOMPRESSED AUDIO

The most popular method for encoding uncompressed audio is called Linear Pulse-Code Modulation (You'll usually see it referred to as just PCM audio however). It is used for CD, DVD, and Blu-Ray audio, as well as being the preferred audio format for WideOrbit Automation. Typically files that are encoded using LPCM will use an extension of .WAV or .AIFF (If you're using a Mac). The downside of using PCM audio is that it takes a considerable amount of storage space—1 minute of 16bit PCM encoded audio with a sample rate of 44.1KHz (the standard for CD Audio), takes about 10MB.

LOSSLESS AUDIO

The idea behind lossless audio codecs goes something like this: if you were to look at the average pop song, there will inevitably be points where the data is repeated. If the voltage was the same for 230 samples in a row, instead of writing the same voltage 230 times, you could simply write: "1.5V for samples 0-230".

Lossless audio does not damage the signal in any way—there should be absolutely no difference between the original waveform and the reconstructed waveform.

The most popular formats for losslessly encoding audio are FLAC and WMA Lossless. These can typically compress a file to about 50-60% of its original size.

LOSSY AUDIO

Whereas lossless audio codecs work by storing the information in a more efficient method, lossy codecs look at the original audio file and determine what parts of the file you won't notice going missing. For example, imagine sticking your head inside a kick drum. You would hear the kick drum extraordinarily well—so well that you might not appreciate how skilfully the ride is being played at the same time.

Audio that is encoded using a lossy codec is permanently damaged—when reconstructed it will not be the same as the source signal. Moreover, the quality can never be improved—it can only be continually reduced.

However, if you use a codec of sufficiently high quality, the difference between the original and the lossy version can be almost imperceptible. Try listening to the songs from TM Studios which are encoded in both Linear PCM and MP3 and listen for any differences.

Getting rid of all that extraneous information pays off when it comes to file size. Remember how a song encoded using 16bit/44.1KHz PCM will use 10MB per minute? An MP3 file using the 128kbit/s setting will use just under 1MB. If you use the 192kbit/s setting (the standard we use for our online stream), the file grows to about 1.5MB. The highest quality MP3 files use a setting of 320kbit/s and result in a file size of 2.3MB.

Below are two copies of Get Lucky by Daft Punk Ft. Pharrel Williams. One is encoded at 320 kbps, while the other is encoded at 128 kbps. Try and note the differences.

Obviously then, the question is why don't we use MP3 files for everything? Mostly, it's because WideOrbit Automation only works with PCM audio files.

However, it's also because we employ a significant amount of audio processing in order to get the best/loudest on-air sound possible. MP3 files, even 320kbit/s files, don't benefit from the processing nearly as much as PCM files do. In fact, in some cases, it can even make the file sound worse. Listen to a song on-air played through Automation, and then listen to the MP3 version on-air to see for yourself.

AUDIO
PART 4
FUNDAMENTALS
LEVELS

- GAIN
- NOMINAL OUTPUT
- INTERFACING WITH THE AUDIO ENGINE
- DIGITAL AUDIO SIGNAL LEVELS
- METERING

GAIN

Very simply put, gain is a measure of how much the "magic smoke" (technically, an electrical circuit...) can amplify a signal. It's useful, because a lot of signals need amplification. For example, the level that a microphone produces is extraordinarily low—so you use a preamp to amplify that signal to where you can work with it. Eventually, you'll probably want to hear it through some speakers—and you'll amplify it again. Each time you do, you're increasing the gain.
Mathematically, the formula for calculating gain looks like this:

$$GAIN = \frac{SIGNAL\ OUTPUT\ LEVEL}{SIGNAL\ INPUT\ LEVEL}$$

Most commonly, gain is measured on a logarithmic scale, using decibels. The decibel is a logarithmic unit which indicates the ratio of either a "power quantity" or a "field quantity" relative to some reference level. How you calculate gain changes depending on whether you are using a "power quantity" like acoustic intensity, or a "field quantity" like the voltage of an electrical signal—which is what we're interested in. Mathematically, the equation for calculating the dB gain of a field quantity looks like this:

$$dB\ Gain = 20\log\frac{VOLTAGE\ OUT}{VOLTAGE\ IN}$$

NOTE: Because gain operates on a logarithmic scale, if you were to amplify the signal of a voltage by 6dB the voltage would be twice as much. So applying 6dB of gain to a signal measured at .775V would result in 1.5V.

However, if you're working with Power Quantities (like SPL or the Watt output of an amplifier), then doubling occurs every 3dB. So if your SPL meter measures a dB output of 96dB, 99dB would be twice as loud. Thank physics and try and keep it straight.

However, if all you say is "an electrical circuit has a gain of 6dB", all you've said is that the magic smoke makes signals twice as loud as they were before. You have no idea of how idea of how loud it was before, and you have no idea how loud it is now, only that it's twice as loud as it was before. That's where reference levels come in.

There are only two reference levels you need to be familiar with: dBu and dBV. A suffix of dBu means that it's with reference to .775V, dBV, with reference to 1.0V. Professional audio equipment uses dBV for signal level measurements—consumer audio equipment uses dBu.

NOTE: You will occasionally see dBm. dBm indicates a reference level of 1 milliwatt. It can be converted into a voltage level, provided the impedance is specified. Typically, if you see this with relation to audio equipment, it indicates a an impedance of 600Ω--which in turn results in a voltage of .775V. Therefore, for impedances of 600Ω, dBm and dBu are equivalent.

NOTE: All the voltages here are RMS, or Root Mean Squared voltages. It's a complicated term, but basically it means average.

NOMINAL OUTPUT LEVEL

All audio equipment, whether it's consumer or professional, has a nominal output level—an ideal level at which it operates. If it operates lower, extra noise gets added to the signal, and if it operates significantly above, then the signal gets clipped or distorted.

There's a couple of other terms you should be aware of: Dynamic Range, Signal to Noise Ratio, and Headroom.

Dynamic Range: The difference between the maximum output level and the minimum output level, or noise floor.

Signal to Noise Ratio (SNR): The difference between the nominal output level and the noise floor.

Headroom: The difference between the nominal output level and the maximum output level.

In a perfect world, the average level of your signal will hover around the nominal output level, with peaks approaching the maximum output level. Well-designed professional audio equipment will typically have a maximum output level between +24dBu and +28dBu.

Pretty much all consumer equipment has been standardized to have a nominal output level of -10dBV, while professional audio equipment has been standardized to have a nominal output level of +4dBu. Every so often, you will find equipment that gives you the option of choosing which output level to use. For WIKD you should always choose +4dBu.

NOTE: -10dBV is about .316V RMS, while +4dBu is about 1.23V RMS.

Equipment that operates at a nominal output level of either -10dBV or +4dBu is said to operate at Line Level.

There are two other groups of signal level that you should be aware of: Mic level and Instrument level. Instrument level, as you would imagine, is the direct output of an instrument, such as a guitar or keyboard. Mic level refers to the output directly from a microphone. Typically Mic level signals are the quietest—which is why they are typically run through a microphone preamp to amplify the signal to line level (-10dBV or +4dBu). Mic level signals are (usually) the quietest of all types of signals.

Instrument level signals are typically much hotter than mic level signals, as well as operating at a much higher impedance. Typically, in order to interface an instrument with a mixing console, the instrument will be plugged into a DI (Direct Inject) box that outputs a mic level signal, which can then be plugged into a mic preamp. You should avoid connecting instrument level signals directly into a microphone preamp, unless it specifically states that it can be used as an instrument input.

A SMALL NOTE ON DIGITAL SIGNAL LEVELS

Thankfully, there is really only one standard for measuring digital audio signal levels: dBFS, or decibels relative to Full Scale. The maximum level of a digital signal is 0dBFS, all other levels will be negative values. 0dBFS indicates that all of the bits in the word have been set to “1”. It is impossible to digitall represent anything louder.

Technically, dBFS cannot be used for analog levels, nor is there a standard conversion factor. Instead, you can use a reference level set by the equipment manufacturer. For example, Logitek sets -20dBFS=+4dBu, while Omnia sets -18dBFS=+4dBu.

SIGNAL METERING

There are two big reasons for metering audio signals: level and loudness. Loudness metering is primarily the domain of Europeans and TV broadcasters who have to adhere to strict standards. American radio is not limited by these same regulations and therefore our focus is on mostly on level metering.

VU METERS

VU Meters are the most ubiquitous of all meters, and probably the ones you have most likely seen before. They are designed to reflect the average volume of material, and respond in a similar fashion to the human ear. As such, they are unable to indicate the peak levels of content.

A typical VU Meter will have a scale ranging from -20VU to +3VU, with 0VU representing the nominal operating level.

For example, if you were using a VU Meter to monitor a professional quality CD player (with a nominal output level of +4dBu), then the VU Meter would be set such that 0VU=+4dBu. Most quality audio equipment has a maximum output level of between +20dBu and +24dBu. By setting 0VU=+4dBu, you ensure that there is approximately 20dB of "headroom" for your peaks and more... expressive content.

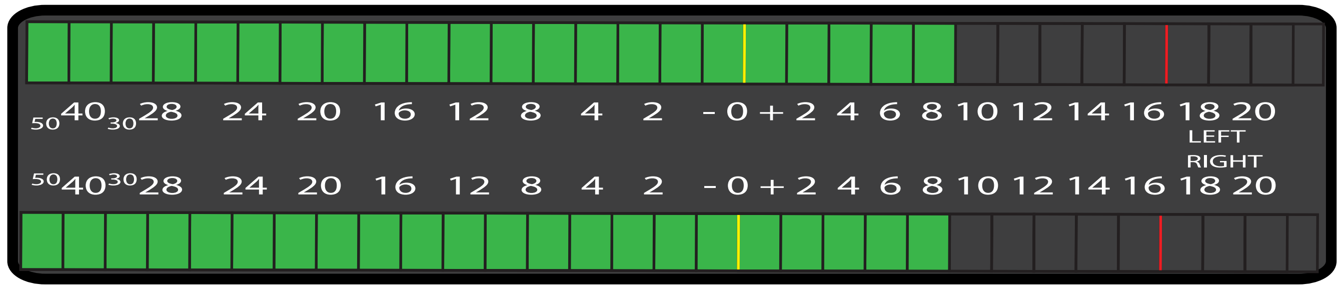
PEAK METERS

Whereas a VU meter will intentionally ignore super-fast peaks, a Peak meter is designed to respond as quickly as possible, and therefore show the exact level at every instant. Peak meters are especially useful with regards to digital audio, because digital clipping (generally), sounds horrendous—even if the signal only exceeds 0dBFS for a single sample. Analogue clipping happens somewhat more gently, and in some cases, can even sound good.

THE LOGITEK METERS

The meters on the Logitek Numix are combination VU/Peak meters. That is, VU style levels are indicated by the solid green bar, while peak levels are indicated by the red bar. 0VU corresponds to -20dBFS. For monitoring analogue sources, make sure you have configured the appropriate gain settings in AEConfig. (I.e. equipment with a nominal output of +4dBu should have the normal signal level set to +4. This will ensure that outputs of +4dBu correlate to 0VU after the AD conversion.)

Ideal operation of the Logitek should result in programme levels reaching -20dBFS, with peaks reaching -12dBFS to -6dBFS on particularly loud material. In reality, however, most music WIKD receives is incredibly compressed and is typically mastered to levels between -6dBFS and -3dBFS—levels far in excess of the "nominal" range. As a result, normal operation should see programme levels reaching -12dBFS, with peaks reaching -6dBFS.



AUDIO
PART 5
FUNDAMENTALS
SIGNALS & PROCESSING

· PHASE & POLARITY

· EQUALIZATIONS

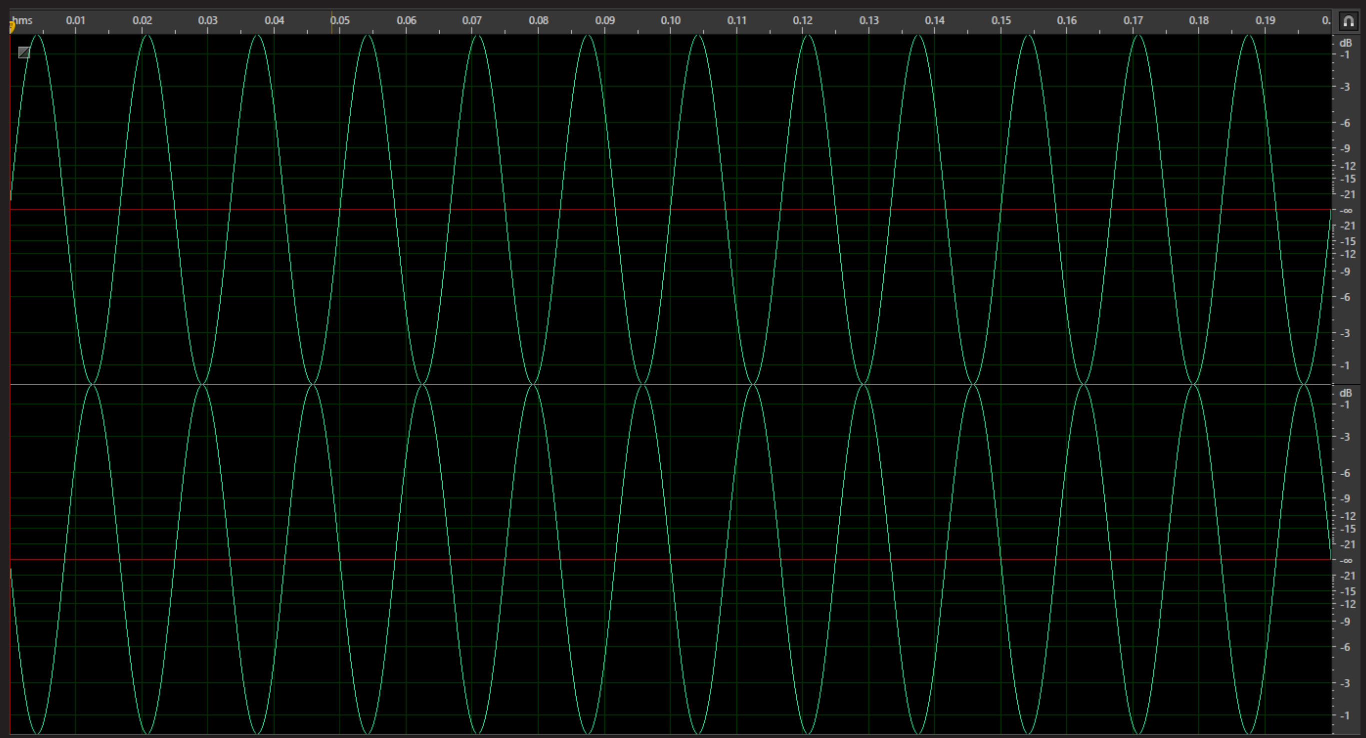
· DYNAMICS PROCESSING

· BROADCAST AUDIO PROCESSORS

PHASE & POLARITY...

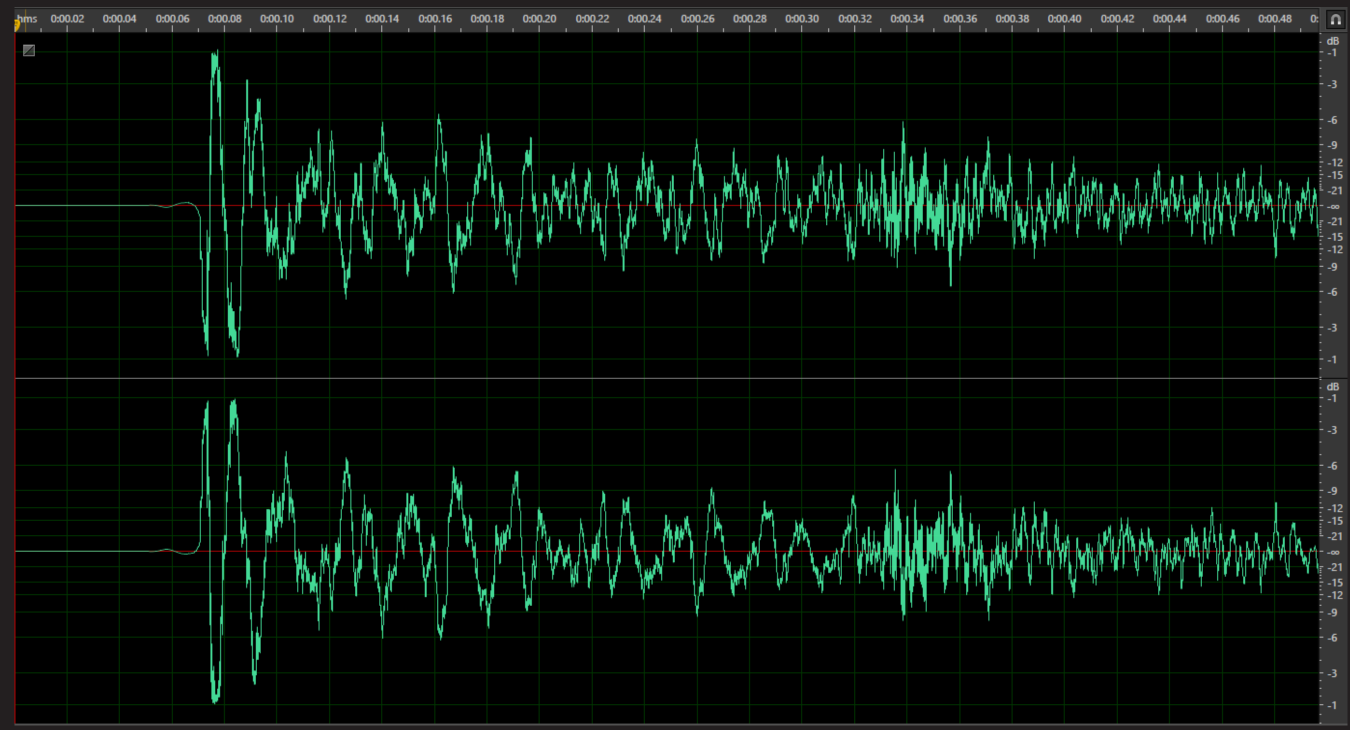
You will often hear the terms phase and polarity used interchangeably. They are not, however, the same, and it is important that you understand both terms, and their differences.

In the audio world, the phase of the signal refers to the amount by which the signal has been delayed relative to the original signal.



Phase is measured in degrees. A signal that is 180° out of phase with itself (like in the figure above), will actually null itself out if the signals are summed—you wouldn't hear anything!

However, and this is important, the only way that can happen is if the signals have the SAME frequency. This is because the phase of the signal is relative the frequency of the signal. You could not, for example phase delay the signal to the right by 180°, because it is not a periodic waveform. What you can do, however, is invert the polarity.



The polarity of a signal simply refers to whether a signal has a positive or negative amplitude at a specific point in time. If you invert the polarity of a signal, you simply flip the signal—negative becomes positive, and vice versa.

If you were to invert the polarity of a signal, and then sum it with the original signal, the sum would be 0. This is what happens in balanced circuits remember?

The ONLY time that a polarity inversion and a phase delay are the same, is when you are considering a simple sine wave—that is, a periodic sine wave of a single frequency that has been delayed by 180° with respect to itself—like in the figure to the left.

A SMALL NOTE ON MAGIC SWITCHES

Most well equipped audio consoles will usually have a small button that will claim to flip the phase of the incoming signal. What they mean to say, is that it will switch the polarity of the signal—it will invert the signal. If they could phase swap complex signals they would be magic.

These buttons can be used to reduce (or induce) phasing problems, or to correct problems with bad cables or nonstandard equipment.

Just to re-iterate:
*The **phase** of a signal can be delayed or shifted.*
*The **polarity** of a signal can be switched.*

LET'S BE CLEAR,

PHASE &

POLARITY

ARE NOT

THE SAME DAMN

THING

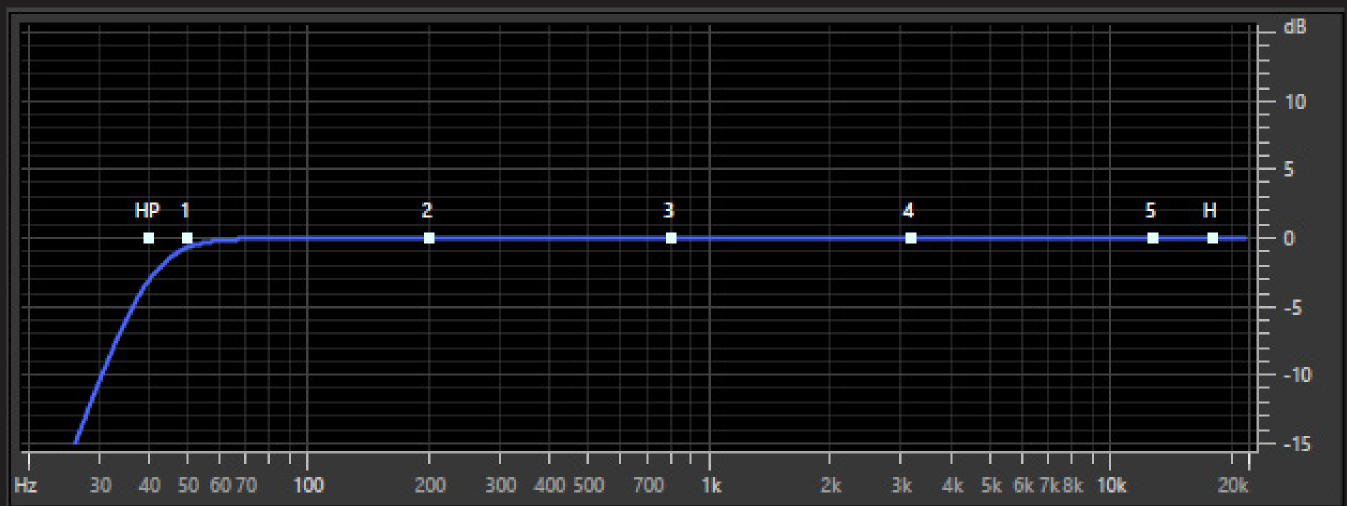
EQUALIZATION

The basic idea of equalization is that of boosting or reducing certain frequency ranges in an audio signal. The bass and treble knobs on your car radio are an example of basic equalizers. In professional audio, however, equalizers are slightly more complicated.

PASS FILTERS

The simplest forms of EQ are the high and low pass filters. A high pass filter applied at a certain frequency will “roll off” the frequencies below that point, while letting everything higher than that “pass through”. Likewise, a low pass filter does the same thing in the opposite direction.

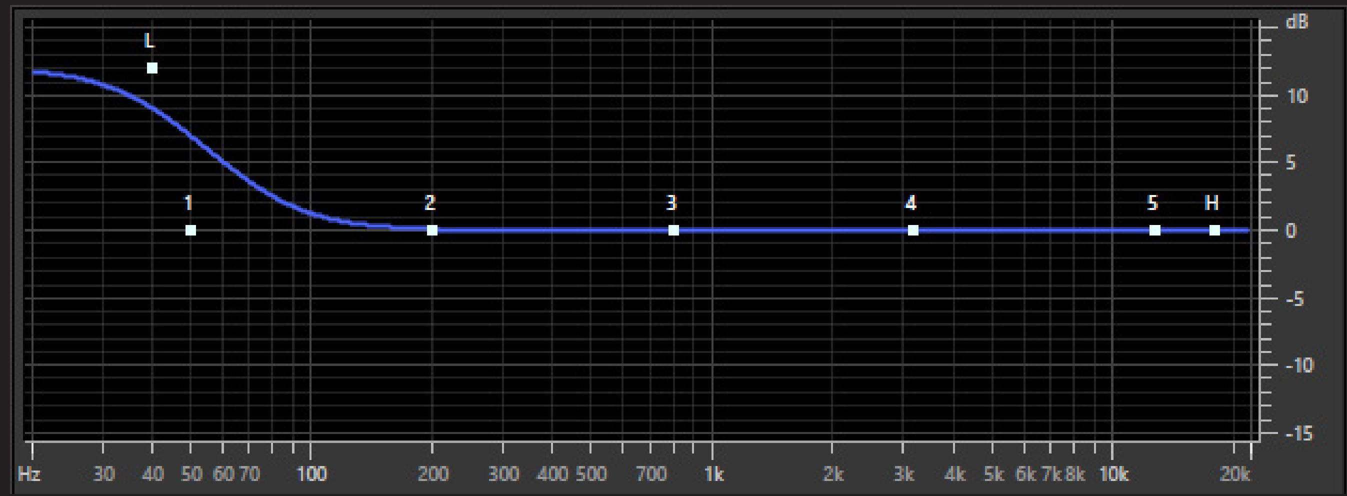
High pass filters are much more common, and are typically used in conjunction with microphones to reduce the low frequency rumble from sources.



(LISTEN TO A HIGH PASS FILTER IN ACTION)

SHELVING FILTERS

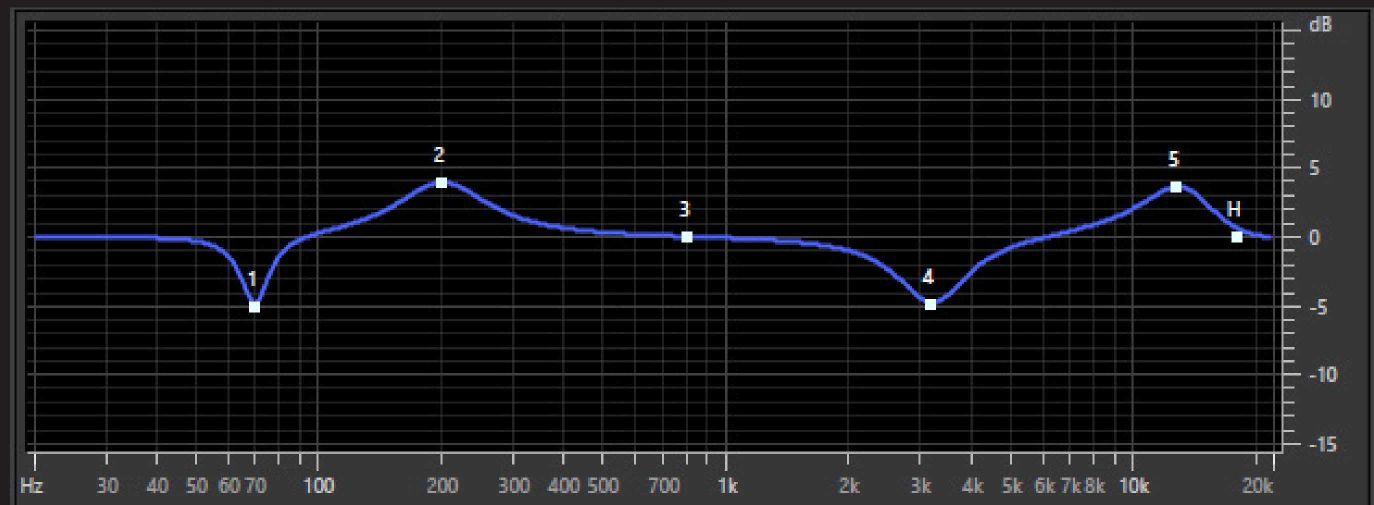
Like pass filters, shelves can be either high shelves or low shelves. A high shelf filter will boost or reduce frequencies above a selected frequency, while a low shelf will boost or reduce frequencies below a selected frequency.



(LISTEN TO A SHELVING FILTER IN ACTION)

PARAMETRIC EQUALIZERS

In a graphic EQ that bandwidth of the affected signal (how many of the center frequencies buddies will be affected essentially) is fixed, as is the center frequency. Parametric EQ’s allow you adjust not only how much of a boost or reduction is applied, but to what frequency, and the amount of surrounding frequencies that will be affected. Because of the increased freedom offered by parametric EQ’s, they typically use far fewer bands—in fact the equalizers used by the Logitek Audio Engine are parametric style EQ’s and consist of only 4 bands—low, low mid, high mid, and high, which is usually enough.



ARBITRARILY

SMILEY FACING

THE

EQUALIZER CURVE

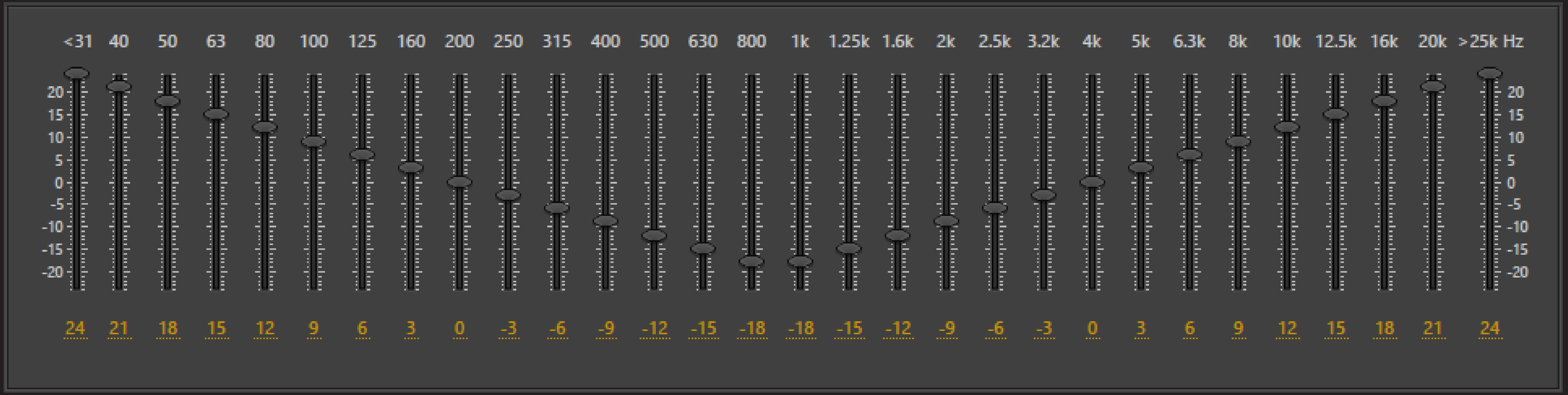
IS NOT

ACCEPTABLE

(SAD FACES ARE TOTALLY OK THOUGH)

GRAPHIC EQUALIZERS

Graphic equalizers are pretty simple to use. They have little faders on the front that correspond to a specific frequency. Move the fader up, and you boost that frequency and some of its buddies. Move it down, and you reduce that frequency and its buddies. They're a lot more common in live sound and recording than they are in broadcast.



(THIS IS AN EQUALIZER THAT HAS BEEN "SMILEY FACED". IT'S SAD ON THE INSIDE)

DYNAMICS PROCESSING

Simply put, the dynamic range of an audio signal is the difference between the softest and loudest point. Dynamics processors exist to do funny things to the dynamic range of a signal.

COMPRESSORS & LIMITERS

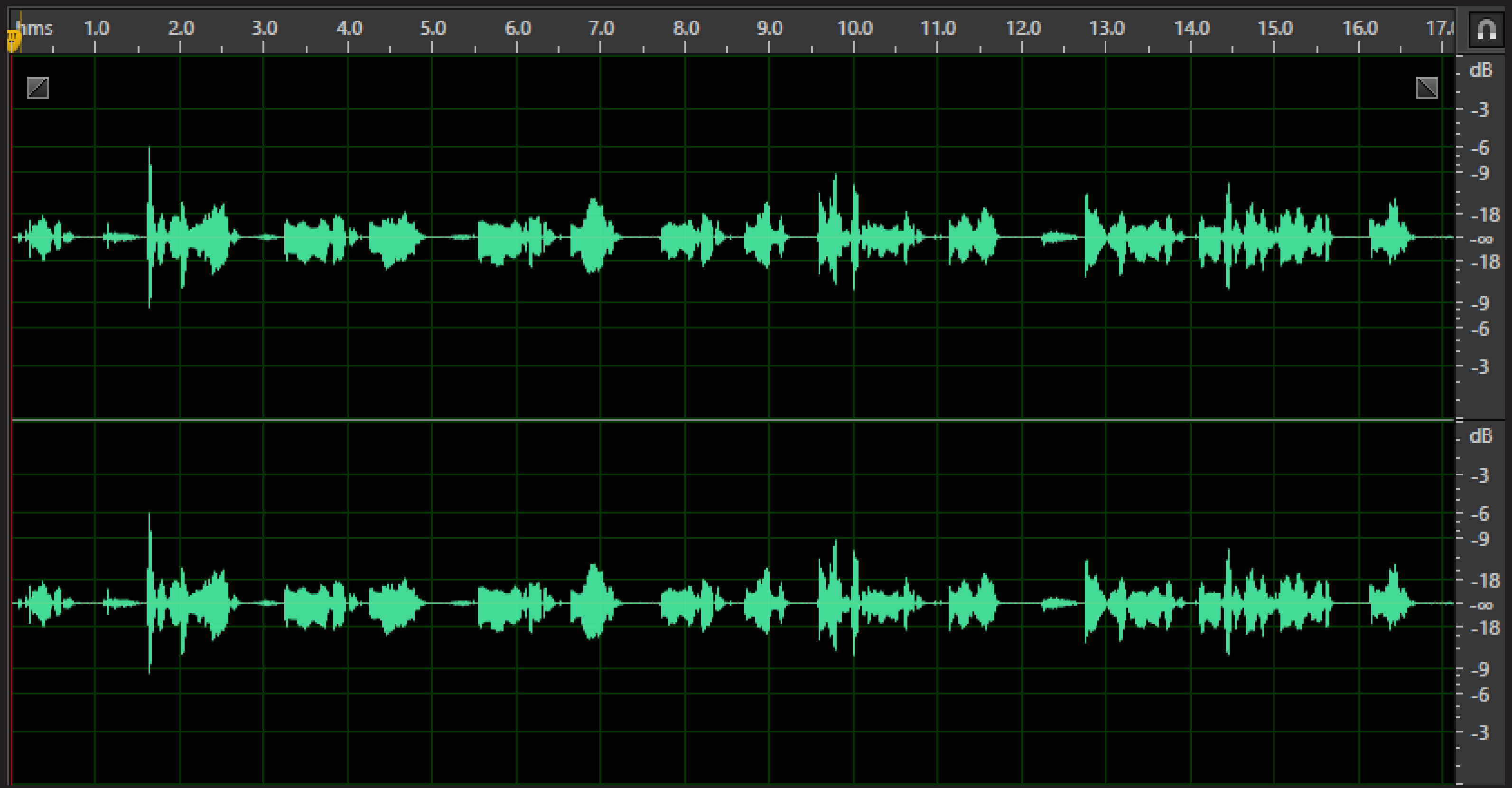
Compressors are the most common form of dynamics processors. Simply put, a compressor reduces the dynamic range of an audio signal by making the loud parts softer. In slightly more technical terms, any time the incoming signal goes over a "threshold" level, the compressor starts reducing the gain applied to the signal—in effect making it quieter.

Most professional compressors will have five major settings: threshold, ratio, attack, release, and output gain. The threshold defines the point at which the compressor starts reducing the signal. The ratio defines the amount by which the signal is reduced—a ratio of 1:1 would not affect the signal at all, while a ratio of 10:1 would mean that for every 10dB increase in the input signal, the output signal would only be increased by 1dB. (Incidentally, a ratio of 10:1 is considered very high, most ratios will be in the range of 1-5:1.)

The attack time controls how fast the compressor will start reducing the output signals gain once the signal crosses the threshold, while the release time controls how quickly the compressor will stop affecting the signal once it goes below the threshold.

One of the reasons for using a compressor is to ensure that the peaks are not significantly louder than the rest of the audio—think of when you watch TV and turn down the volume because a commercial is so much louder than the TV show. When you do this however, you end up with a lot of headroom—and that's where the output gain comes in. Once the signal has been compressed, you can safely turn up the volume of everything without having to worry about the peaks causing clipping.

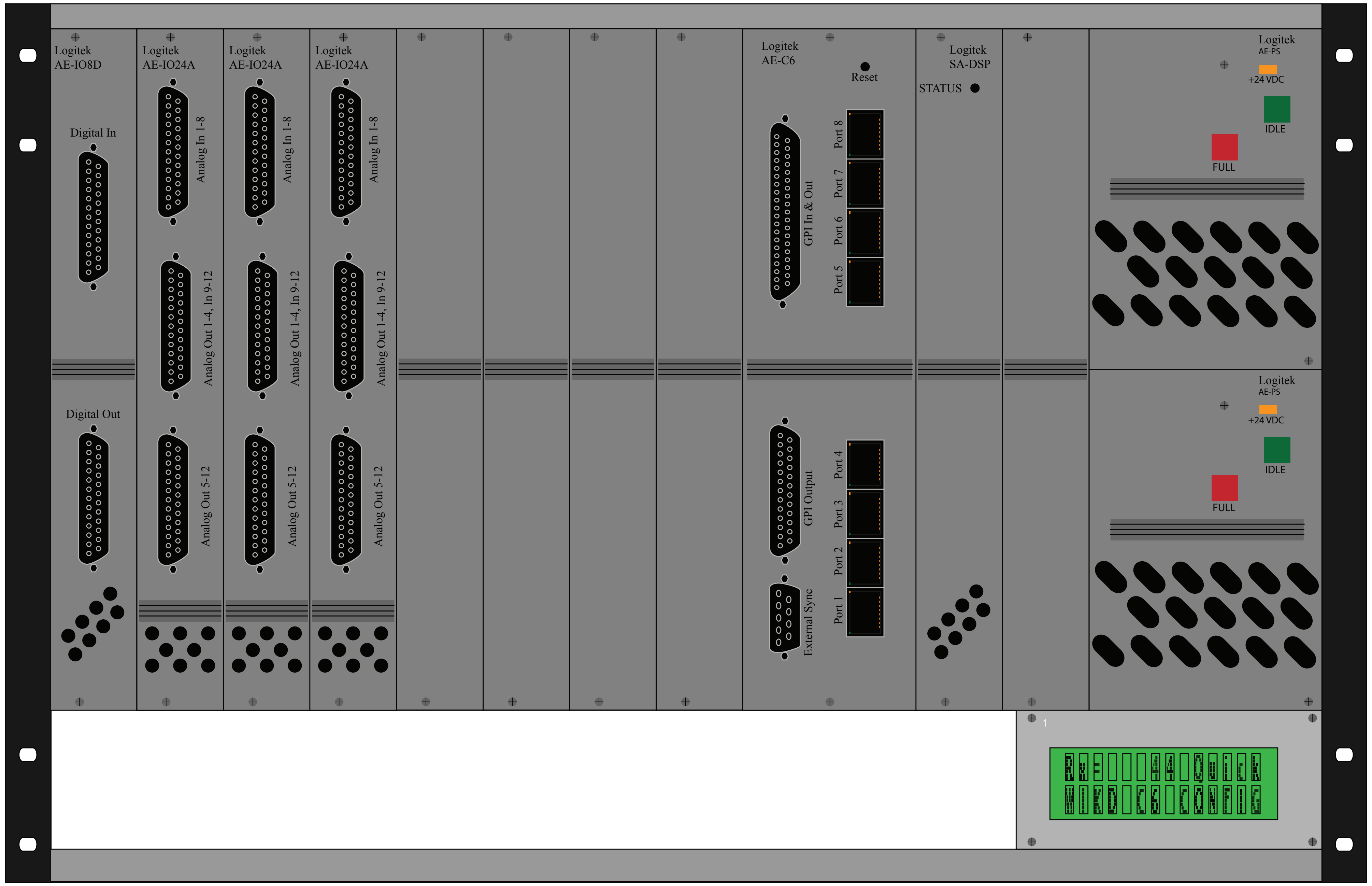
A limiter is a specialized version of a compressor wherein the ratio is set to 10:1 (or greater) in order to prevent the signal peaking over a set threshold--this is especially useful as a safety device to prevent digital clipping, which sounds *awful*.



The top signal is a recording of WIKD's VoiceOver artist, Rachel McGrath. No processing has been applied. The image below is the signal after being processed by a compressor and brickwall limiter. Click on the images to hear the recordings.

THE AUDIO ENGINE ROUTING FUNDAMENTALS

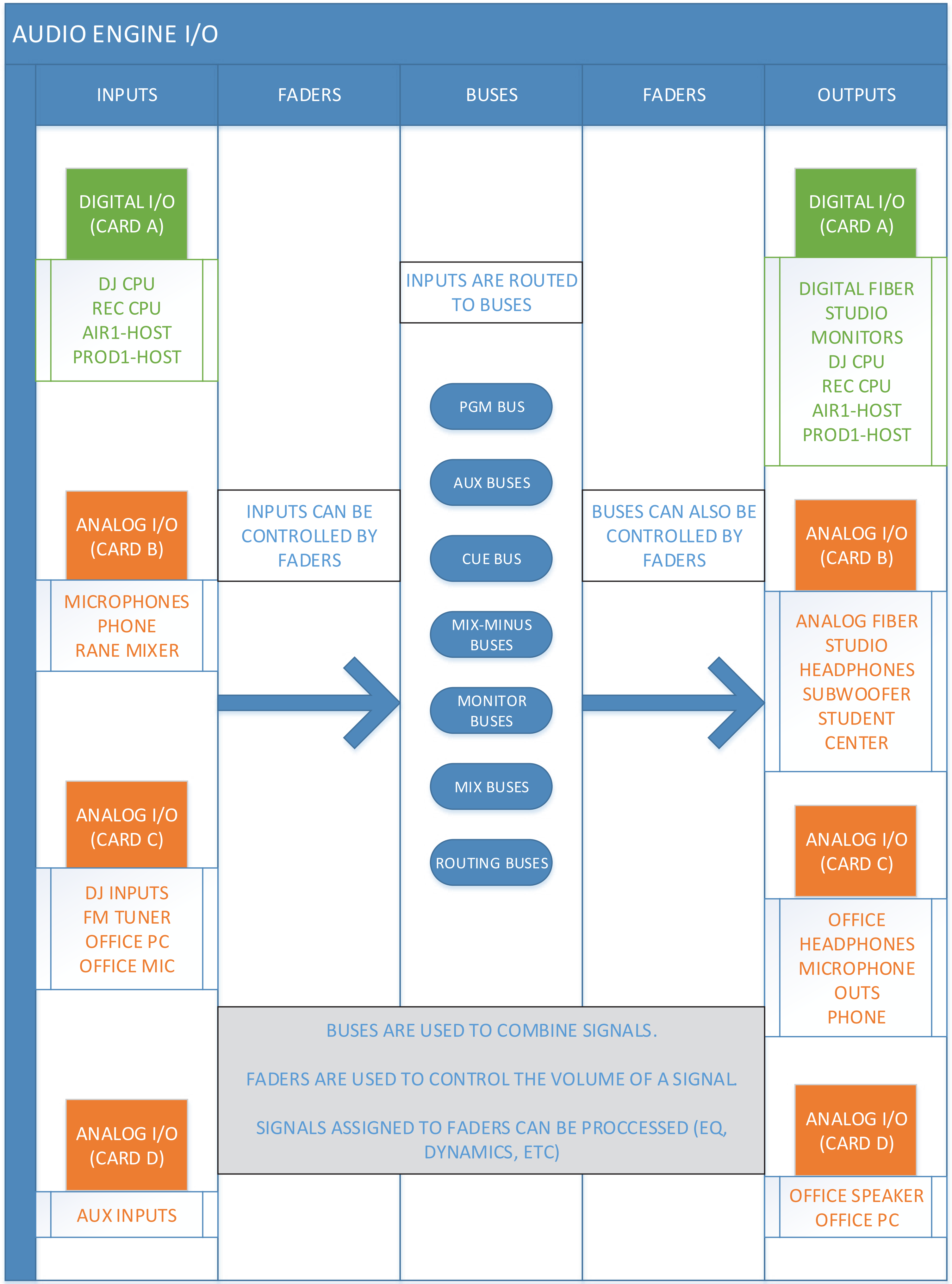
PART 6



THE LOGITEK AUDIO ENGINE

The Audio Engine is technically an Audio Router, which itself is the bastardized son of a basic mixing console. At the most simplistic level it's responsible for accepting a bunch of physical inputs, applying some processing to them, mixing some of the signals, and then sending those mixes to various physical outputs.

The Audio Engine can be divided into three major sections—the physical inputs, the virtual buses, and the physical outputs. Various forms of signal processing can be applied at each of the stages.



THE INPUT & OUTPUT STAGES

The input stage consist of the physical I/O connections on the Audio Engine. Here the audio is connected, and if necessary, gain is applied to boost or reduce the incoming signal. Then, if the incoming signal is analog, it is converted to a digital signal. If it's a digital signal, the Audio Engine ensures that it is using the same sample rate as the audio engine.

Inputs can then be assigned to a fader, where you can adjust the volume of the signal, as well as the trim, and apply EQ and compression to that input.

Faders don't have to actually exist—if you look in Supervisor, you will see that we are currently using 24 faders, even though the Numix only has 18 physical faders. The only advantage to having a physical fader is that you can physically control it, as opposed to having to control it through Supervisor.

Trim is gain in the digital world—if you find that even after adjusting the gain the signal still isn't loud or soft enough, you can adjust the trim by 10dB. However, if a signal was too noisy or too distorted before being converted to digital, adjusting the trim won't fix that.

Pretty simple right?

The output stage operates exactly the same as the input stage, except that if you assign an output to a fader you can only adjust the volume.

THE BUS STAGE

(PART 1)

Here, all the audio signals are driven to school.

THE BUS STAGE

(PART 2)

The simplest definition of an audio bus is a collection of audio signals. The most important bus is the Program (or PGM) bus. This is what goes on-air. Operation is simple—physical inputs are assigned to faders and the faders are assigned to the PGM bus. The amount of signal going to the PGM bus is determined by the fader's level.

THE ROUTING BUSES

One of the great features of the Audio Engine is that it includes a built in broadcast delay that can delay audio by up to 30 seconds. (However, the Numix can only show a delay time of up to 9.9 seconds so....)

In order to delay audio, you must send that audio through the ROUTE buses. There are three available ROUTE buses—ROUTE 1, 2, and 3.

Currently we use ROUTE 1 to delay PGM audio.

THE AUX BUSES

The Audio Engine has 8 Auxiliary buses. These work in exactly the same way as the PGM bus. They can operate in AFL or PFL mode. AFL mode, or After Fader Listen mode, means that the amount of signal going to the bus is determined by the fader level. PFL mode, or Pre Fader Listen mode, means that the signal will not be affected by the fader level.

In addition, the AUX buses can also be set to operate in "Independent Mode" wherein the fader does not have to be turned on in order for it to send signal to the bus.

AUX's 1-3 are unique buses. AUX's 4-8 are shared with the Mix-Minus buses.

THE CUE BUSES

The CUE bus works exactly like the AUX and PGM buses with one exception—if you assign an input to the CUE bus it is instantly routed to the Numix's internal speaker. By default the CUE bus operates in PFL mode.

THE MIX-MINUS BUSES

A mix minus bus is slightly more complicated than an AUX or PGM bus, in that it's a mix of other buses, not just inputs. A good way of understanding what a mix minus bus is, is to understand why you might use one.

Consider that you have a caller that you wish to have a conversation with on-air. You would need to be able to hear yourself, the caller, and any background music or FX. The caller, on the other hand, only needs to hear you and any background music/FX—because of the delay in phone transmissions, if you fed their own voice back to them they would quickly become very confused.

The solution is a mix-minus bus, made up of the PGM bus MINUS the phone callers audio. In slightly more technical terms a mix minus bus is comprised of two parts: the source bus, and the minus bus. For each mix minus bus, the Audio Engine allows you to choose from five different source buses: PGM, CUE, AUX1, AUX2, and AUX3.

The Audio Engine allows for 25 mix minus buses. Buses 1-3 are mono buses only. Buses 4-25 can be linked together to be used as a stereo bus. Buses 16-25 are shared with AUX 4-8

You can assign multiple inputs to a single mix minus bus, but each input can only be assigned to one mix minus bus.

Mix Minus buses are created using AEConfig. Use the System tab to select the source bus for each mix minus. You can assign a mix minus bus to an input on the input configuration page.

THE MONITORING BUSES

The Audio Engine includes three monitoring buses: MONITOR, HEADPHONE, and STUDIO. The monitor buses act as a sort of "super" bus, in that they can be used to listen to buses, such as the AUX or PGM buses. In addition, each of the monitor buses has a physical volume controller on the Numix.

Currently the MONITOR bus is mapped to the studio monitors, while the HEADPHONE bus is mapped to the studio headphones. Somewhat counterintuitively, the STUDIO bus is actually mapped to the office speakers.

By default, the monitor buses can only "listen" to the PGM bus, and AUX's 1-3. This can be adjusted in the Surface Settings tab of AEConfig.

MEET THE
AUDIO ENGINE

The heart of WIKD's operation is the Logitek Audio Engine. At the most basic level, it is essentially an audio mixing console. It can accept a multitude of inputs, apply various forms of signal processing to them (Gain, EQ, Compression), and then mix those inputs together and send them to various outputs.

However, in actuality, it's considerably more complicated than that. Whereas in a typical mixing console everything is contained in one chassis, the Audio Engine splits things up.
For starters the user interface, the Numix Surface, which consists of the faders, the knobs, and the buttons, is completely separated from the actual mixing console (the Audio Engine).

Likewise, the Audio Engine is also separated into multiple parts, or cards. The cards are installed into the Frame, which also contains the power supplies, and a backplane that allows the cards to communicate with each other. By utilizing this modular construction, the Audio Engine can be customized as needed. At a minimum, an Audio Engine must have an I/O card, a DSP card, and a router card installed. WIKD's Audio Engine contains a single Digital I/O card, three Analog I/O cards, a DSP card, and a routing card.

LOGITEK AE-IO8D

D I G I T A L   I / O   C A R D

The IO-8D is a digital Input/Output card. It is cable of supporting 8 stereo AES or S/PDIF inputs, as well as 8 stereo AES or S/PDIF outputs. The inputs use the top DB25 connector, while the outputs use the bottom DB25 connector. In addition to serving as a connection point, the IO-8D can adjust the gain of individual inputs and outputs, as well as perform sample rate conversions (for example, if Audio Engine uses a sample rate of 44.1KHz, the IO-8D will adjust the sample rate of any inputs to match). The IO-8D is installed in Slot A. (Logitek recommends installing digital cards in the left-most available slot to reduced EMI/RFI.)

LOGITEK AE-PS

P O W E R   S U P P L Y

The Audio Engine uses load sharing redundant Power Supplies. If only one power supply is installed it should be placed in the bottom slot. Both power supplies must be configured in AEConfig for load sharing or hot swap features to work.

LOGITEK AE-C6

C O N T R O L L E R

The C6 card is the communications center for the audio engine. This is where control surfaces, like the Numx, are connected, as well as the Supervisor monitoring software. The C6 card is also where the Audio Engine configuration is stored—essentially the C6 card is responsible for telling all the various audio signals where to go, and if they need to be processed. In addition, card also contains 15 GPI’s and 15 GPO’s, which are used for interfacing the Audio Engine with other equipment.

LOGITEK AE-IO24A

A N A L O G   I / O   C A R D

The IO-24A is an analog Input/Output card capable of supporting 12 mono inputs and outputs. Inputs 1-8 use the top DB25 connector, while the middle DB25 connector supports Inputs 9-12, and Outputs 1-4. Outputs 5-12 use the bottom DB25 connector. Like the IO-8D, the IO-24A can adjust the gain of individual signals. In addition, the IO-24A card converts all analog inputs to digital for mixing/routing/processing. Any signals routed to its output’s are then converted back to an analog signal. The IO-24A cards are installed in slots B, C, and D.

This is an ornery beast. ->

LOGITEK SHARCATTACK DSP

D S P   C A R D

The SharcAttack DSP card is where all signal processing and mixing happens. It provides EQ, Compression, and Limiting capabilities for up to 32 separate channels. The DS card is installed in Slot J.

QUICK

HIT THE RESET BUTTON!

THE QUICK RESET

This will reboot all cards in the Audio Engine, while retaining all current routes, gain levels and other DSP settings. There will be a momentary audio interruption.

To perform a quick reset:
Press the black Reset button on the AE-C6 card. The LCD screen will display “Quick” in the upper right corner.

You can also perform a Quick Reset from Supervisor or AEConfig (when loading a new config).

THE FULL RESET

This will reboot all cards and reload the last configuration saved to the C6 card. In addition it will reset any routes, gain levels and DSP settings. There will be an audio loss of several seconds. Once the reset has been completed, all of the channels will be turned off, so it is vitally important that you fire the Initial Triggers or turn on a channel. In addition, while a Full Reset is being performed the Numix will freeze and become unresponsive. Once you see the level meters return to 0, it is safe to begin using the board again. Occasionally, the Numix screens will corrupt and not properly display level metering. This can be fixed by power cycling the Numix.

A Full Reset should be performed if you need to load a new config that has routing changes, or if the running config file has become corrupted.

A Full Reset WILL cause an audio interruption of several seconds.

To perform a Full Reset:
Press the red ACCEPT/FULL button on the Power Supply and the black RESET button on the AE-C6. Release the black RESET button first, and once the LCD screen displays “FULL” release the red ACCEPT/FULL button.

You can also initiate a Full Reset from Supervisor or AEConfig (when loading a new config).

THE NO CONFIG RESET

This will force the Audio Engine to reboot all the cards and boot without a loaded config file. This is useful if the config file has been seriously corrupted, or a card has been damaged.

To perform a No Config Reset:
Press and hold the green ESCAPE/IDLE button on the Power Supply and the black RESET button on the C6 card. Release the black RESET button first, and once the LCD screen displays the message “Config Not Loaded” release the green ESCAPE/IDLE button.

THE AUDIO ENGINE

PART 7

SUPERVISOR & COMPANY

- THE BASICS
- SUPERVISOR
- AECONFIG
- vBUTTON
- vFADER
- vDELAY
- COMMAND BUILDER

THE B A S I C S

Logitek Supervisor is the software monitoring and control for the Audio Engine. It allows you to easily view the current status of any of the inputs, outputs, or DSP buses, as well as the status of GPI/O's.

In addition, it also serves as gateway to the vTools software suite. The vTools software suite is a collection of programs that can be used to control various components of the Audio Engine and associated console.

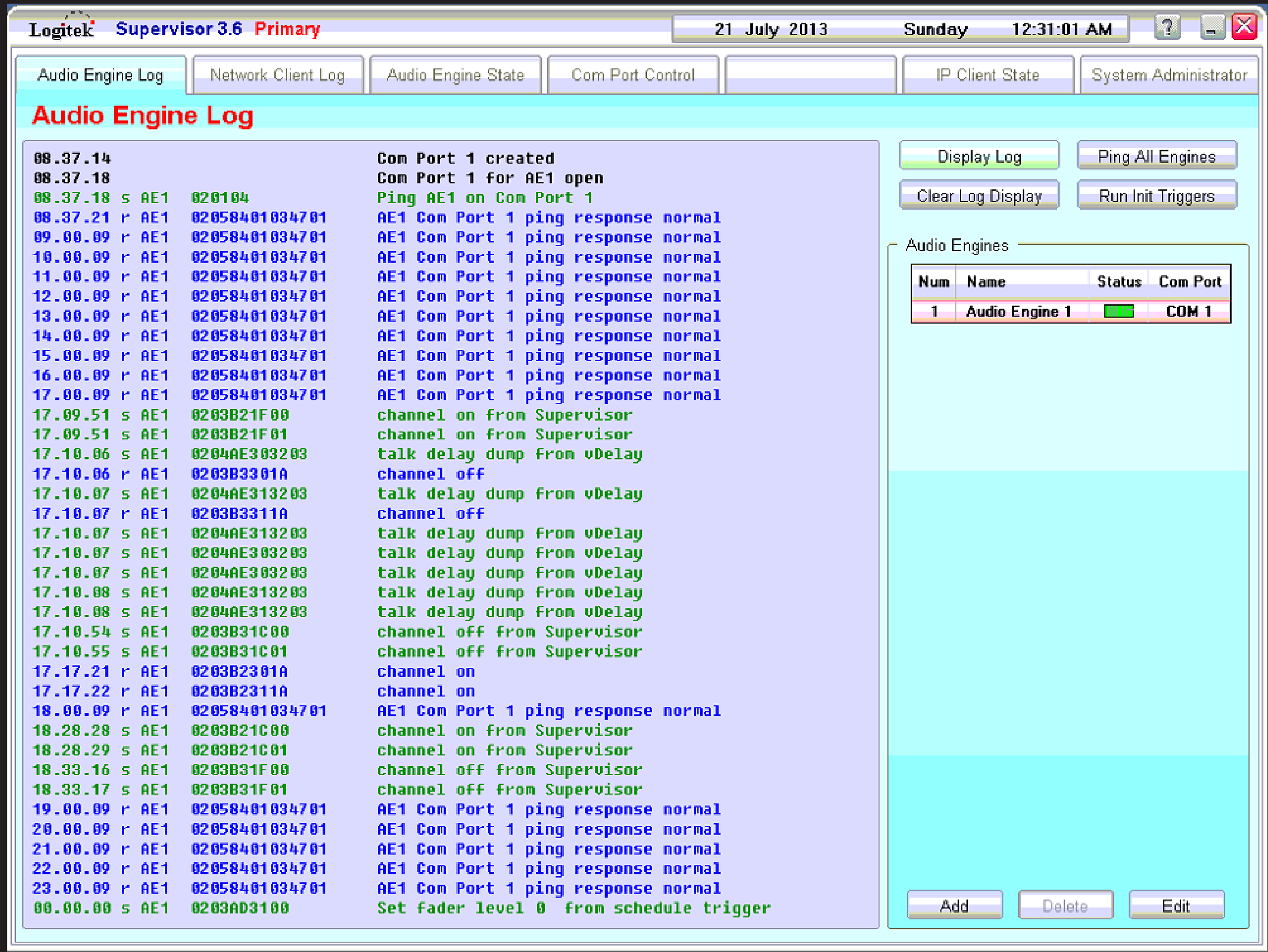
Command Builder is a rudimentary programming software that allows the construction of macro functions which can be executed by Supervisor.

AUDIO ENGINE L O G

When Supervisor is launched it defaults to the Audio Engine Log tab. This page (amusingly enough) displays a running log of everything the Audio Engine does, from faders being turned off, to routes being changed, to bus assignments, etc. etc. It's only really useful for diagnostic purposes (as in, "Oh balls I've broken everything again....").

On the right side of the page are options to clear or display the log, as well the "Ping All Engines" and "Run Init Triggers" buttons. "Triggers" are Logitek terminology for macros, and Initial (Init) Triggers are macros that are set to run everytime the Numix boots up. For example, the current set of Initial Triggers resets bus assignments and fader states, initializes the delay, and applies the proper EQ and compression settings.

Lastly, at the bottom, are options to add, remove, and edit Audio Engines. From time to time, the serial connection between Supervisor and the Audio Engine will become....and break. Simply select the COM Port to which the Audio Engine is connected and click "Assign". Clicking on "Edit" will also allow you to ping the Audio Engine as well as perform Soft and Full Resets.



The Network Client Log is basically the same as the Audio Engine Log, except that it logs commands from any connected clients (primarily the vTool clients). Again, only useful when you're in serious trouble.

HEADS UP!
NAME CHANGE

Up until now "Bus" has meant a collection of audio signals. For example, one would assign a local source (such as a microphone) to a fader so that the signal can be processed and routed. The fader can then be assigned to certain buses, such as the PGM bus. Make sense right?

In Supervisor however, a bus refers to an On/Off toggle. For example, Bus 1 (when referring to a fader) indicates whether the Fader has been assigned to the PGM Bus. Likewise, Bus 2 refers to Aux 1, Bus 3, Aux 2, and so on and so forth.

Buses are also used to indicate whether any processing has been applied to a channel (for example, turning Bus 22 on, will activate the dynamics processing for that source, while Bus 23 will turn on the Equalizer). There are 255 total buses per destination device. The purpose of most of these buses is not documented, so unless you know in advance what a bus does…don't touch it. ☺

AUDIO ENGINE STATE

At the top of the Status page are the PGM and MONITOR meters for the Numix console. Below are three tabs: “Engine State Vector”, “Variable State Vector”, and “Fader Source Devices Allowed”.

The “Engine State Vector” tab is the default page, and the most useful. Here, you can see the status of each of the faders attached to the Numix, the state of the DSP buses, and the sources of the physical outputs.

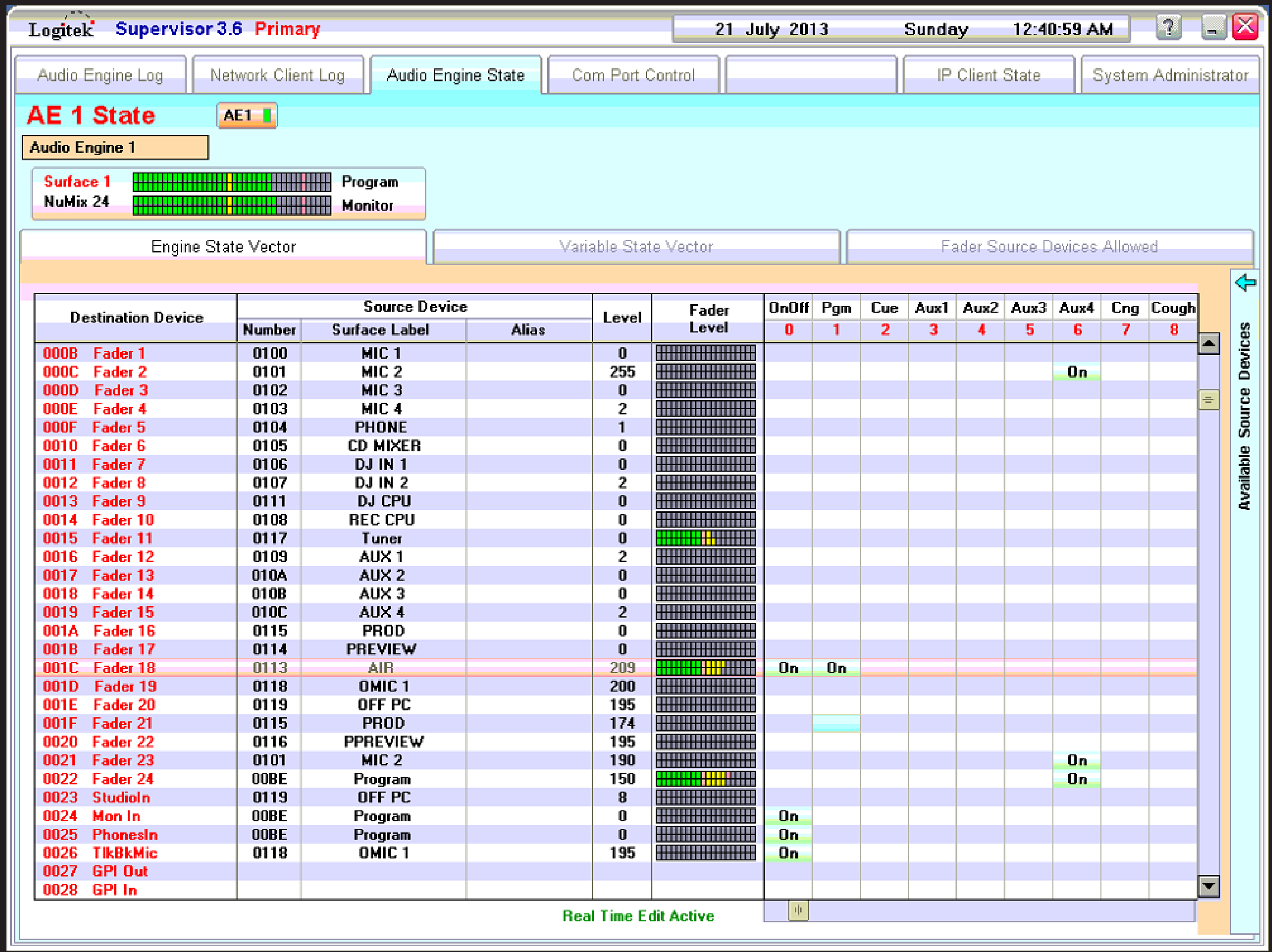
There are four distinct “groups” of destination devices: Audio Engine GPI/O’s (Device 0001 and 0002), Numix Faders 1-24 (Device 000B through Device 0022), Numix DSP Sources (Device 0023 through Device 0032), and Physical Outputs (Device 006E through Device 0082).

Oh, do those device numbers look weird to you? Logitek has a terrible infatuation with hex, so those device numbers are in hex. Yeah. Also in hex are all of the Audio Engine commands. Have fun! ☺

You can access a details page for each of the 24 faders by clicking on the value in the “Level” column. From this page you can adjust every single parameter for the fader as if you were in front of the Numix, including level, bus assignments, and dynamics and equalization processing.

You can also access the details page for the DSP Buses, but you can (or at least, you should only) adjust the level and whether the bus is on or off. Theoretically, you can apply dynamics and equalization proccessing, but you really shouldn’t. You should also never assign any of the DSP buses to any of the major DSP buses (for example, don’t assign Route 1 In to the PGM bus).

The Engine State Vector tab also allows you to change the source device for any of the faders, DSP buses, or Physical Outputs. To do this, simply hover over the arrow on the far right of the screen and the “Available Source Devices” tab will pop up out. Simply select the source device you wish to assign to the destination device and click on it.



The Variable State Vector tab keeps track of the values of both user and system defined variables. User Defined variables are created in Command Builder for specific trigger files. For the most part the Vector State tab is only used for diagnostic purposes.

The Fader Source Devices Allowed tab allows you to define what sources may be assigned to which faders. It’s the equivalent of the Surface Settings tab in AEConfig.

SYSTEM ADMINISTRATOR

THE SYSTEM SETUP PAGE

The System Setup page is used for configuring Supervisor to work with the Audio Engine. There are many settings here, but the important ones are listed below:

- Supervisor Process Priority: This should be set to “High”
- Base TCP/IP Port: This is the port that vTools programs will use to communicate with Supervisor. The default is 10200.
- Audio Engine Controller: “C6 Card” should always be checked.

THE SOFTWARE LICENSE PAGE

While the majority of the vTools suite is freely available, Logitek also makes three other programs for purchase; vMix, MatrixIP, and vScreen. Should you purchase a license for one of these, you would enter the license key here.

THE SPECIAL FUNCTIONS PAGE

This page is used to set a few miscellaneous options, most of which are self-explanatory. In particular, make sure that “Allow Real Time Editor” is checked, or you will not be able to make changes in the “Engine State Vector” tab.

THE USER SETUP PAGE

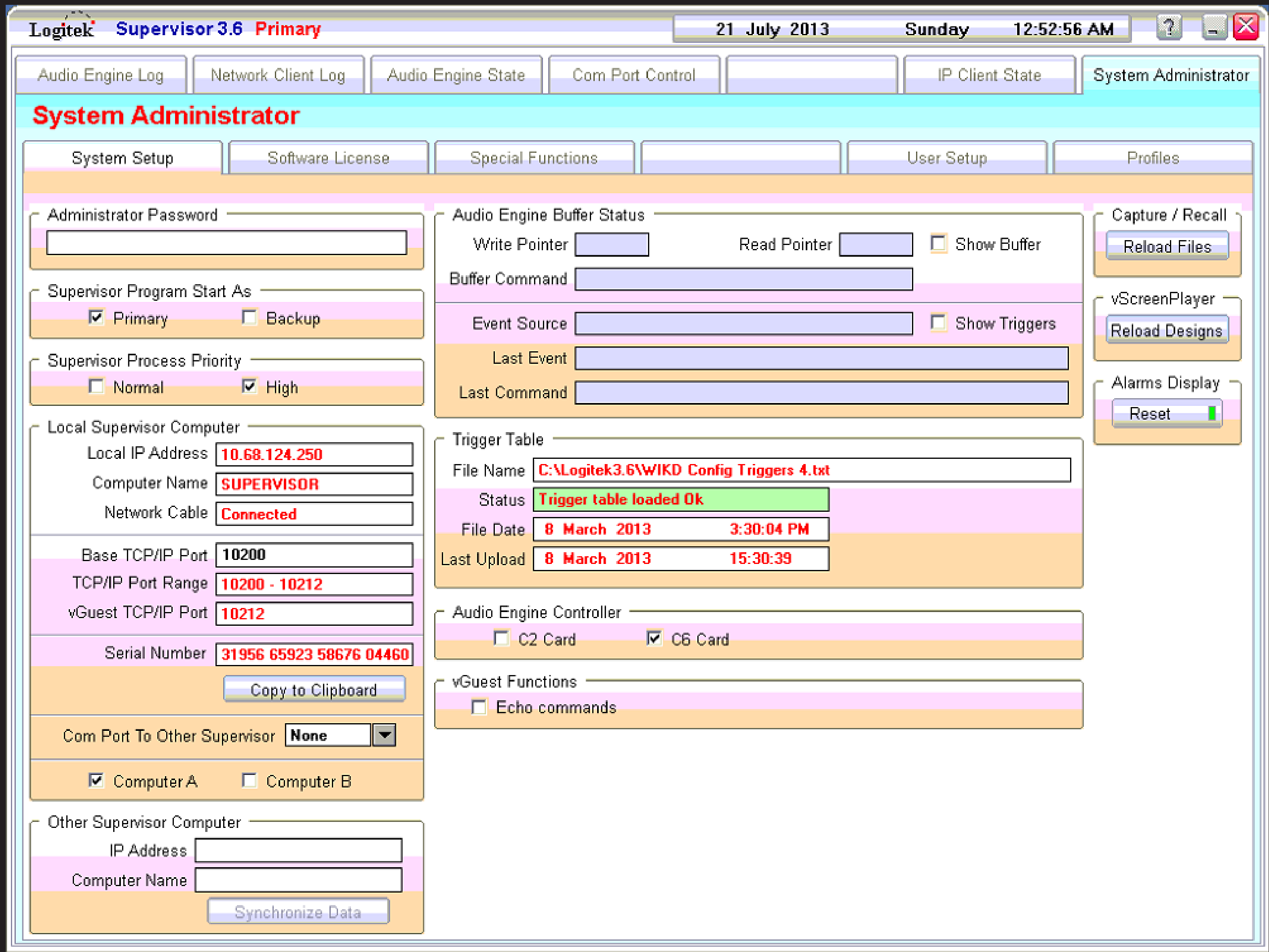
The User Setup page is used to define the accounts that can connect to Supervisor, and the programs to which they have access, and the amount of connections they may create. For example, the “admin” account shown below, can be used to connect AEConfig, Command Builder, vRoute, vFader, vGuest, and vDelay. The number indicates the specific profile to be loaded upon login.

You should not make any changes to the User Setup page until you have thoroughly read through the Supervisor and vTools manuals.

THE PROFILES PAGE

The profiles page is used to set up specific configurations for vTools applications. For example, the vButton profile page is used to define the different vButton layouts available, while the vDelay profile page is used to define which Routes will be monitored.

Until you have read the through the Command Builder, Supervisor, and vTools manuals (thoroughly dammnit!) you should not touch anything on this page.



Seriously! Read the damn manual!

A E C O N F I G

The easiest way to learn how to use AEConfig is to simply start playing with it. As such this part will guide you through the process of recreating the AE32 configuration file for WIKD. It will not cover every possible option, but luckily Logitek wrote a pretty comprehensive manual.

The basic steps to create the configuration file are as follows:

1. Define Engine Configuration and Control Surface (Hardware Configuration)
2. Define Audio Inputs (Input Settings)
3. Define Audio Output (Output Settings)
4. Configure the Control Surface (Surface Settings)
5. Set System Parameters (System Page)

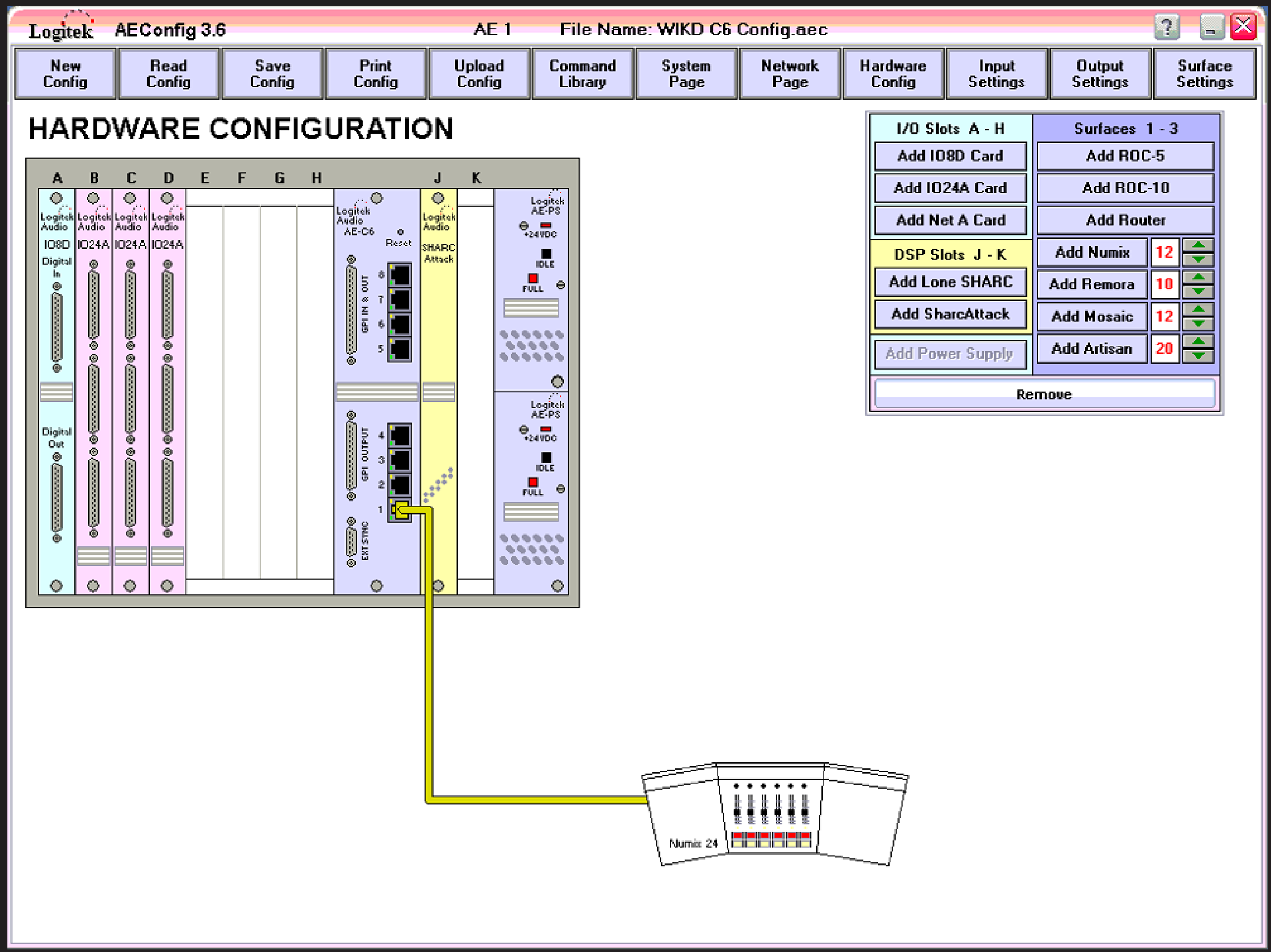
If you'd like a more comprehensive tutorial, try using this. It was written for the original version of AEConfig, but most everything is still relevant.

HARDWARE CONFIGURATION

When creating a new AEConfig file, this is where you will start. Here you can define what is installed in the Audio Engine—the number of I/O cards, the type and number of DSP cards, what version controller card, and whether redundant power supplies. Lastly, you can also define the number of surfaces connected, and where they are connected. The Hardware Configuration must match the physical configuration EXACTLY.
WIKD's Audio Engine is configured as follows:

- SLOT A: IO8D
- SLOT B: IO24A
- SLOT C: IO24A
- SLOT D: IO24A
- SLOT E: BLANK
- SLOT F: BLANK
- SLOT G: BLANK
- SLOT H: BLANK
- SLOT J: SharcAttack DSP
- SLOT K: BLANK
- 2x Power Supplies
- 24 Fader Numix (Port 1)

WAIT. WHAT? WIKD only has an 18 Fader Numix! That's correct, however faders do not necessarily have to physically exist. The final six faders only exist in the virtual world, and can only be controlled by software.



AECONFIG
INPUT SETTINGS

The Audio Engine has no way of magically discovering what you have plugged in, so it is up to you to define all of your inputs. This includes basic things, like the name, and where the source is connected, but also control what features the source will have access to. For example, you may wish to allow a mono source, such as a microphone or phone channel to be panned, but it doesn't make much sense to allow Automation to be panned.

The important settings are as follows:

INPUT INFORMATION & PIN SETTINGS

Here you can assign both a Unique Name as well as a Surface Label. Surface Labels represent the name that will be displayed on the Numix surface and are limited to 8 characters. In addition, the number of channels associated with the source can be defined. For the most part, channels will either be mono (1 channel) or stereo (2 channels), but the Audio Engine does support channel densities up to 5.1 (6 channels).

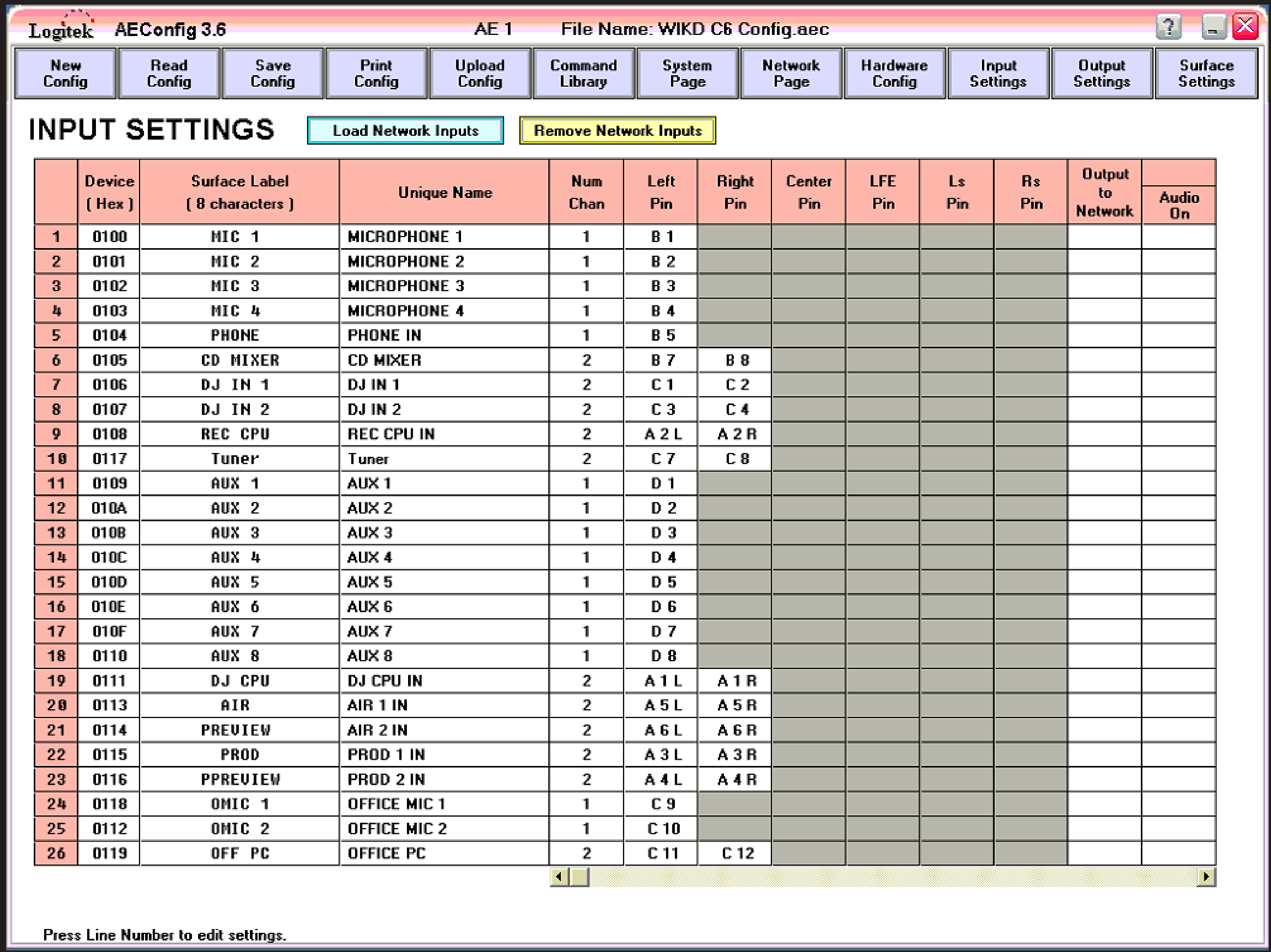
In addition, you will also need to define which pins the source is connected to.

NORMAL SIGNAL LEVEL

This field allows you to adjust the gain applied to the input BEFORE the A/D conversion. In general, most professional equipment should be set to +4, and consumer equipment to -8. Some equipment, such as the microphone preamps, have a nominal signal output level of 0dBu, and as such do not need any gain adjustments. Likewise, digital inputs do not need any adjustments.

MIX MINUS BUS

Here you can assign the input to a specific Mix Minus Bus. Mix Minuses in the Audio Engine consist of tow parts: the source bus, and the minus bus. The Audio Engine allows the creation of up to 25 separate mono Mix Minus buses. There are five possible source buses: PGM, CUE, AUX1, AUX2, and AUX3. For a specific Mix Minus bus, you must first specify the source bus (this is done on the SYSTEM page). From there, you can assign an input to a specific Mix Minus bus from the Input page. Inputs can only be assigned to one Mix Minus bus.



	Device (Hex)	Surface Label (8 characters)	Unique Name	Num Chan	Left Pin	Right Pin	Center Pin	LFE Pin	Ls Pin	Rs Pin	Output to Network	Audio On
1	0100	MIC 1	MICROPHONE 1	1	B 1							
2	0101	MIC 2	MICROPHONE 2	1	B 2							
3	0102	MIC 3	MICROPHONE 3	1	B 3							
4	0103	MIC 4	MICROPHONE 4	1	B 4							
5	0104	PHONE	PHONE IN	1	B 5							
6	0105	CD MIXER	CD MIXER	2	B 7	B 8						
7	0106	DJ IN 1	DJ IN 1	2	C 1	C 2						
8	0107	DJ IN 2	DJ IN 2	2	C 3	C 4						
9	0108	REC CPU	REC CPU IN	2	A 2 L	A 2 R						
10	0117	Tuner	Tuner	2	C 7	C 8						
11	0109	AUX 1	AUX 1	1	D 1							
12	010A	AUX 2	AUX 2	1	D 2							
13	010B	AUX 3	AUX 3	1	D 3							
14	010C	AUX 4	AUX 4	1	D 4							
15	010D	AUX 5	AUX 5	1	D 5							
16	010E	AUX 6	AUX 6	1	D 6							
17	010F	AUX 7	AUX 7	1	D 7							
18	0110	AUX 8	AUX 8	1	D 8							
19	0111	DJ CPU	DJ CPU IN	2	A 1 L	A 1 R						
20	0113	AIR	AIR 1 IN	2	A 5 L	A 5 R						
21	0114	PREVIEW	AIR 2 IN	2	A 6 L	A 6 R						
22	0115	PROD	PROD 1 IN	2	A 3 L	A 3 R						
23	0116	PPREVIEW	PROD 2 IN	2	A 4 L	A 4 R						
24	0118	OMIC 1	OFFICE MIC 1	1	C 9							
25	0112	OMIC 2	OFFICE MIC 2	1	C 10							
26	0119	OFF PC	OFFICE PC	2	C 11	C 12						

Press Line Number to edit settings.

AECONFIG
INPUT SETTINGS

FEATURES

- FADER START: If this is enabled, moving the fader up from -∞ will turn the channel on.
- CUE BUS: If this is set to Pre Fader, then the input source will be assigned to the Cue Bus irrespective of the fader level. If it is set to Post Fader, then the input source will reflect the fader position when assigned to the Cue Bus.
- CUE @ INF.: If this is enabled, setting the fader to -∞ will automatically assign the input to the Cue Bus.
- ALLOW MODE: If this is enabled you will be allowed to adjust the polarity of the channel.
- ALLOW PAN: This determines whether an input can be panned. This should only be enabled on mono channels.
- ALLOW EFFECTS: If this is enabled, the input will have access to EQ & Dynamics control.
- ALLOW TRIM: Allows the digital gain of the input to be adjusted. Remember, this is applied after the A/D conversion.
- OPTION 2: Does nothing.
- TIMER DISABLE: If set to “No”, then a timer will appear above the fader on the Numix surface. The timer will indicate how long the channel has been on.

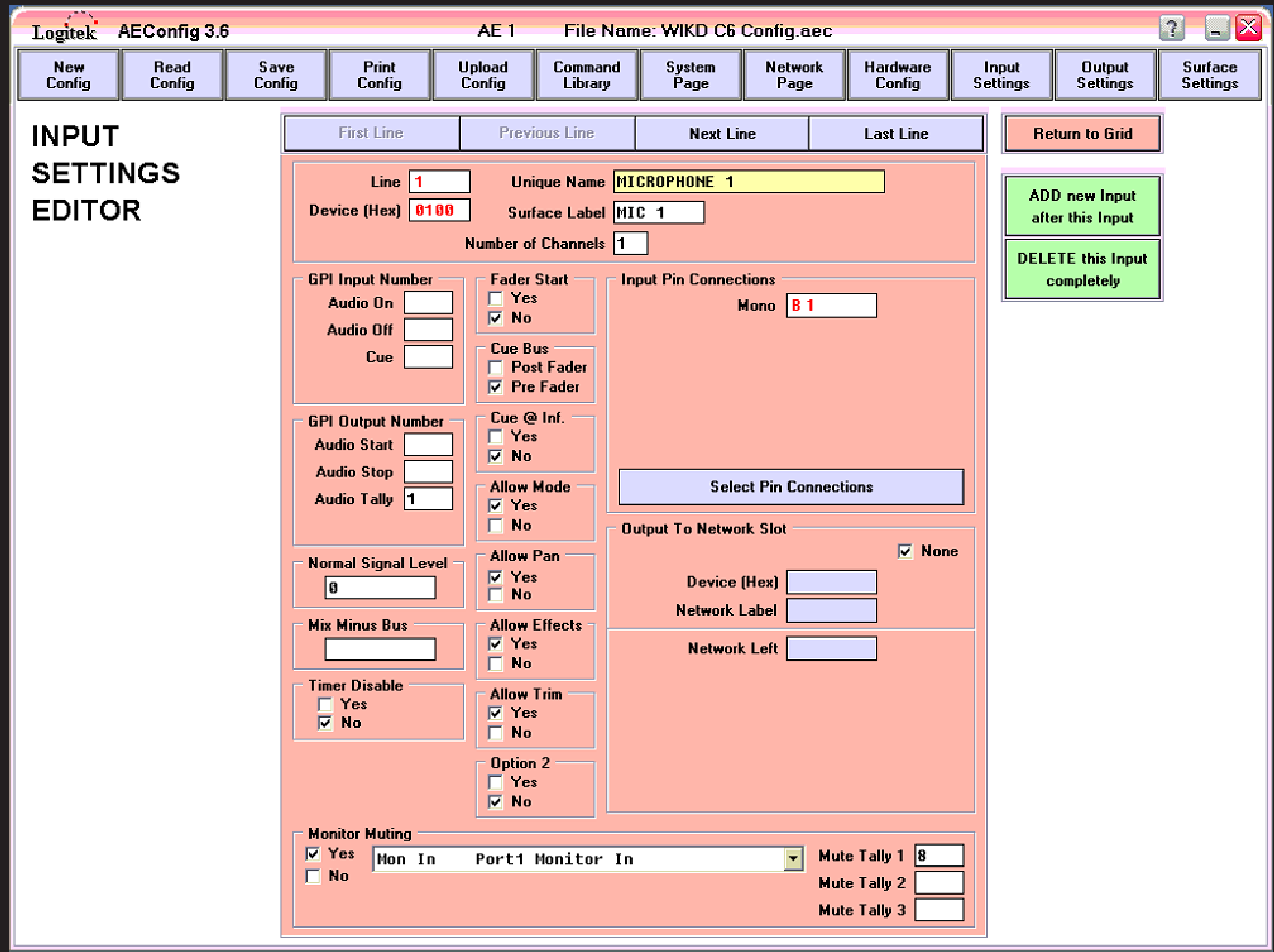
MONITOR MUTING & MUTE TALLY’S

The Monitor Muting section allows you to mute a specific output when the input is turned on. You can choose from the MONITOR, STUDIO, or PHONES output. For example, the studio microphones in WIKD all have Monitor Muting enabled for the MONITOR output. This means that as soon as a microphone is turned on, the MONITOR output is muted.

In addition, you can also set three separate Mute Tally’s. These will trigger a relay closure for as long as the channel is on.

GPI/O

Inputs within the Audio Engine can be turned on or off, as well as assigned to the Cue Bus, by GPI closures. In addition, the Audio Engine can also trigger relay closures if a channel is turned on or off. The “Audio Tally” option will keep a relay closed for as long as a channel is on. The Audio Engine has a total of 15 GPI’s and 15 GPO’s. In addition, the Numix also has 12 GPI’s and 12 GPO’s. Audio Engine GPI/O’s use numbers 1-15, while GPI/O’s on the Numix use numbers 101-112.



The Output Settings page is incredibly simple to use, which is why it doesn’t get it’s own page. Simply name the output, define the number of channels, and assign it to the appropriate physical pins. You can also change the output level, and the digital output format.

AECONFIG
SURFACE SETTINGS

The Surface Settings page is somewhat complicated. Within the Surface Settings page are four tabs: Port 1, Port 2, Port 3, and Output settings. The Audio Engine allows for up to three separate surfaces to be connected, on Ports 1, 2, and 3. Each Port contains a number of DSP Buses; for example, the PGM Bus, the MONITOR Buses, the AUX Buses, etc. etc.

In order to use a Surface on Port 2 or 3, you must have a second DSP card. In addition, Surfaces on Ports 2 & 3 do not have as many DSP Buses, or as much processing power.

CONFIGURING THE PORT 1 SURFACE

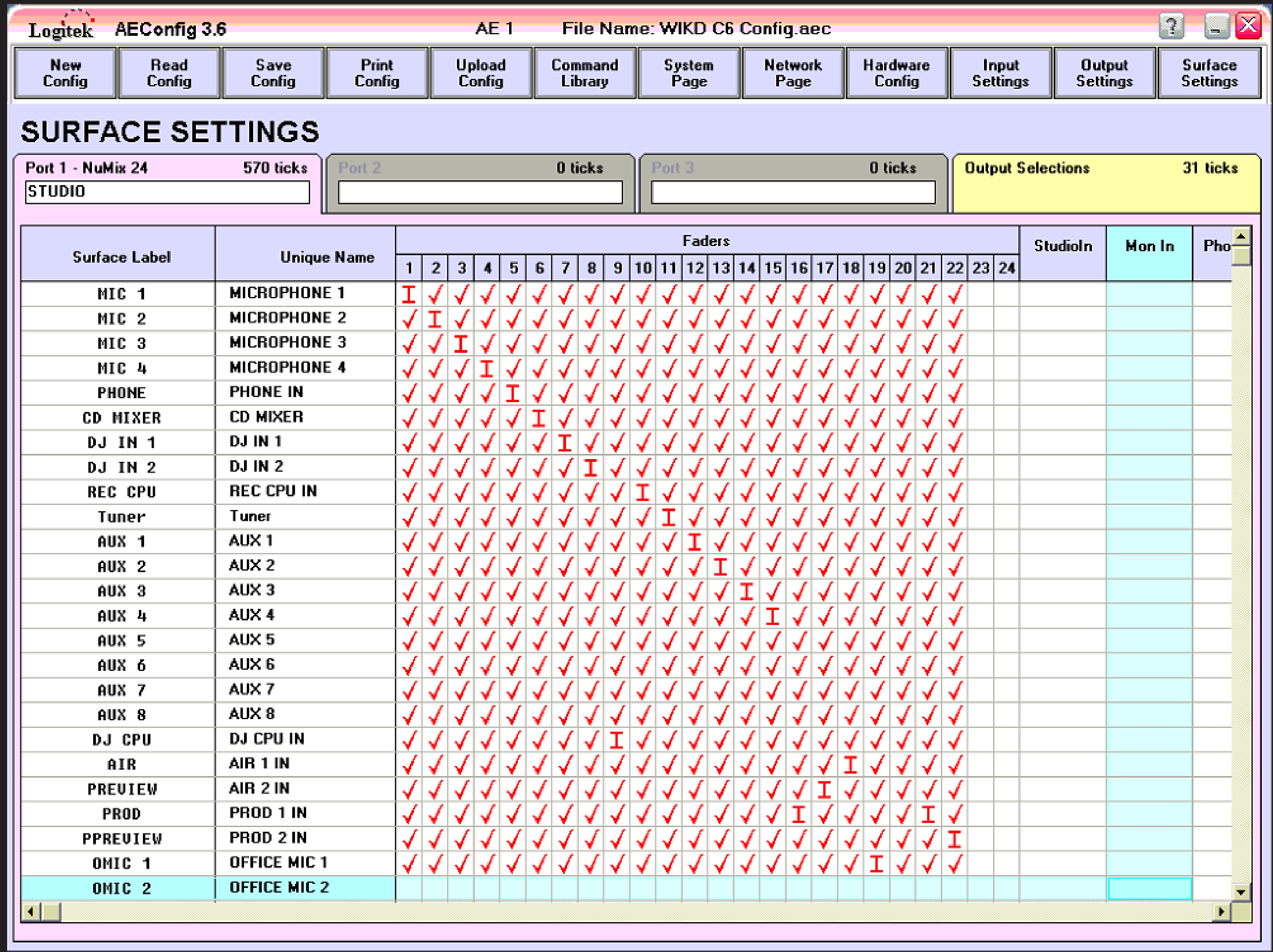
The Surface Settings page for Port 1 allows you to define what physical inputs will be accessible from which faders. Faders are listed on the top, while physical inputs are listed on the right. To declare an input accessible from a specific fader, simply click the intersection between the source and the fader. A check mark indicates the source can be accessed from that fader, while an “I” indicates that it is the default source.

After the faders and physical inputs, come the DSP buses. DSP Bus inputs are listed on the top, while DSP Bus Outputs are listed on the left.
Physical Inputs are red, while DSP Buses are green.

WIKD only uses a few basic default routes, listed below:

STUDIOIN, MON IN, PHONEIN:
- Port 1 Program Out (Default Source)
- Port 1 Aux 1
- Port 1 Aux 2
- Port 1 Aux 3
- Port 1 Router 1 Out

ROUTE1IN, ROUTE2IN:
- Port 1 Program Out (Default Source)



OUTPUT SELECTIONS

This page allows you to map both physical inputs and DSP Buses to physical outputs. The physical outputs are listed at the top, while physical inputs and DSP Buses are listed to the right.
As before, physical inputs are red, while DSP Buses are green.
For a full list of I/O mappings, refer to the WIKD Signals.xlsx file.

AECONFIG
SYSTEM PAGE

The System Page allows you to make changes to the core configuration of the Audio Engine. For the most part, it should only ever be touched when you are creating an initial configuration.

The most critical settings are the Sample Rate and Controller Card settings. The Controller Card must be set to “C6” as this is the version WIKD uses. The Sample Rate should be set to 44.1 KHz, as the Audio Engine will serve as the clock source for all digital inputs and outputs.

Technically, you could set the Sample Rate to 48 KHz (or even 32 KHz if you’re an ass...). The Audio Engine is capable of performing Sample Rate Conversions, and the sound cards in the Automation and DJ computers can be set to any desired sample rate. However, all of the songs WIKD receives from TM Studios use a sample rate of 44.1 KHz. While Sample Rate Conversions would likely not be noticeable to the average listener, it’s good practice to keep the signal chain as pristine as possible.

The System Page also allows you to configure Mix Minus Buses. For each of the 25 buses you can select the source bus (PGM, CUE, AUX 1, AUX 2, AUX 3). The “Bus Always On” option means that the Source Bus will always be sent regardless of whether the Minus Bus is On or Off. The “Add Mic When Off” option will send the Talkback Microphone to the physical output when the Minus Bus is turned off.

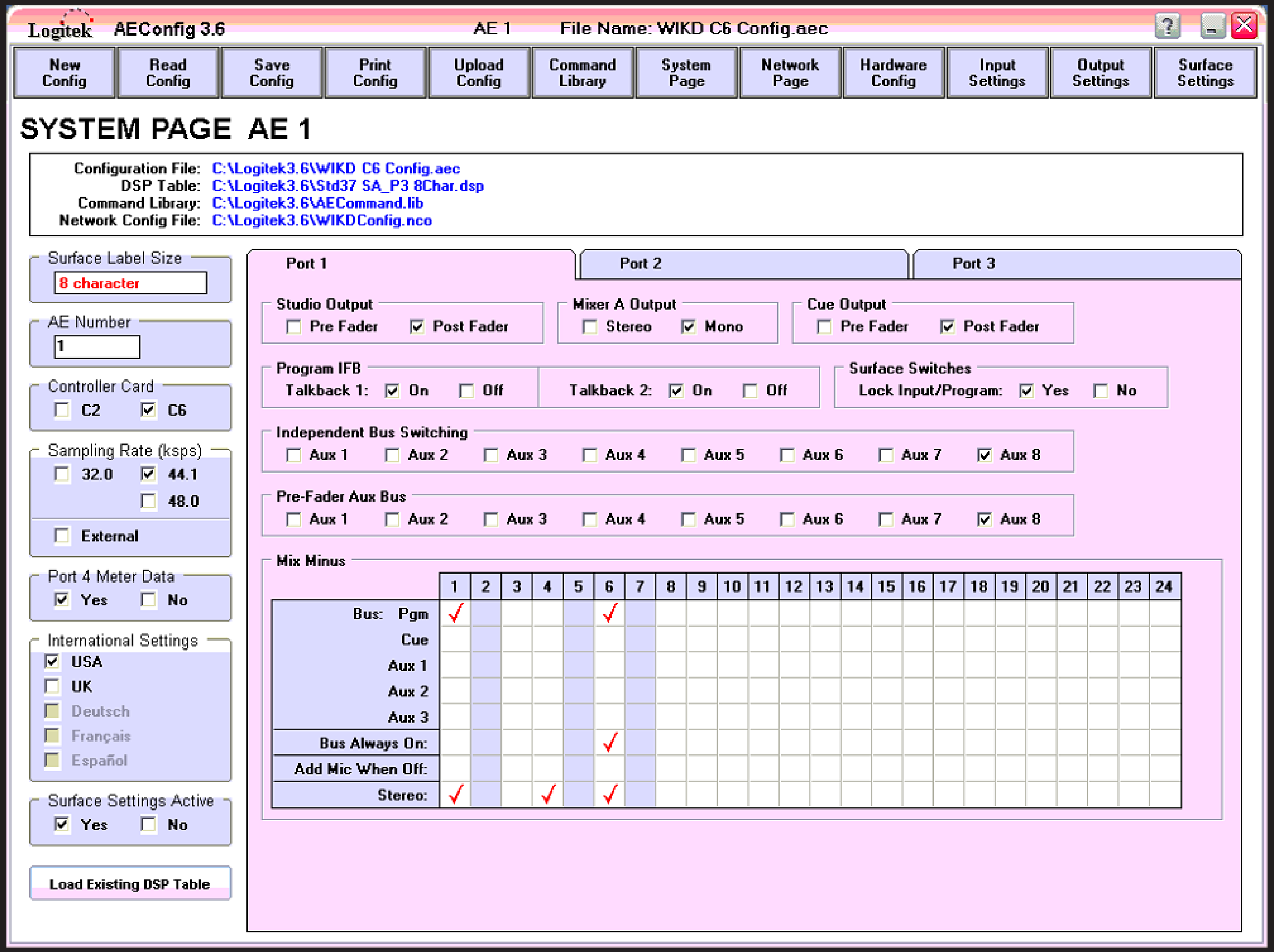
There are several other options available on the Systems page. The “Lock Input/Program” switch will lock a faders PGM bus assignment if checked. In other words, if a fader is turned ON, the PGM bus assignment cannot be changed.

“Independent Bus Switching” means that the AUX will be fed pre On/Off switch, while checking the “Pre-Fader Aux Bus” will allow the selected AUX to be fed Pre-Fader. (Imagine that!)

The Studio and Cue buses can be set to either Pre or Post Fader, and the Mixer A Output can be set to either Mono or Stereo.

The DSP Table is essentially a master table of all the different buses and processing available for each of the different consoles, as well as the hex addresses for individual buses. In order to create and upload a configuration you must supply a DSP Table. The correct DSP table to use is entitled:
“Std37 SA_P3 8Char.dsp”

If you cannot find this file (it is located within the Logitek directory, as well as on Google Drive), you can contact Logitek Support.



AECONFIG
CONFIG UPLOADS

There are two ways to upload a configuration file to the Audio Engine. The first is the direct method, and requires that the computer running AEConfig is connected to the Audio Engine's control port. The second method is to upload the configuration file through Supervisor, the Audio Engine's monitoring software. In order to connect via Supervisor, you will have to supply the IP address of the machine running Supervisor, as well as the Server Port, Computer Name, and User Name and Password.

If Supervisor is running, you MUST upload the configuration file through Supervisor.

Once you have selected a method, the next steps are the same. Select the Audio Engine and then Check, Upload and Store the configuration file. You can then choose to perform a Soft or Full Reset.

SOFT RESET

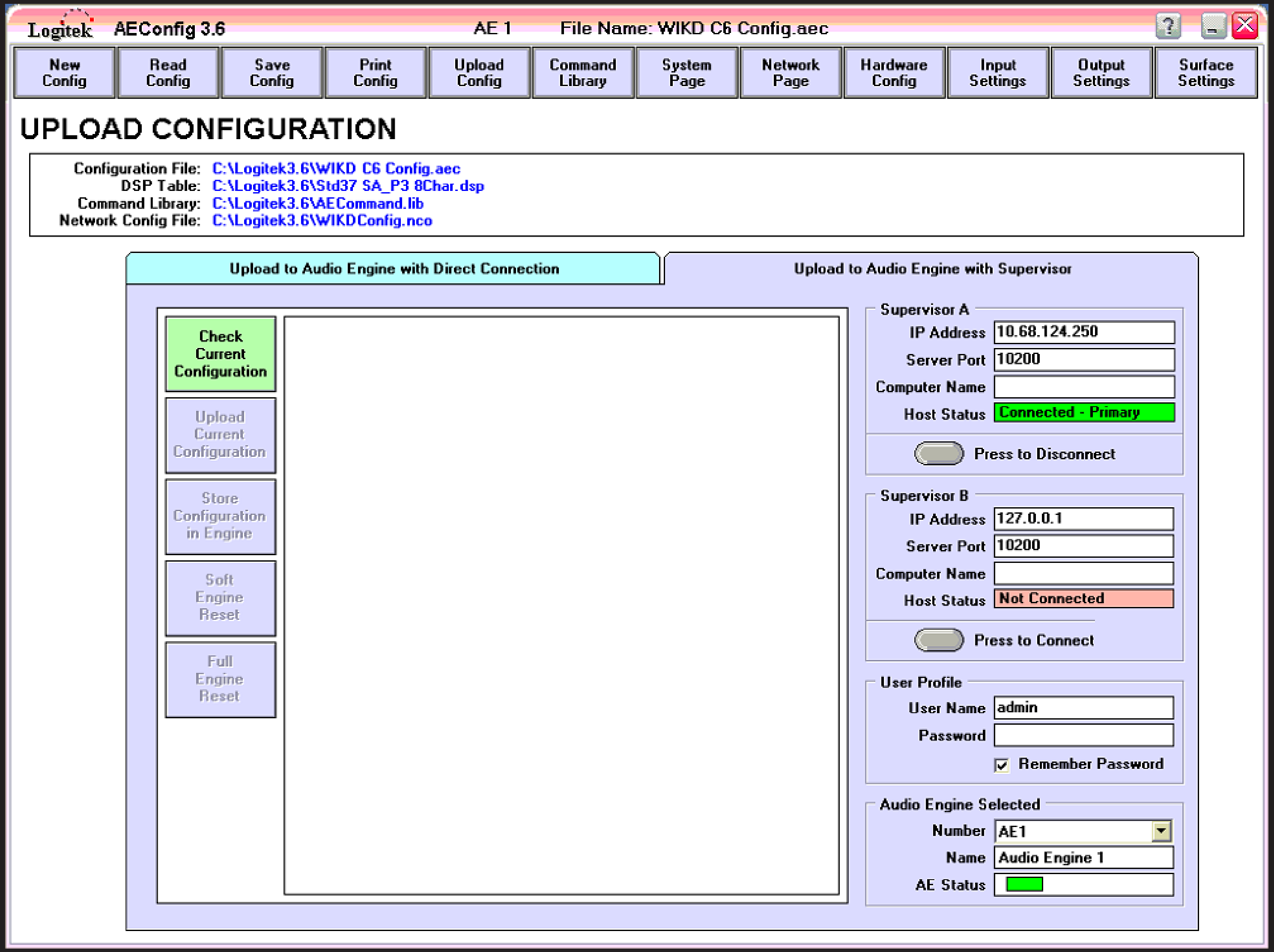
A Soft Reset will load all of the settings EXCEPT for Default Routes and new Outputs. If you have not changed either of those, then a Soft Reset is sufficient. The advantage of a Soft Reset is that it can be done while the board is live On-Air.

FULL RESET

A Full Reset will reload ALL settings. It will restore all Default Routes, Bus Assignments, and Fader Levels.

A FULL RESET WIL INTERRUPT AUDIO FOR SEVERAL SECONDS.

It is rare that you will need to perform a Full Reset.



v B U T T O N

Technically, vButton is used to emulate the Button panels present on the Numix—either Bridge Buttons 1-12, or Softkeys 1-12. However, as it turns out, vButtons can actually exist by themselves, which unlocks a huge amount of power.

To operate vButton you simply need to download the vButton executable from Logitek's website, or from the Engineering NAS. Open the executable and click the red “On Line 1” button in the top right corner.

There are three different profiles that can be accessed in vButton, each with varying levels of control.

DJ PROFILE

This profile allows you to control the source and volume of the Office Speaker, as well as the input source to the Office PC.

USERNAME: DJ
PASSWORD:

OPERATIONS PROFILE

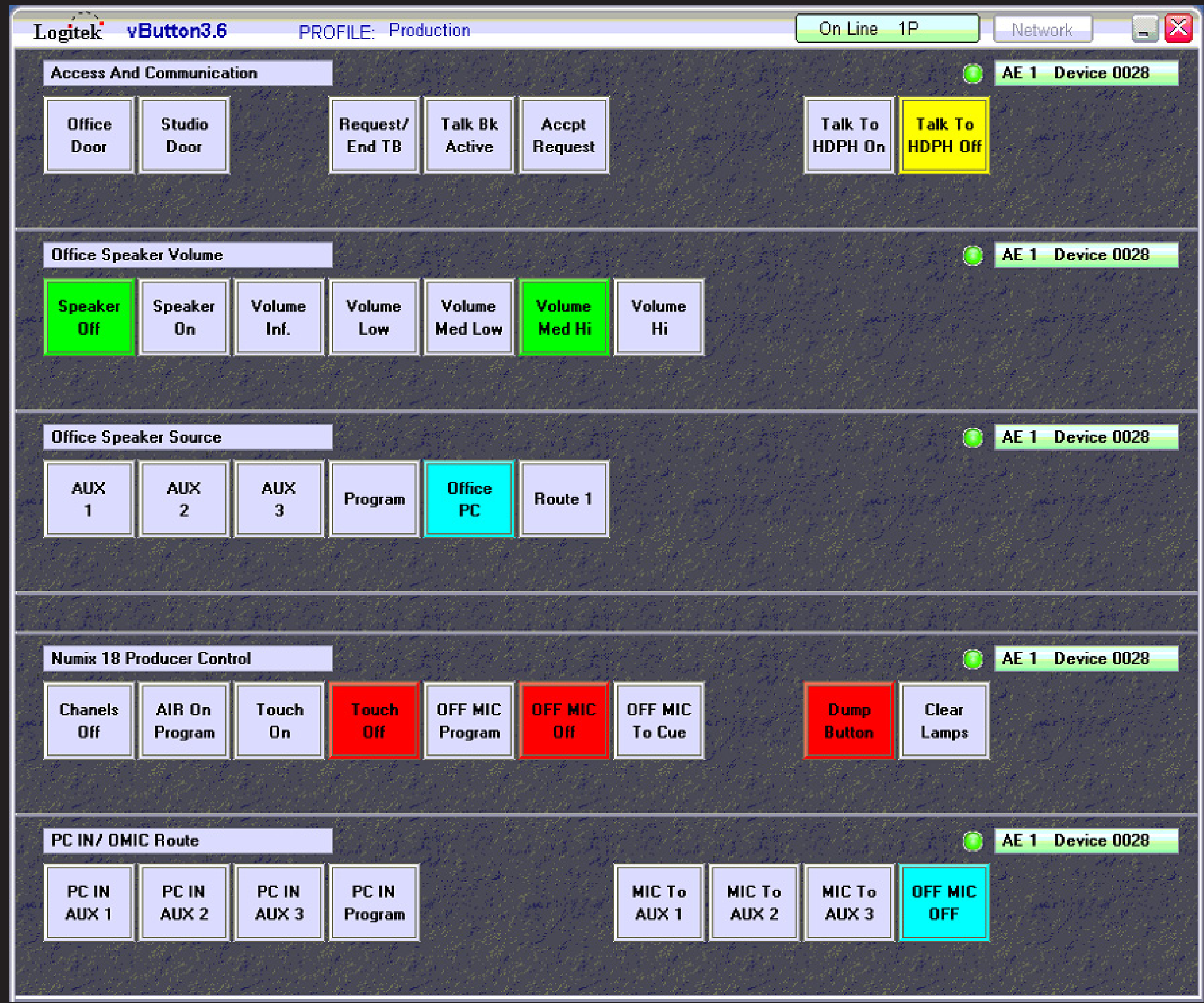
This profile allows you to turn control of the Office Touch Screen On or Off, turn the Office Microphone Off, or assign it to the Cue speaker of the Numix, as well as route the Office Microphone to any of the AUX buses. In addition, it can do everything the DJ profile can.

USERNAME: OPS
PASSWORD: 2012

PRODUCER PROFILE

The Producer profile is intended to allow somebody to produce an entire show, and as such has “advanced” talkback capabilities. It can initiate a conversation between the Office and Studio, route the Office Microphone directly to the Studio Headphones, as well as initiate a Dump. In addition, it can assign the Office Microphone to the PGM bus, and turn all of the channels on the Numix off. Lastly, it can also unlock the Office and Studio doors, which is the only reason anybody uses the damn program.

USERNAME: Producer
PASSWORD: 2012a



A conversation between the studio and office can be initiated by either party. In order to initiate a conversation from the Office, press the “Talk Bk Active” button. This will pop up a message on the soft screen of the Numix—if the studio chooses to accept your call, they can press Softkey Button 3. This will route the Office Microphone to the Studio Monitors, and Microphone 2 from the Studio to the Office Speakers. To end the conversation press “Request End TB”, or press Softkey Button 6.

To initiate a conversation from the Studio, press Softkey Button 6, and press the “Accept Request” button in vButton.

The “Talk to HDPH On” button will route the Office Microphone to the Studio Headphones in addition to the PGM bus—this allows somebody in the Office to communicate directly with the DJ's, without going On-Air.

The “OFF MIC To Cue” button will assign the Office Mic to the Cue Speaker of the Numix. This is useful for quick communication. Do NOT do this if somebody is On-Air.

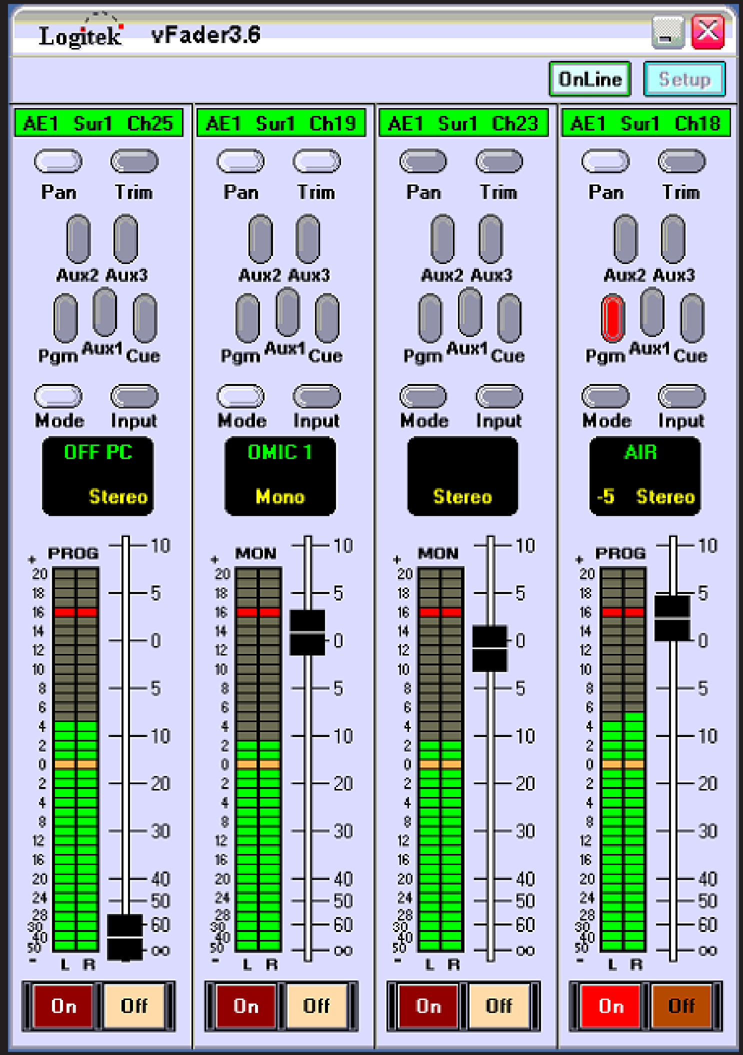
vFADER

vFader allows you to control up to four faders at a time. To operate vFader you simply need to download the vFader executable from Logitek's website, or from the Engineering NAS. Open the executable and click the red "Setup" button in the top right corner. vButton can be accessed using the "admin" account.

Once you have logged into vFader, you can choose the Faders you wish to control in the Setup menu.

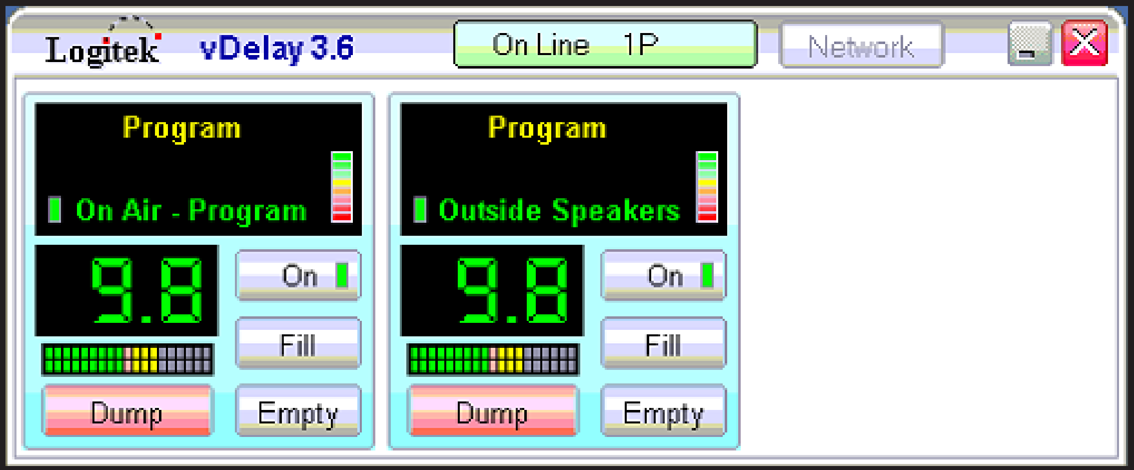
In order to actually control the faders, you must click the red "OnLine" button in the top right. This will log vFader into Supervisor and start displaying real-time updates to the status of the faders. From here, you can adjust the level, pan, and trim of a Fader, assign it to the PGM, AUX, or CUE buses, and change the Mode of the fader (Polarity switch, Left only, Right only, etc.).
In addition, you can also change the source of the fader by pressing the "Input" button.

vFader has been retired by Logitek and is no longer actively supported. For the most part, there's no real reason to use it.



vDELAY

vDelay provides a real time update of the status of any active Delays. WIKD uses two delays: one for the FM and Online streams, and one for the outside streams. vDelay will display the current amount of delay time available, and provides buttons to activate, fill, and empty the delay, as well as a Dump button for each of the two delays. In general, you should not leave vDelay open, as it can corrupt the display timer on the Numix.



C O M M A N D B U I L D E R

CommandBuilder is used to create triggers (macros) that can be executed by Supervisor. These triggers range from the very simple (starting a recording), to the complex (an intercom system).

To fully understand how CommandBuilder works, you should read the manual, which does an excellent job of explaining how to create trigger files and procedures (procedures are a collection of triggers). This guide will instead list out the triggers that currently exist, and what they do.

As much as possible, we have tried to leave documentation within the trigger files themselves.

INITIAL TRIGGERS (TRIGGER #1)

The initial triggers fire every time Supervisor is launched, or whenever the “Execute Init Triggers” button is pressed in Supervisor. In order, it executes the following actions:

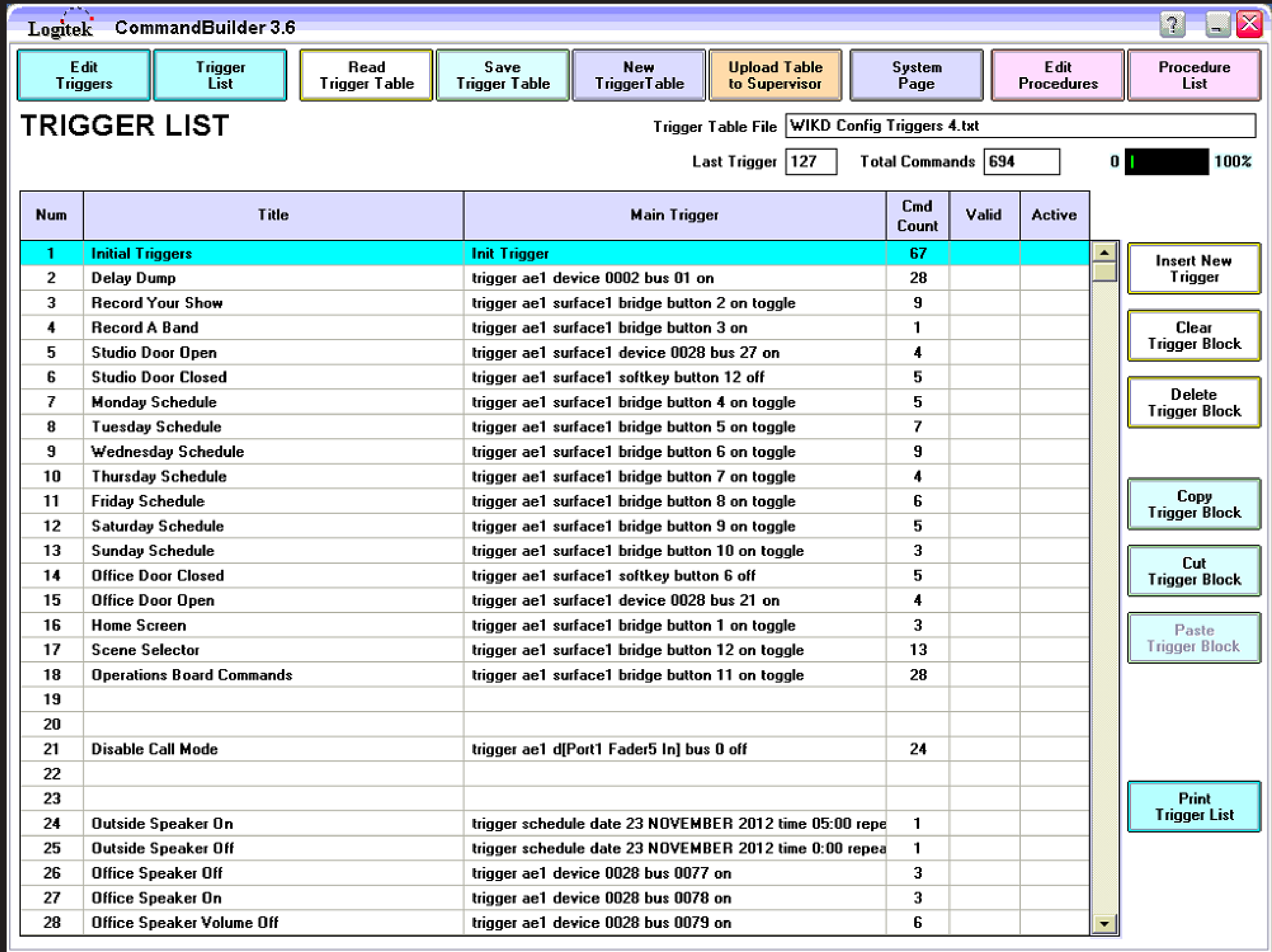
- •Turns Faders 1-24 off, and un-assigns them from any buses. Fader 18 (AIR) is left alone.
- •Initializes the Delay
- •Writes the default text to the fader wedge screens and softkey screen
- •Sets Fader Trims
- •Sets Fader Dynamics Processing
- •Sets Fader Equalization
- •Resets lamp statuses on the Numix and vButton control panels
- •Disables the Office Touchscreen

DELAY DUMP (TRIGGER #2)

This trigger fires anytime the dump button is pressed. It can also be triggered by turning Bus 01 of Device 0002 On. It will dump both delays and display a message on the softkey screen, as well as flashing the bridge lamps for 10 seconds.

RECORD A SHOW (TRIGGER #3)

This trigger fires anytime Bridge Button 2 is pressed. It will send an IP Message to REC CPU that launches Audacity, and then routes the PGM Bus to the digital REC CPU input. It will then flash the lamp of Bridge Button 2 until it is pressed again, at which point it will route the AUX 1 Bus to the digital REC CPU input.



Num	Title	Main Trigger	Cmd Count	Valid	Active
1	Initial Triggers	Init Trigger	67		
2	Delay Dump	trigger ae1 device 0002 bus 01 on	28		
3	Record Your Show	trigger ae1 surface1 bridge button 2 on toggle	9		
4	Record A Band	trigger ae1 surface1 bridge button 3 on	1		
5	Studio Door Open	trigger ae1 surface1 device 0028 bus 27 on	4		
6	Studio Door Closed	trigger ae1 surface1 softkey button 12 off	5		
7	Monday Schedule	trigger ae1 surface1 bridge button 4 on toggle	5		
8	Tuesday Schedule	trigger ae1 surface1 bridge button 5 on toggle	7		
9	Wednesday Schedule	trigger ae1 surface1 bridge button 6 on toggle	9		
10	Thursday Schedule	trigger ae1 surface1 bridge button 7 on toggle	4		
11	Friday Schedule	trigger ae1 surface1 bridge button 8 on toggle	6		
12	Saturday Schedule	trigger ae1 surface1 bridge button 9 on toggle	5		
13	Sunday Schedule	trigger ae1 surface1 bridge button 10 on toggle	3		
14	Office Door Closed	trigger ae1 surface1 softkey button 6 off	5		
15	Office Door Open	trigger ae1 surface1 device 0028 bus 21 on	4		
16	Home Screen	trigger ae1 surface1 bridge button 1 on toggle	3		
17	Scene Selector	trigger ae1 surface1 bridge button 12 on toggle	13		
18	Operations Board Commands	trigger ae1 surface1 bridge button 11 on toggle	28		
19					
20					
21	Disable Call Mode	trigger ae1 d[Port1 Fader5 In] bus 0 off	24		
22					
23					
24	Outside Speaker On	trigger schedule date 23 NOVEMBER 2012 time 05:00 repe	1		
25	Outside Speaker Off	trigger schedule date 23 NOVEMBER 2012 time 0:00 repea	1		
26	Office Speaker Off	trigger ae1 device 0028 bus 0077 on	3		
27	Office Speaker On	trigger ae1 device 0028 bus 0078 on	3		
28	Office Speaker Volume Off	trigger ae1 device 0028 bus 0079 on	6		

COMMANDBUILDER
TRIGGER LIST

RECORD A BAND (TRIGGER #4)
This trigger fires anytime Bridge Button 3 is pressed. It sends an IP Message to REC CPU that launches Pro Tools.

DAILY SCHEDULE (TRIGGER #7-13)
These triggers will display the daily shwo schedule on the softkey screen of the Numix.

HOME SCREEN (TRIGGER #16)
This trigger fires anytime Bridge Button 1 is pressed. It resets the softkey screen and displays the default WIKD logo.

SCENE SELECTOR (TRIGGER #17)
This trigger fires anytime Bridge Button 12 is pressed. It opens a menu on the softkey screen that allows a user to choose between the Default, Band, and Production snapshots. If a fader is turned on or assigned to the PGM Bus, Supervisor will not attempt to modify that fader's source, instead displaying a warning message.

OPERATIONS BOARD COMMANDS (TRIGGER #18)
This trigger fires anytime Bridge Button 11 is pressed. It opens a menu on the softkey screen that allows the user to reset the board, reset the softkey screen, turn the bridge lamps on or off, enable or disable producer mode, and enter call recording mode.

OUTSIDE SPEAKERS (TRIGGER #24 & 25)
The outside speakers are turned off at Midnight every day, and then turned back on at 05:00.

STUDIO PHONE CALLS (TRIGGER #56 & 57)
This trigger is typically executed using the vButton control panel. It will route the Office Mic and a dimmed version of the PGM Bus to the Studio Headphones, so that you can talk to an On-Air DJ. This is especially useful if calls are being processed using the office phone, as it allows the Office to act as a call screener and notify the DJ's of the incoming callers.

OFFICE TALKBACK (TRIGGER #71 & 81)
This trigger fires anytime Softkey 1 on the Numix is pressed or the "Request/ End TB" button is pressed in vButton. Doing so will flash the "Accpt Request" button on the vButton control panel, or display a message on the softkey screen. Once either Softkey 3 or the "Accpt Request" button has been pressed, the Office mic will be routed to the Studio monitors, while Microphone 2 will be routed to the Office Speakers. The conversation can be ended at any time by pressing Softkey 1 or the "Request/ End TB" button in vButton.

SUBWOOFER VOLUME (TRIGGER #85 & 86)
Every weekday at 07:00 the output to the subwoofer is reduced. It is returned to a normal level at 17:00 every weekday.

EAS ALERT (TRIGGER #89 & 90)
This trigger will be activated anytime WIKD begins sending an EAS Alert so that can DJ's will know they are being interupted. As soon as a DJ sees the notice they should stop talking (if they are On-Air) and wait for the notice to pass. As soon as the alert has finished sending the trigger will close, causing the delay to dump three times (so that it is completely empty). As soon as the alert has passed, the DJ should resume talking to prevent dead air (noting that they have no dump).
This prevents their being a disorienting jump for the listener. If the DJ is playing music they do not need to do anything.

AUTOMATIC MICROPHONE PAN (TRIGGER #120-127)
Any time all four microphones are turned on and assigned to the PGM Bus, Microphones 1 & 2 will be panned to the left, and Microphones 3 & 4 will be panned to the right. This allows four people to talk simultaneously, whilst still maintaining clarity and definition. As soon as any of the microphones are turned off, all of the microphones return to being center panned.

PRODUCER MODE
Producer Mode can be enabled from the Operations Board Command Menu (Bridge Button 11). Doing so will un-assign Microphone 2 from the PGM Bus, and then route Microphone 2 and a dimmed version of the PGM bus to the Studio Headphones. This allows the Board Operator to communicate with the rest of the DJ's while they are On-Air. Pressing Softkey 7 exits Producer Mode.

CALL RECORDING MODE
Call Recording Mode can be enabled from the Operations Board Command Menu (Bridge Button 11). Doing so will turn all four studio microphones and the phone on and assign them to the AUX 5 Bus. It will also assign the four studio microphones to the AUX 6 Bus, while assigning the Phone to the AUX 7 Bus. The AUX 5 Bus will be routed to the Studio Headphones, while the AUX 6 Bus (Studio Microphones) will be routed to the Phone and to the digital input of REC CPU. The AUX 7 Bus (Phone) will be routed to the analog input of REC CPU. Pro Tools will also be launched. This allows a two track recording to be created with the microphones and phone audio on separate tracks. This makes it easy to edit a phone call for On-Air use.

To exit Call Recording Mode, simply turn off the Phone fader.

RF PART 8 FUNDAMENTALS

· THE BASICS

· STEREO GENERATION

· RDBS

· CREATING THE MPX SIGNAL

· TRANSMISSION POWER

· ANTENNA POLARIZATION

· ANTENNA GAIN

· COVERAGE PREDICTION

· TRANSMITTER SAFETY

THE BASICS

There are two major types of radio transmissions—AM (Amplitude Modulation), and Frequency Modulation. In America, AM stations are licensed between 540 KHz and 1710 KHz in 10 KHz increments. FM Stations are licensed between 87.7 MHz and 108.0 MHz in 200 KHz increments.
The general theory of how AM and FM work is the same. A carrier wave in either the AM or FM band (102.5 MHz in WIKD's case) is modulated by the electrical signals of the audio. In AM, the amplitude of the carrier wave is adjusted based on the modulating signal (the audio), while the frequency remains constant. In FM, the amplitude of the carrier wave is constant, but the frequency and phase of the carrier wave are adjusted.

The technical explanation for how audio signals become radio signals works something like this:

A voltage controlled oscillator (VCO) generates a carrier frequency—this is an electrical signal that will be modulated to carry the message audio. The signal carrying the message is applied to the input of the VCO. As the voltage of the message signal is increased, the frequency of the carrier signal is increased, and vice versa.

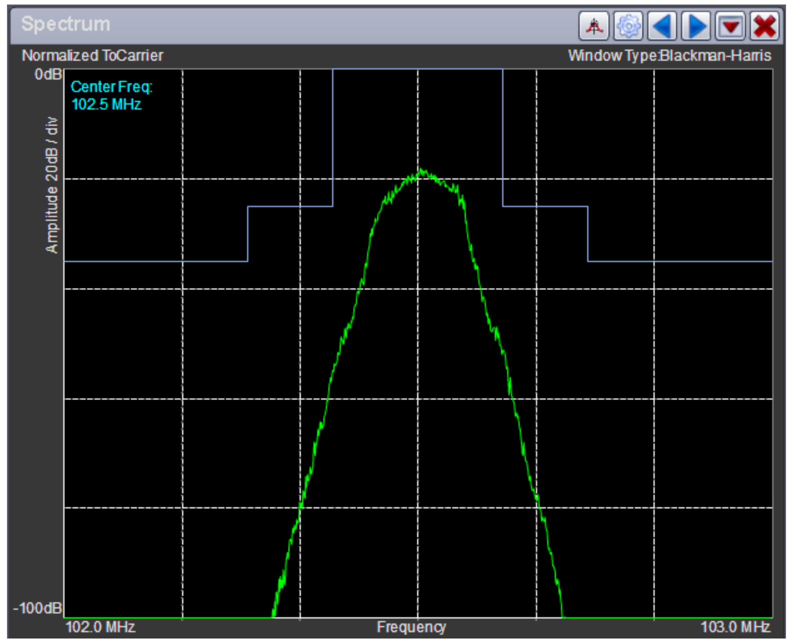
For example, if a carrier wave signal with a frequency of 100 MHz is modulated by a 1000 Hz sine wave with a peak voltage of 1V, the modulated signal would increase to 100.02 MHz, decrease to 99.98 MHz, and then return to a frequency of 100 MHz in a single millisecond (which, incidentally, is the period of a 1000Hz tone.) This results in a peak deviation of ±20 KHz. If you were to increase the voltage of the message to 2V, the peak deviation would be ±40 KHz.

Remember! The louder the audio, the greater the deviation from the carrier frequency.

Unfortunately, it's a rare occasion when you are broadcasting a simple tone, and it's an even rarer occasion when the math works out so simply, but hopefully you get the idea.
In practice, when an actual audio signal is applied to a carrier signal, many things happen. Remember how every single sound, no matter how complex, can be shown to be the composite of various sine and cosine waves? (If you don't that's cool too I suppose. Check out Fourier Series.)

Well, when a complicated audio signal is applied to a carrier signal, the resulting signal will be incredibly complex. But, if you were somehow able to look at the audio signal and analyze it for a single instant in time, you could actually see all the different frequencies that make up that signal, and the amplitude of each of those frequencies. This actually exists, it's called a frequency response chart, and it can be created by taking the Fourier transform of the audio signal.

If you were to look at the frequency response chart for the modulated signal you would likely see something like this:



Notice how the signal occupies almost 200 KHz of bandwidth? If you think about this, it actually makes a little bit of sense. Consider the modulated signal from the example above—as the frequency of the modulated signal increases from 100 MHz to 100.02 MHz, the frequency will pass through every value in between 100 and 100.02 MHz. When you analyze the output signal over a period of time you can see an average of how much time the signal spent at various frequencies.

As a result, when you look at the modulated signal using a spectrum analyzer you get a really complex waveform. In fact, while the majority of the energy will be concentrated around the carrier frequency, there will be "sidebands" that also contain a significant amount of energy. And while, theoretically, there are an infinite number of sidebands, generally only the ones closest to the carrier frequency have enough power to be significant.

Remember how amplitude and frequency deviation are directly related? The reason it's so important is that if your audio was loud enough, you could generate significant sidebands far removed from the carrier frequency. Eventually, those sidebands will start interfering with other radio signals.
Luckily, our friends at the FCC (no really!), have established some rules designed to stop radio stations from interfering with each other. Specifically, the modulated signal may not deviate from the carrier signal by more than ±75 KHz. This is referred to as the 100% modulation limit. In certain cases, stations can modulate up to 110%.

STEREO GENERATION

Before the 1960's mono sound was the standard for AM, FM, and TV. In 1961, however, the FCC approved the use of stereophonic sound, provided it was implemented in such a way that the stereo signal was still backwards compatible with monophonic receivers.

The solution that was devised involved transmitting a mono signal (generated by adding the Left and Right channels), a difference signal (generated by subtracting the Right channel from the Left channel), and a stereo pilot signal (a 19 KHz tone). Stereo FM receivers listening to this signal could recreate the stereo sound by adding the mono signal to the difference signal to generate the Left channel ([L+R] + [L-R] = 2L). The Right channel is generated by subtracting the difference signal from the mono signal ([L+R] - [L-R] = 2R).

The presence of the 19 KHz pilot tone indicates to a receiver that there is stereo content available. It also serves as a reference to allow the FM receiver to decode the stereo information.

RADIO DATA SYSTEM

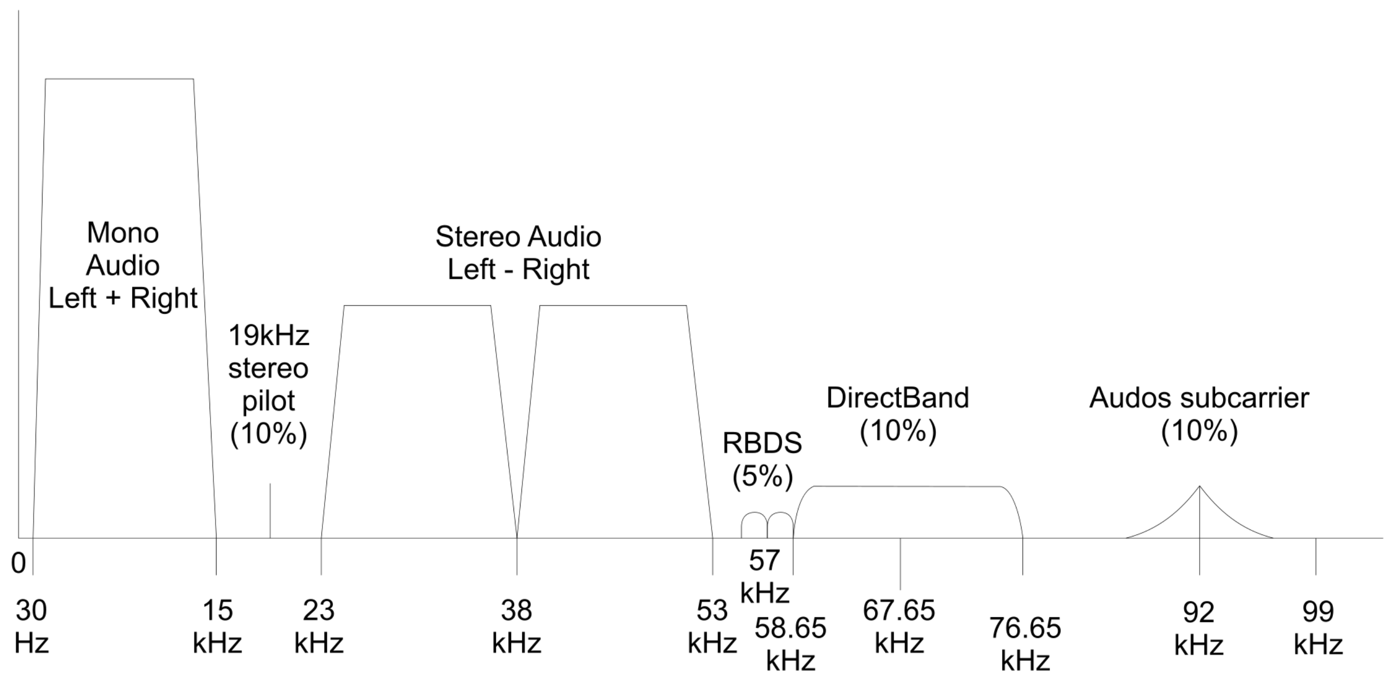
The Radio Data Broadcast System, more commonly known by its European name, RDS, is a system for transmitting small amounts of digital information over the FM broadcast band. RDS can be used to transmit a multitude of different types of data, but most typically it is used to convey the station's name, as well as the artist and title of the current song.

The two most common data codes are PS (Programme Service), and RT (Radio Text). PS Codes are 8 character static displays, typically used to give the call letters or station ID. Some RDS Encoders will can create dynamic PS codes, but this is rare.

RT codes allow for messages up to 64 characters in length, and are most commonly used to display the song artist and title, as well as station promos. Receivers that support RT codes are relatively rare, though newer cars with higher end radios will support it.

CREATING THE MPX SIGNAL

The multiplex (MPX) signal is the composite signal of the mono audio, stereo audio, pilot tone, RDS, and any other subcarrier signals. The spectrum looks something like this:



In order to transmit the stereo audio and RDS subcarriers in addition to the mono audio, one must engage in some mathematical trickery, known as frequency division multiplexing (FDM). FDM allows you transmit several signals at once by using a separate frequency band for each signal. In FM Broadcasting, the mono audio occupies the space from 50 HZ to 15 KHz. The stereo audio signal is generated using a form of amplitude modulation known as DSB-SC (Double-Sideband Suppressed-Carrier Transmission). Like FM, this method uses a carrier wave, but it is suppressed so as not to waste energy.

One of the key points about this form of AM is that the message audio is contained in the sidebands—the carrier wave does not carry any information, other than to say that the information exists. The sidebands are mirror copies of each other. This is why the stereo audio occupies twice the spectrum bandwidth of the mono signal. One will also notice that the stereo audio is centered on 38 KHz—this is where the suppressed carrier operates.

The reason the carrier wave can be suppressed is because of the 19 KHz pilot tone. FM Stereo receivers can multiply the pilot signal by two to generate the 38 KHz carrier wave necessary to decode the stereo audio signal.

One will also notice that the RDS data is centered around 57 KHz, the third harmonic of 19 KHz for the same reason.

However, remember how the FCC mandates a total modulation limit of 100% (i.e. a maximum frequency deviation of ±75 KHz)? If all of the subcarriers were transmitted at the same level as the mono audio signal, the modulation would likely be far in excess of the 100% modulation limit.

As such, the FCC has established limits on the amount to which subcarriers can be injected. For stereophonic transmissions with no subcarriers the total peak modulation must not be more than 100%. For every 1.0% of subcarrier injection, a 0.5% increase in the peak modulation limit is tolerated.

For example, WIKD uses only one subcarrier—the RDBS data, which is injected into the composite signal at a level of 10%. Therefore, we are permitted to have a maximum peak modulation level of 105%.

Luckily, our Nautel transmitter takes care of all of this, by employing a hard limiter to ensure modulation levels do not peak past 105%.

TRANSMISSION POWER

The modulation of the carrier wave isn't the only thing that's strictly regulated by the FCC. Transmitter Power Output must be closely monitored. According to the FCC TPO must be maintained within 90% and 105% of the approved TPO. The approved TPO is specified on the station's license. WIKD's approved TPO is 262W.

But wait. LPFM stations are only licensed for a maximum of 100W right?

Sort of. LPFM stations can be licensed for up to 100W Effectively Radiated Power (ERP). This is the power that's actually radiated from the antenna, and it takes into consideration Transmitter Power Output, transmission line loss, connector loss, antenna gain, and antenna polarization. ERP can be calculated by using the following formula:

$$TPO = \frac{ERP}{10^{\frac{-\text{dB Atten.}}{10}}}$$

Here, the ERP would be 100W, while the decibel amount of attenuation is calculated by summing the antenna gain (in decibels), with the decibel loss for each of the connectors, and the decibel loss of 150ft. of coaxial cable.

$$\text{dB Atten.} = \text{Antenna Gain} + \text{Connection Loss} + \text{Line Loss}$$
$$\text{dB Atten.} = -3.37\text{dB} + 2 * 0.5\text{dB} + 1.5 * .661\text{dB}$$
$$\text{dB Atten.} = -4.4615\text{dB}$$

Therefore, in order to achieve an ERP of 94W (specified by our license), WIKD must use a TPO of 262W.

$$TPO = \frac{94W}{10^{\frac{-4.4615dB}{10}}}$$
$$TPO = 262.590W$$

ANTENNA POLARIZATION

Radio antennas come in a multitude of varieties but they can be broadly grouped by their polarization. In general antennas are either Circularly, Horizontally, or Vertically polarized. In short, the polarization of an antenna refers to the direction in which electrical waves are transmitted relative to the direction of travel.

The important part of all this is that if a receiver antenna is vertically polarized it can't receive horizontally polarized signals and vice versa. Circularly polarized signals have equally powerful components in both the horizontal and vertical planes, and can therefore be received by any type of antenna. As a result, CP antennas require twice the power of a horizontally or vertically polarized antenna.

For the most part, most FM transmitter antennas are either circularly or horizontally polarized. In rare cases some stations will use a vertically polarized antenna to minimize interference with TV stations. (TV channel 6 occupies the space from 82-88 MHz, with the sound carrier at 87.75 MHz. FM stations operating in the noncommercial education band (88-92 MHz) can sometimes cause significant interference to nearby Channel 6 TV stations. As a result, some stations elect to use a vertically polarized antenna to minimize interference.)

It is important to remember that the as a signal travels from the transmitter antenna to a receiver antenna, the polarization of the signal will likely change, if only because of the angle of incidence. In addition, any reflections will change the polarization. As a result, most receiver antennas are circularly polarized.

ANTENNA GAIN

In addition to the polarization of an antenna, the other defining feature is the gain of the antenna. The gain of an antenna determines how much power input will be required to achieve the required ERP. For example, WIKD's SPX DCR-L1 has a gain of .47, or -3.37dB. Disregarding all other considerations, the antenna would require a power input of 213W to achieve an ERP of 100W. This makes sense because the DCR-L1 is circularly polarized, and therefore radiates 100W in both the horizontal and vertical planes.

The antenna gain can be increased by using multiple bay antennas—essentially, stacking a bunch of antennas on top of each other. (Following in the time honored tradition of "if one is good, two is better"). However, really high gain antennas (i.e. 6-10 bay antennas), can cause areas of poor reception near the transmitter. Thankfully, this won't really be a problem for WIKD. ☺

COVERAGE PREDICTION

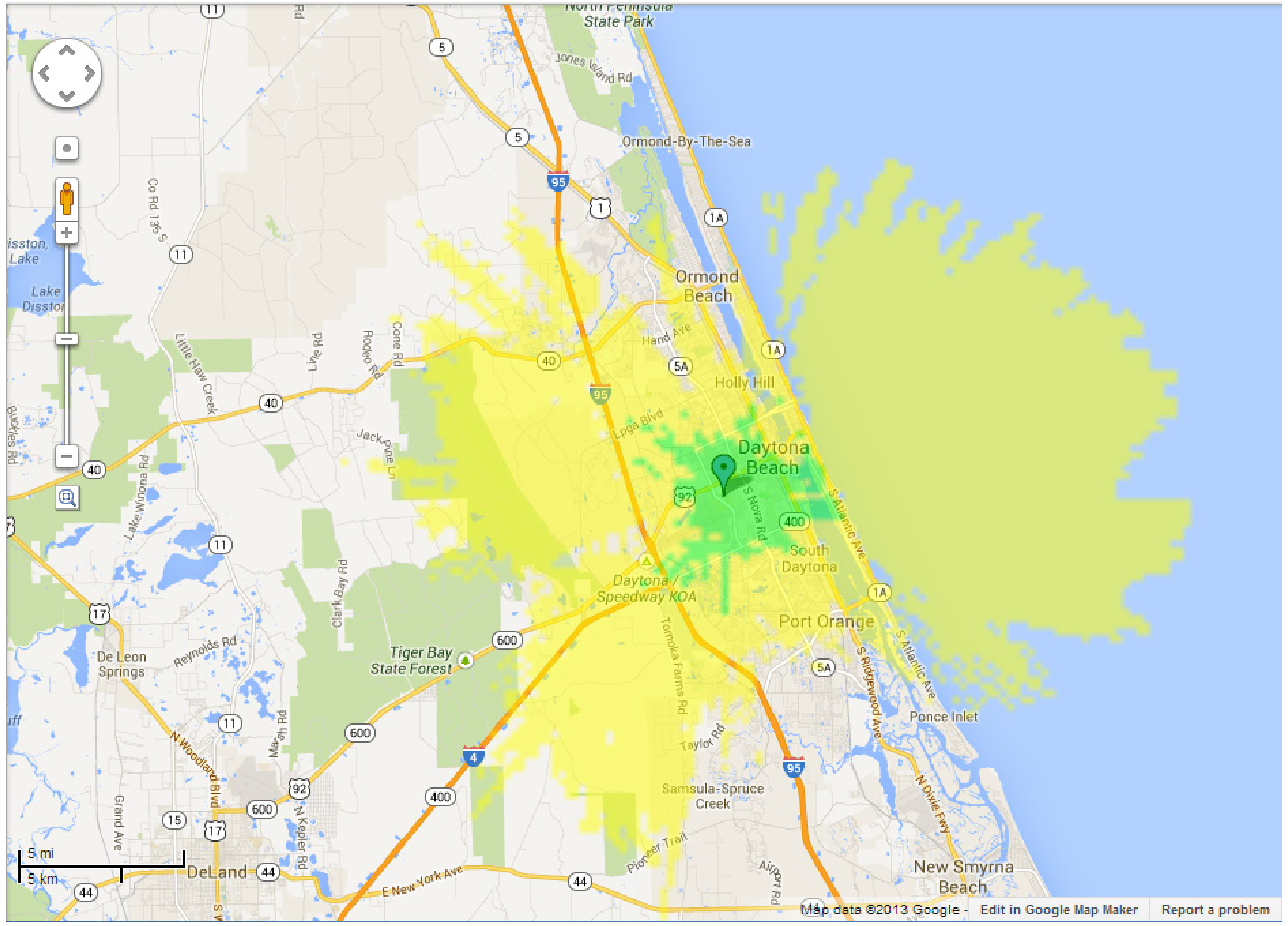
It is actually surprisingly complicated to generate a map of predicted coverage for a specific radio station. This is mostly because radio signals do a lot of funny things in between the transmitter and receiver, which are really hard to account for, and can have a significant impact on the coverage of a signal.

Nonetheless, there are still several methods. The most important of these is defined by the FCC and is called the 60dBu Service Contour. Incidentally, it also corresponds to the area of protected service for LPFM stations. Theoretically, if a listener is located within the 60dBu service contour they should be able to get good, consistent reception.

The 60dBu service contour is calculated by using the FCC F(50,50) charts. These charts correlate the transmitting antenna height with the expected signal strength at a specific distance. They are called the F(50,50) charts because they are based on a specified signal strength occurring at least 50% of time, for at least 50% of the locations located at the specified distance, and at a height of 9m.

For example, by using the FCC's service contour calculator, we can see that LPFM stations (100W at 30m) have a 60dBu service contour located at 5.6 km from the transmitter. This, however, is a broad approximation of the actual coverage, and is really only useful as a legal tool.

The reason the FCC's service contour maps are not necessarily accurate is because they neglect so many factors—namely the surrounding terrain, and any RF interference. There are more advanced models, but these typically require the help of a specialized broadcast engineer to generate. However, you can play around with Radio Mobile's free web based tool for generating coverage maps using the Longley Rice model here.



TRANSMITTER SAFETY

As one would imagine, transmitters are very high powered devices. In addition, antennas are essentially giant lightning rods floating high in the sky. Also, Florida is the lighting capital of the U.S. As you can imagine, lighting protection is somewhat important.

The general theory behind lighting protection goes like this: It can't be stopped, so send it somewhere that doesn't matter as quickly as possible (i.e. before it blows anything/anyone important up).
If you're really interested in this, Nautel has a tremendous guide to transmitter site preparation available here, and Middle Atlantic has an equally excellent guide to electrical wiring available here. I'm only going to cover how WIKD is setup.

The cornerstone of our lighting protection system is the use of a single point of grounding, which is connected to a copper counterpoise system. The counterpoise consists of four 12' #2 copper rods buried 1' away from the shed. The rods are then connected to each other by #2 copper rods buried 6" deep. All the joints were flash welded together. The counterpoise is connected to a solid copper ground bus, located to the right of the entrance door.

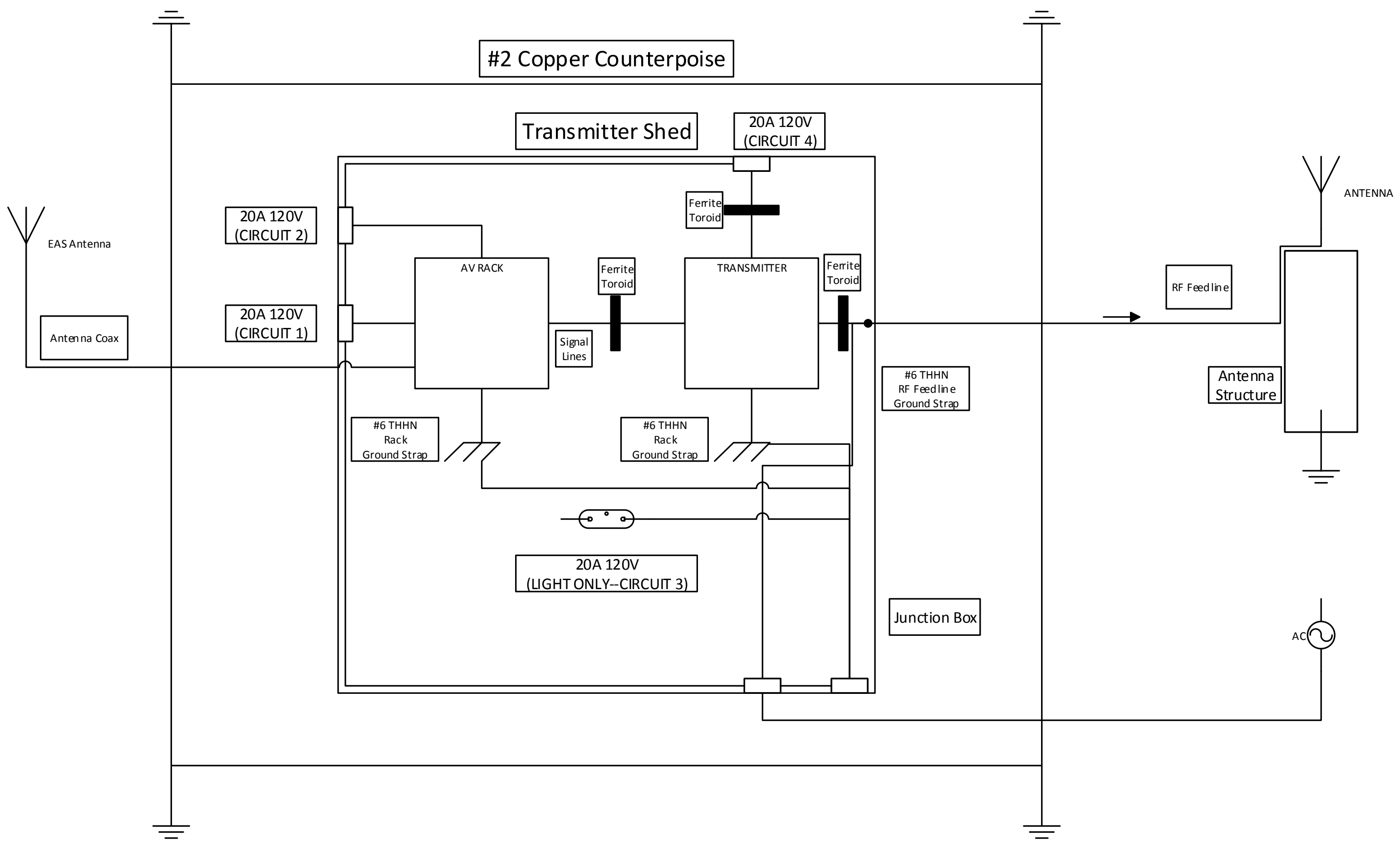
The ground from the incoming AC Power is bonded to the ground bus as soon as it enters the shed.

Both the AV and Transmitter racks are connected to the ground bus by #6 THHN copper wire. This is to ensure that the racks exhibit the same potential as all of their equipment—thereby ensuring (hopefully), that the racks do not become electrified in the event of a lightning surge.
The reference (signal) ground of the Nautel is connected to the transmitter rack. This isn't ideal—technically it should be connected directly to the ground bus, but we haven't experienced any problems so far.

The coaxial cables to both the transmission and reception antennas both have their shields tapped and connected to the ground bus.

There are six separate circuits available in the shed, as follows:
- Circuit 1: 20A, 120V, AV RACK
- Circuit 2: 20A, 120V, AV RACK
- Circuit 3: 20A, 120V, Overhead Light
- Circuit 4: 30A, 120V, AV Rack
- Circuit 5: 40A, 208V, Air Conditioner
- Circuit 6: 20A, 120V, Transmitter Rack

The system was specified by Aiden Jones of Giles Electric, and installed by Dennis Fiddler, in conjunction with the ERAU Facilities Management department. If you ever find any issues with it, immediately contact Facilities Management.



It is a bad idea to lick the output of the transmitter.

FCC PART 9 REGULATIONS

· BASICS

· LPFM SPECIFIC RULES & REGULATIONS

· RULES APPLICABLE TO ALL BROADCAST STATIONS

· PHONE CALLS & CONTESTS

· THE EMERGENCY ALERT SYSTEM

· OBSCENE, INDECENT, AND PROFANE MATERIAL

· "ENHANCED UNDERWRITINGS"

·TECHNICAL TRANSMISSION REGULATIONS

THE CODE OF FEDERAL REGULATIONS

The single most important responsibility for the Chief Engineer is making sure that WIKD is compliant with all FCC regulations. Failure to do so can result in fines (large ones!), or even worse, the revocation of our license.

This guide is intended to highlight some of the most important regulations relevant to maintaining day to day compliance, but by no means are they a thorough guide. It is good practice to periodically at least skim through the CFR's.

The relevant regulations are contained in Chapter 1, Subchapter C, Part 73 of Title 47 (Telecommunications) of the Code of Federal Regulations. Regulations concerning the Emergency Alert System are found in Chapter 1, Subchapter A, Part 11 of Title 47.

There are several different ways you can view the CFR's. The most official method is to use the Electronic CFR published by the Government Printing Office, available here.

Each year, (October 1st for Title 47), all of the changes to the CFR's are compiled, and the yearly copy of the CFR's are released. A hard copy of the previous year's CFR's are usually made available in January. You can also purchase a hard copy of the CFR's from the Government Printing Office.
However, the FCC continuously amends the CFR's throughout the year, and while the ECFR will reflect these changes, there is no easy way to see what changes have been made.

Luckily, the Cornell University Law School maintains an electronic copy of the CFR's that list all changes made after the publication of the yearly edition of the CFR's. It is considerably easier to use and read.

47 CFR Part 73 is additionally divided into a number of subparts as follows:

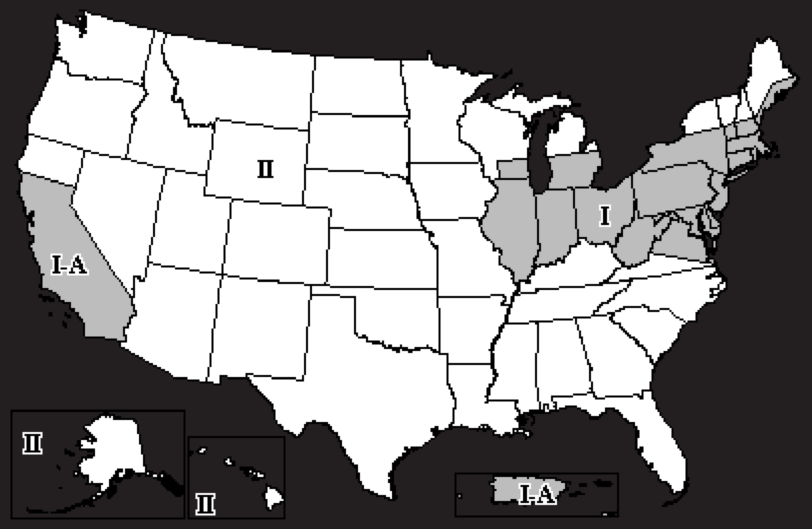
Important parts are bolded.

Subpart A – AM Broadcast Stations (73.1-73.190)
Subpart B – FM Broadcast Stations (73.201-73.333)
Subpart C – Digital Audio Broadcasting (73.401-73.404)
Subpart D – Noncommercial Educational FM Broadcast Stations (73.501-73.599)
Subpart E – Television Broadcast Stations (73.601-73.699)
Subpart F – International Broadcast Stations (73.701-73.788)
Subpart G – Low Power FM Broadcast Stations (LPFM) (73.801-73.881)
Subpart H – Rules Applicable to All Broadcast Stations (73.1001-73.4280)
Subpart I – Procedures for Competitive Bidding and for Applications for Noncommercial Educational Broadcast Stations on Non-Reserved Channels (73.5000-73.5009)
Subpart K – Class A Television Broadcast Stations (73.6000-73.6027)
Subpart L – Incorporated Standards (73.8000-73.8000)
47 CFR §73.310—FM Technical Definitions
47 CFR §73.3500—Application & Report Forms

THE
B A S I C S

• The FCC has designated the FM band in America to consist of frequencies from 87.9MHz to 107.9MHz. The FCC has divided the FM band into 201 separate channels, with channel 200 being 87.9MHz and channel 300 being 107.9MHz. WIKD occupies channel 273.

• The FCC has also divided the United States into three different zones; Zone I, Zone I-A, and Zone II. Florida is in Zone II. The divisions were originally based on the population distribution in



• The FCC classifies stations based upon their ERP (Effective Radiated Power), and their HAAT (Height Above Average Terrain). The most powerful class of station is a C class station which is licensed for a minimum of 100KW ERP, with an antenna height of at least 451 meters HAAT. The complete definition for station classes can be found in 47 CFR §73.211.

Station Class	Minimum ERP	Maximum ERP	Minimum HAAT	Contour Distance (km)
A	0.1 KW		100m	28km
B1	6 KW	6 KW	100m	39km
B	25 KW	50 KW	150m	52km
C3	6 KW	25KW	100m	39km
C2	25 KW	50 KW	150m	52km
C1	50 KW	100 KW	299m	72km
C0	100 KW	100 KW	450m	83km
C	100 KW	100 KW	600m	92km

• The FCC maintains regulations defining how close station may be to each other. These regulations are determined based primarily upon the stations ERP, and antenna HAAT. In addition, the FCC mandates different distances based upon whether stations are co-located on the same channel, and first, second, or third adjacent to each other. The complete definitions can be found in 47 CFR §73.807.

• WIKD's nearest co-channel station is W273CA-FM, "WLOQ 102.5—Orlando's Smooth Jazz". They are a translator station currently operated by Clear Channel.
  • WIKD's first adjacent frequencies are 102.3MHz, and 102.7MHz.
  • WIKD's second adjacent frequencies are 102.1MHz and 102.9MHz.
  • WIKD's third adjacent frequencies are 101.9MHz and 103.1MHz.

• In addition the FCC classifies stations as AM Broadcast Stations, FM Broadcasts Stations, Non-Commercial Educational FM Broadcast Stations, Digital Audio Broadcast Stations, and Low Power FM broadcast stations. In addition there are FM boosters and translators, which, at a basic level, rebroadcast another station's signal.

LPFM SPECIFIC RULES

- LPFM stations are required, at a minimum, to broadcast at least 36 hours per week, with at least 5 hours of operation on at least 6 of the days of the week. However, stations licensed to educational institutes (such as WIKD!) are not required to operate on Saturday or Sunday, or to observe the minimum requirements during days which are designated by the school calendar to be vacation or recess periods.

- LPFM stations are required to submit applications to the FCC in order to be granted a license. These applications can later be amended or changed. A minor amendment/change is classified as:

    - The relocation of the transmitter site by 5.6 kilometers or less.

    - Changes in the operating frequency (Provided that reduced interference can be shown).

    - Amendments to the time-share agreement.

- A major amendment/change is a change which drastically alters the specifications of the original license.

- Minor amendments/changes can be filed at any time; Major amendments/changes can only be filed during filing windows.

- WIKD's license expires February 1, 2020 at 3:00AM. The FCC will open a license renewal window in 2019.

- LPFM stations are NOT required to keep a Public File. A public file is a large and complicated document which details a stations operation and it's compliance with numerous FCC regulations and standards.

- Instead, WIKD is required to keep a Station Log which contains an equipment maintenance log, an EAS activation log, the most recent copy of the station's license, and a political file notification.

- The equipment maintenance log details any station outages due to equipment malfunction, servicing or replacement, as well as any operations not in accordance with the station's license.

- Any entries to this log must be signed and dated, as well as including the time. Sheets must be numbered and dated. If corrections must be made, cross through the erroneous portion and write the correction next to it. Corrections must be signed and dated.

- EAS logs contain an entry detailing all sent and received EAS activations.

- A political file notification is simply a notification that WIKD has not given air time to any political candidate during the current year.

- These logs must be kept for 2 years.

- Letters from the public are not required to be kept, but it is good practice to do so.

RULES APPLICABLE TO ALL BROADCASTERS

- The FCC can conduct inspections during normal business hours and at any time the station is in operation. However, because WIKD has undergone a Mock Inspection by the FAB (Florida Association of Broadcasters), WIKD is exempt from random FCC inspections until June 6, 2015, UNLESS there has been a public complaint, or there is a noted issue with the tower & it's lighting, or in the event that WIKD causes interference with another station or entity.

- WIKD is not required to observe ANY of the rules regarding the antenna tower lighting, because the tower is Embry Riddle's responsibility.

- The FCC requires that all stations issue a Station ID at the following times:
  At the beginning and end of operation.
  At the top of every hour OR at a natural break in programming.

- Station ID's must consist of the stations call letters, immediately followed by the station's city of license. For WIKD this is:
  "WIKD-LP DAYTONA BEACH"

PHONE CALLS & CONTESTS

- The FCC has declared that it is illegal to broadcast or record a caller without first notifying them of the station's intention, UNLESS the caller can be assumed to be aware.

- An aware caller is considered to be someone who is either associated with the station or the caller has initiated the call, and it is obvious that it is in connection with a program during which phone calls are traditionally aired live.

- In the event that WIKD chooses to host a contest or lottery, then all relevant rules and regulations must be aired alongside first announcement of the contest/lottery. Relevant rules generally include:

  - How to enter or participate
  - Eligibility requirements
  - Entry deadline dates
  - When the prizes can be won
  - The nature and value of the prizes
  - The time and means of selection of the winner

- The relevant rules must be broadcast periodically afterwards, however they do not need to be announced every time a reference is made to the contest.

GENERAL FCC REGULATIONS

- The FCC classifies stations as being attended or unattended. Attended stations are those which are staffed 24/7 by an operator responsible for operating/ overseeing the transmission and EAS equipment. Unattended stations are those which employ highly stable equipment and monitoring devices which are capable of notifying staff personnel or shutting down out-of-tolerance equipment within 3 hours. WIKD is considered to be an unattended station.

- Out-of-tolerance conditions that would necessitate a shutdown within 3 hours include the following:

  - Excessive TPO (Transmitter Power Output)
  - Excessive Modulation
  - The emission of spurious signals that do not result in harmful interference
  - Out-of-tolerance conditions that would necessitate a shutdown within 3 minutes include the following:
  - The emission of spurious signals that DO result in harmful interference.
  - The operation of the station outside its specified operating hours.
  - Operation of the stations which is substantially variant from its authorized radiation pattern.

- To help with the maintaining of station logs, the FCC requires that stations employ a Chief Operator. The Chief Operator a person designated by the station to oversee the stations logs, as well as its transmission and EAS equipment. In the event of an out-of-tolerance condition they should notify the station's Chief Engineer.

- In the event that a station needs to operate outside the parameters specified by its license, the station can apply for a STA (Special Temporary Authorization). The request should be made at least 10 days in advance of the proposed operation.

FLORIDA EAS DETAILS

- WIKD is part of Operational Area 7. The division of Florida into operational areas is specified in the State EAS Plan.

- WIKD's LP-1 is WPOZ 88.3MHz, Orlando. They are managed by Jim Hoge, and his contact information is available in the State EAS Plan. WPOZ also broadcasts on the following frequencies:
    - WHYZ 91.1 Palm Coast
    - WMYZ 88.7 Clermont
    - W245AZ 96.9 Melbourne/Palm Bay (Translator)
    - W259AS 99.7 Lady Lake/The Villages (Translator)

- WIKD's possible LP-2's are:
    - 92.3 WWKA Orlando
    - 540 WFLF Pine Hills

- The PEP stations for Florida are:
    - WOKV 690 AM, Jacksonville
    - WFLF 540 AM, Orlando
    - WAQI 710 AM, Miami

- In addition, Presidential Messages can be received by NPR stations through the NPR "Squawk Channel". For Operational Area 7, WUCF 89.9 FM, Orlando is the only station with an NPR "Squawk Channel" receiver.

EAS TESTS

- The FCC requires that stations conduct a weekly test of their EAS equipment. This is called a RWT (Required Weekly Test), and consists of the EAS alert tones. It can also include a a simple message indicating that it was simply a test of the system, however this is NOT required.

- Stations are NOT required to relay other stations RWT's.

- In addition, stations are required to conduct a monthly test to ensure that system as a whole functions for the operational area.
    - These tests are called RMT's (Required Monthly Tests).
    - A RMT can ONLY be originated from a LP station.
    - RMT's must be retransmitted by all PN EAS Participants.

- In the event that the Chief Operator notices that WIKD failed to either receive or send an EAS activation, the Chief Engineer must determine why WIKD did not receive/send the alert, take corrective action, and note both the problem and solution the EAS log.

OBSCENE, INDECENT & PROFANE MATERIAL

- Obscene material is defined by the FCC as any programming which passes the following three part test:
  - An average person, apply contemporary community standards, must find that the material as a whole is of an overtly sexual nature.
  - The material must depict or describe, in a patently offensive manner, sexual conduct as defined by law.
  - The material, taken as a whole, must lack serious literary, artistic, political or scientific value.

- Obscene material is NOT protected by the First Amendment, and as such, is prohibited at ALL times.

- The FCC defines indecent material to be that which describes in an overtly offensive manner, sexual, or excretory organs or activities.

- The FCC defines profanity to be language that is so grossly offensive to members of the public that it amounts to a nuisance if they hear it.

- The Supreme Court has determined that indecent material and profanity ARE protected by the First Amendment and as such cannot be prohibited. However, it can be restricted, and as such, is not allowed between the hours of 6:00AM and 10:00PM. Outside of these hours, it is not considered to be actionable by the FCC's Enforcement Bureau.

ENHANCED UNDERWRITINGS

- The FCC does not allow Non-Commercial or LPFM stations to air a commercial service. It explicitly prohibits the airing of a promotional announcement on behalf of for profit entities in exchange for payment of any sort.

- Instead, stations can acknowledge contributions from donors or sponsors.

- In addition to these acknowledgements, the FCC allows for an "Enhanced Underwriting" , which is defined as an acknowledgement which includes the following:
  - Logograms or slogans which identify and do NOT promote.
  - Location Information.
  - Value Neutral descriptions of a product line or service.
  - Brand and Trade names and product of service listings.

- The FCC explicitly prohibits the following in "Enhanced Underwritings":
  - Announcements containing price information.
  - Announcements containing a call to action.
  - Announcements containing an inducement to buy, sell, rent, or lease.
  - Language which is clearly promotional or which compares the product to its competitors.

THE EMERGENCY ALERT SYSTEM

- The EAS (Emergency Alert System) is an emergency warning system that is designed to allow the President of the United States to communicate with the American Public within 15 minutes of its activation. A national alert has never been triggered, and has only been tested once.

- The EAS works by dividing the United States into approximately 550 “Local Areas”, each with two “Local Primary” stations. All of the participating broadcast stations in the Local Area are required to monitor these LP stations. In turn, the LP stations are required to monitor at least two sources of Presidential Messages, either a PEP station or State Primary Station.

- A PEP station is designated a National Primary (NP) station because it is a source of National Activations. A State Primary (SP) is a source of State Level Activations originating from the Governor or Emergency Operations Center.

- In the event of a national activation, FEMA or the White House will initiate a Presidential Message that is relayed to each of the 77 PEP stations. In turn, each of the LP-1 stations then begin immediately retransmitting the Presidential message. This message is then rebroadcast by all of the remaining stations.

- In addition to monitoring NP or SP stations, LP stations typically monitor the NWS, NOAA, and local emergency management offices. These alerts make up approximately 70% of all EAS activations.

- All radio stations are designated as EAS Participants unless they are designated as a NP, SP, or LP. However, EAS Participants can choose to be Pariticipating National stations (PN), or Non-participating National stations (NN). NN stations are stations which, in the event of a national activation, transmit the EAS header tones and then sign off for the duration of the alert.

- To summarize:
    - Presidential Messages originate from FEMA/The White House, and then from PEP stations.
    - State Messages originate from the State Governor or the State Emergency Operations Center.
    - Local Messages originate from LP stations.
    - WIKD is a Participating National (PN) Station.

- In addition to the standard EAS, the FCC has introduced a new system called IPAWS (Integrated Public Alert and Warning System). It is a nationwide system that makes use of multiple mediums to disseminate alerts including SMS, Billboards, the Internet, NOAA/NWS, and traditional broadcasts.
    - One of the key features of the IPAWS is the ability for stations to receive alerts through the internet.
    - Currently FEMA is conducting weekly tests of the IPAWS, and as such WIKD should receive a “CAP” alert from FEMA every Monday at 11:05AM local time.

OPERATING POWER REGULATIONS

- LPFM stations with an authorized Transmitter Power Output (TPO) of more than 10W must operate as close as possible to their authorized TPO. In addition, they may not operate at less than 90% of their authorized TPO, or greater than 105%. WIKD’s authorized TPO is specified on our license, and is 262W. This is specified by 47 CFR §73.840.

- The FCC specifies two methods for determining TPO, the direct method and the indirect method. This is specified by 47 CFR §73.267.
  - The direct method uses a calibrated transmission line meter.
  - The indirect method allows one to calculate the TPO by measuring the DC input current and voltage of the final stage RF amplifier, and multiplying them by an efficiency factor.
    - The FCC specifies three different methods for determining the efficiency factor. WIKD uses an efficiency factor of 62%, which is specified in the VS300LPFM’s data sheet.
    - The formula for calculating TPO is as follows:

      $$TPO = E_p \times I_p \times F$$

    - Where E_p is the DC input voltage, I_p is the DC input current, and F is the efficiency factor.
      The DC input current and voltage can be measured from the Nautel AUI. Open the Meters panel, select the Controller folder, and select “Total PA Current”, “Average PA Voltage”, and “DC-RF Efficiency”.

- WIKD uses a rackmounted Bird Wattmeter for determining TPO. The calibration certifications for the element currently in use are contained in the Chief Engineer binder. In the event that a new element must be ordered, make sure to order the calibration certificates as well.

MULTIPLEX SIGNALS & MODULATION LIMITS

- The FCC has set strict limits on the modulation of FM signals in order to prevent interference between FM stations.
- First and foremost, the total modulation deviation relative to the carrier signal may not exceed 75 KHz—this is the 100% limit. This applies for both stereo and mono broadcasts.
- If a station is using subcarriers, the total peak modulation may be increased 0.5% for each 1.0% of subcarrier injection.
    - Peak modulation may NEVER exceed 110%.
    - This is specified by 47 CFR §73.1570
    - For stations broadcasting in stereo the pilot tone must be injected at a level between 8% and 10%. This is specified by 47 CFR §73.322.
- FM stations must ensure that their transmitters operate within the following limits:
    - Any emissions appearing on a frequency removed from the carrier by between 120 KHz and 240 KHz must be attenuated by at least 25dB relative to the level of the unmodulated carrier. This is specified by 47 CFR §73.317.
    - Any emissions appearing on a frequency removed from the carrier by between 240 KHz and 600 KHz must be attenuated by at least 35dB relative to the level of the unmodulated carrier. This is specified by 47 CFR §73.317.
    - Any emission appearing on a frequency removed from the carrier by more than 600 KHz must be attenuated by at least 80dB or $43+\log_{10}$TPO (in Watts), whichever is the lesser attenuation. This is specified by 47 CFR §73.317.
    - The carrier frequency may not depart from its authorized center frequency by more than ±2000 Hz. This is specified by 47 CFR §73.1545.
    - The calibration documents provided by Nautel, as well as the spectrum analysis performed by WIKD in May of 2012 verify that the Nautel VS300LPFM operates within these bounds. It is a good idea to recertify the transmitter every few years to ensure compliance.
- As specified by 47 CFR §73.1590, anytime WIKD installs a new primary transmitter, or subcarrier or stereo generators, WIKD must take new measurements of the equipment performance, and certify that the equipment operates within the limits defined in 47 CFR §73.317.
    - These measurements, along with descriptions of the equipment and procedures used, must be kept on file at the transmitter or remote control site for a period of two years.
- As specified by 47 CFR §73.1660, FM Transmitters for LPFM stations must be Type Certified by the FCC. (Type approval or notification is not the same thing.) The Nautel VS300LPFM is Type Certified by the FCC.

HOW TO'S

PART 10

AND OTHER THINGS

- REMOTE ACCESS
- EAS LOGS
- OPERATIONAL LOGS
- CONTROL SIGNALS

REMOTE ACCESS

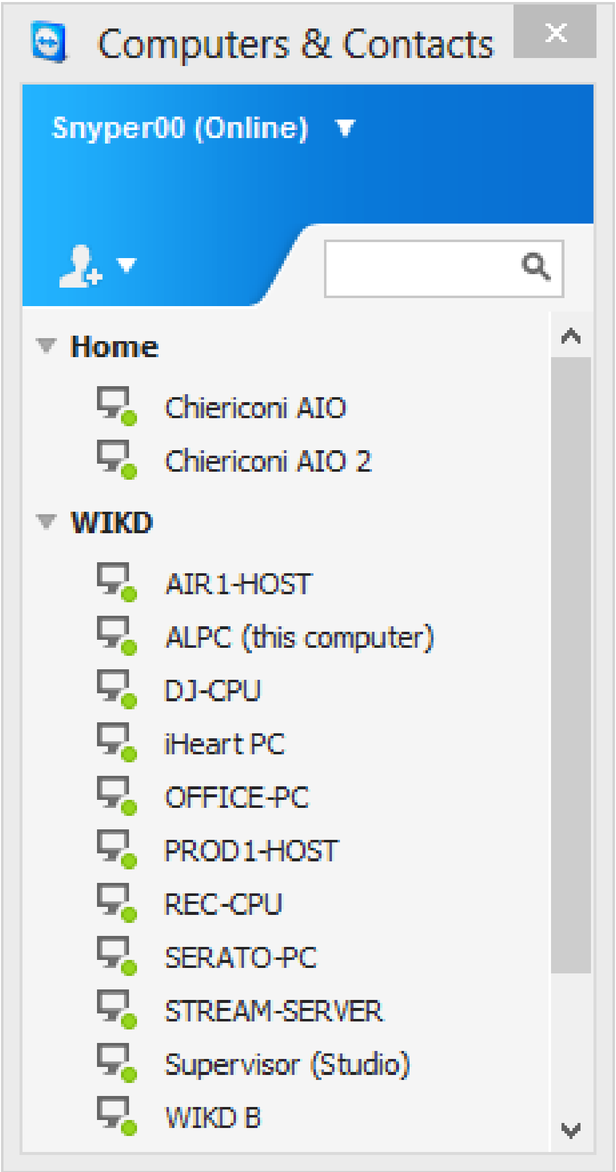
Remote access will save your ass. As long as WIKD still has power and internet, you can control nearly every piece of equipment from your home computer, tablet, or even cell phone.

There are several methods for accessing WIKD's computers and equipment remotely, and you will need to be familiar with all of them.

TEAMVIEWER

Teamviewer is a third party software that allows you to remote control computers on which the software is installed. Teamviewer is handy in that it allows you to access computers from web browsers, smartphones, and the Teamviewer application. Teamviewer allows you to access the following computers: AIR1-HOST, DJ-CPU, iHeart PC, OFFICE-PC, Propellers CPU, REC-CPU, SERATO-PC, STREAM-SERVER, and Supervisor. Teamviewer cannot be installed on computers running Windows Server.

Teamviewer is exceptionally useful for quick troubleshooting, or for showing a local user how to do something. In addition, Teamviewer includes a File Transfer utility that allows you to grab files without actuallyl having to remote into the computer.



REMOTE DESKTOP CONNECTION

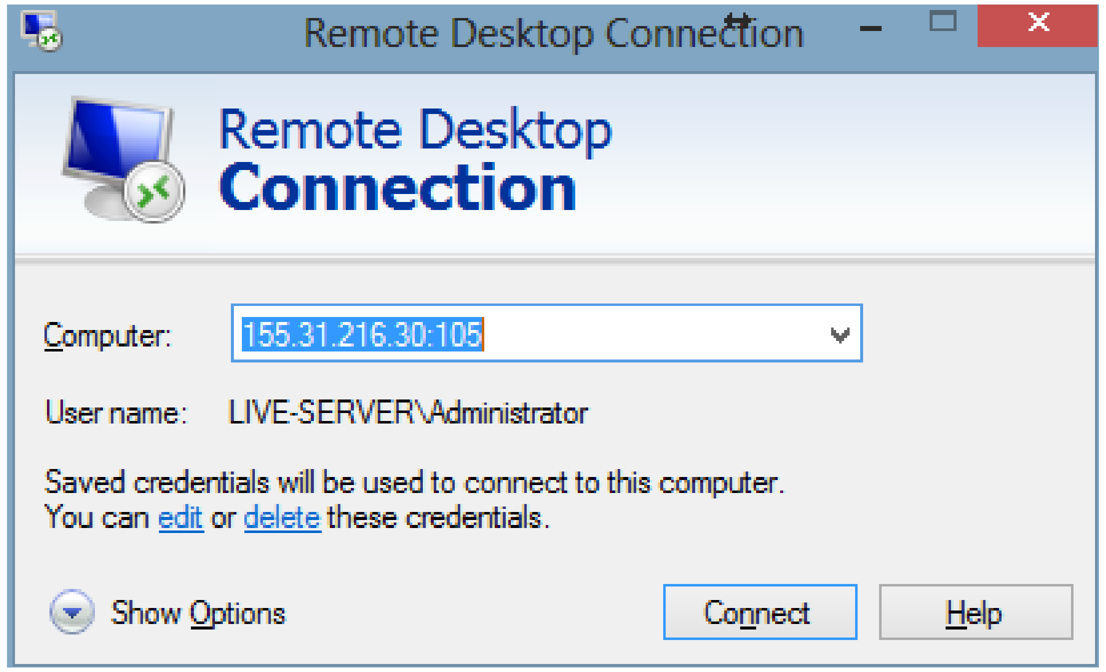
Microsoft Remote Desktop is a native Windows application that, like Teamviewer, allows you to remote control computers. It is different from Teamviewer in that it offers a more “natural” connection to the remote computer.

It is useful for more sustained periods of remote work. Remote Desktop is also more challenging to configure, but can be used on computers running Windows Server. If you are outside WIKD you can use Remote Desktop to connect to the following computers: AIR1-HOST, FILE-SERVER, LIVE-SERVER, OFFICE-PC, PROD1-HOST, and Supervisor.

1. The quickest way to launch Microsoft Remote Desktop is to open the Windows run dialog by pressing the Windows key and R together (⊞+R), and then launching “mstsc.exe”.

2. Alternatively, you can create a shortcut on your desktop. (Right click on the desktop and click “New>Shortcut” and setting the location to “mstsc.exe”.

3. Lastly, you can also download the collection of Remote Access shortcuts from here. These will immediately launch a connection to a specific computer. If you do that, you can skip the steps below.

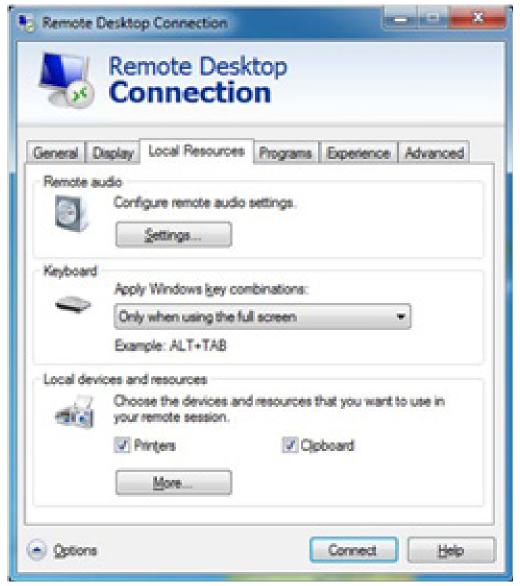


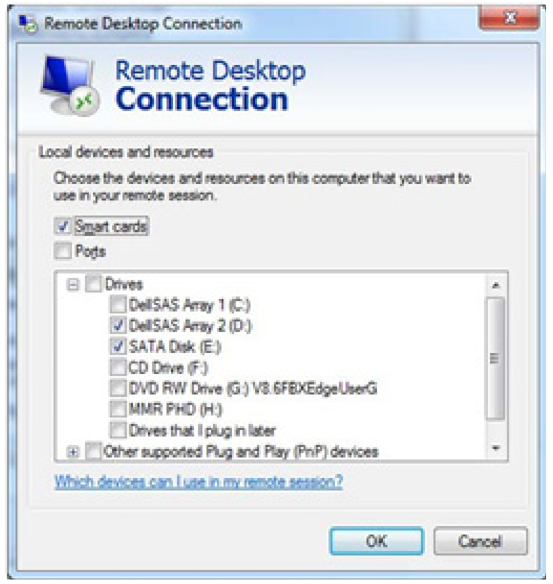
4. Once you have launched Remote Desktop Connection, you will need to specify the address of the computer you wish to connect to. This will change depending on whether you are inside WIKD or outside. You know the passwords.

COMPUTER NAME	INSIDE ADDRESS	OUTSIDE ADDRESS	USERNAME
AIR1-HOST	AIR1-HOST	155.31.216.30:85	AIR1-HOST\WOAFR
FILE-SERVER	FILE-SERVER	155.31.216.30:110	FILE-SERVER\Administrator
LIVE-SERVER	LIVE-SERVER	155.31.216.30:105	LIVE-SERVER\Administrator
OFFICE-PC	OFFICE-PC	155.31.216.30:90	OFFICE-PC\WOAFR
PROD1-HOST	PROD1-HOST	155.31.216.30:70	PROD1-HOST\WOAFR
SERATO	SERATO	NA	SERATO\SERATO
SUPERVISOR	SUPERVISOR	155.31.216.30:90	Supervisor\Eagles FM

5. A nifty little thing with RDC is the fact you can let the remote computer have access to the drives on your computer. Before establishing the connection, if you click on options, and click on the Local Resources tab, shown below.

In the Local devices and resources field, click on More. Under drives, select what you would like the remote computer to have access to on your local machine. In the example below I am allowing access to my physical D and E drives.





REMOTE EQUIPMENT CONNECTIONS

THE NAUTEL

The Nautel includes an Advanced User Interface (AUI) that allows you control every parameter of the transmitter. The main page allows you to view the modulation levels, power output, and various information about the signal itself. In addition, the transmitter can be powered on or off, and placed in Local or Remote mode. The transmitter should be placed in Local Mode ONLY if you need to make changes from the front panel, as it will lock out the AUI. Once the transmitter is placed in Local Mode, it can only be returned to Remote Mode from the front panel of the transmitter.

ALWAYS PUT THE TRANSMITTER BACK INTO REMOTE MODE.

The Nautel can be reached from within WIKDNET by going to 10.68.124.196. If you are outside WIKDNET, it can be reached by going to 155.31.216.30:4050.

THE OMNIA

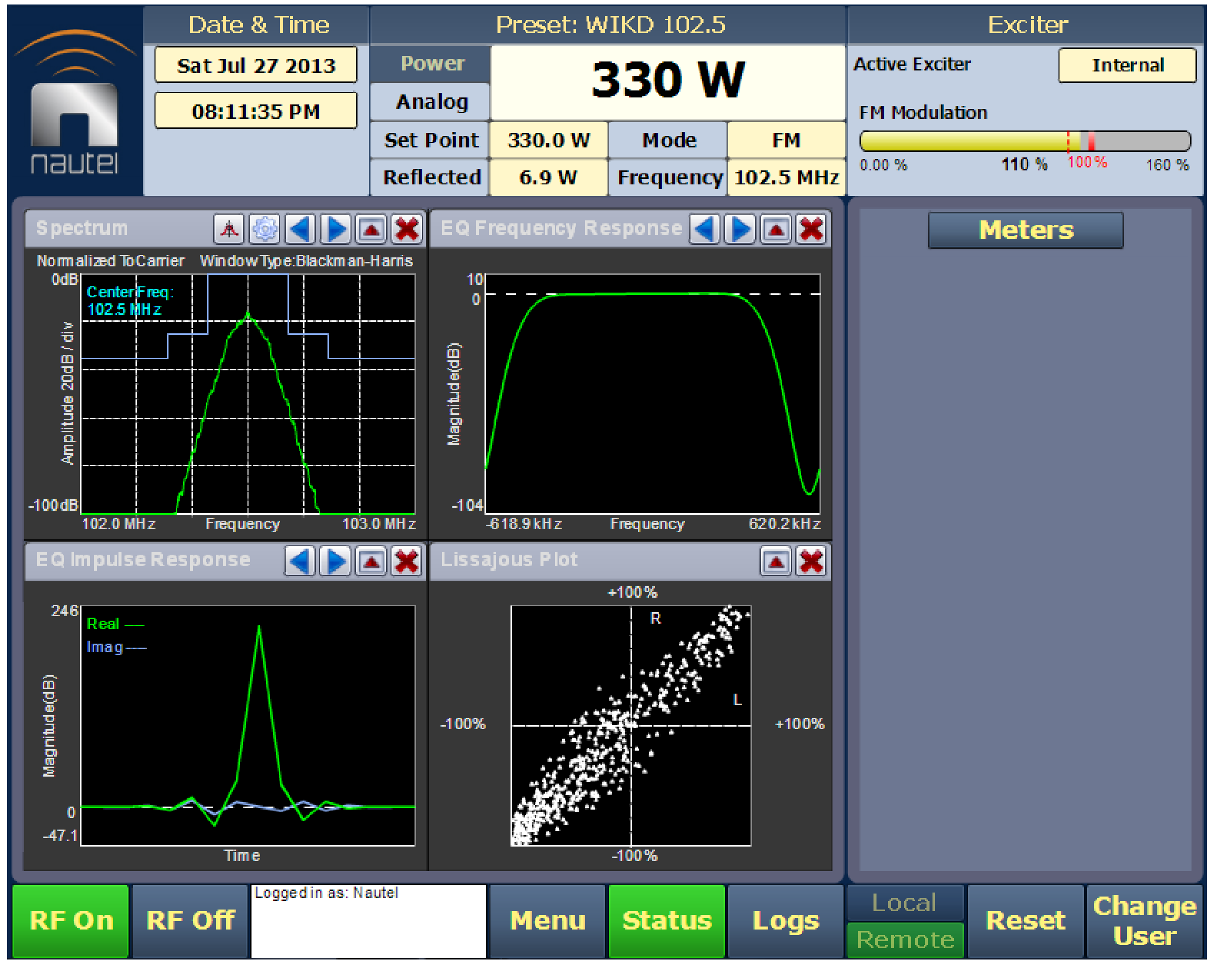
The remote interface for the Omnia also allows you to control every parameter via a web interface. It does require Java to be installed. It is much easier to program the Omnia via the web interface than via the front panel.

The Omnia can be reached from within WIKDNET by going to 10.68.124.19. If you are outside WIKDNET, it can be reached by going to 155.31.216.30:4055.

THE SAGE ENDEC

The Endec's web interface is primarily used to download the EAS Logs. However, it can also be used to listen to the incoming alert audio, as well as view some minor diagnostic information.

The Endec can be reached from within WIKDNET by going to 10.68.124.10. If you are outside WIKDNET, it can be reached by going to 155.31.216.30:10.



THE EAS LOGS

As Chief Operator for WIKD-LP it is your responsibility to print off and verify the EAS Logs each week. This document will guide you through the process.

1. Log into the Sage Digital Endec Unit. From within the studio open up any browser and navigate to the Sage Digital Endec bookmark in the remote control folder.

2. The username and password is admin.

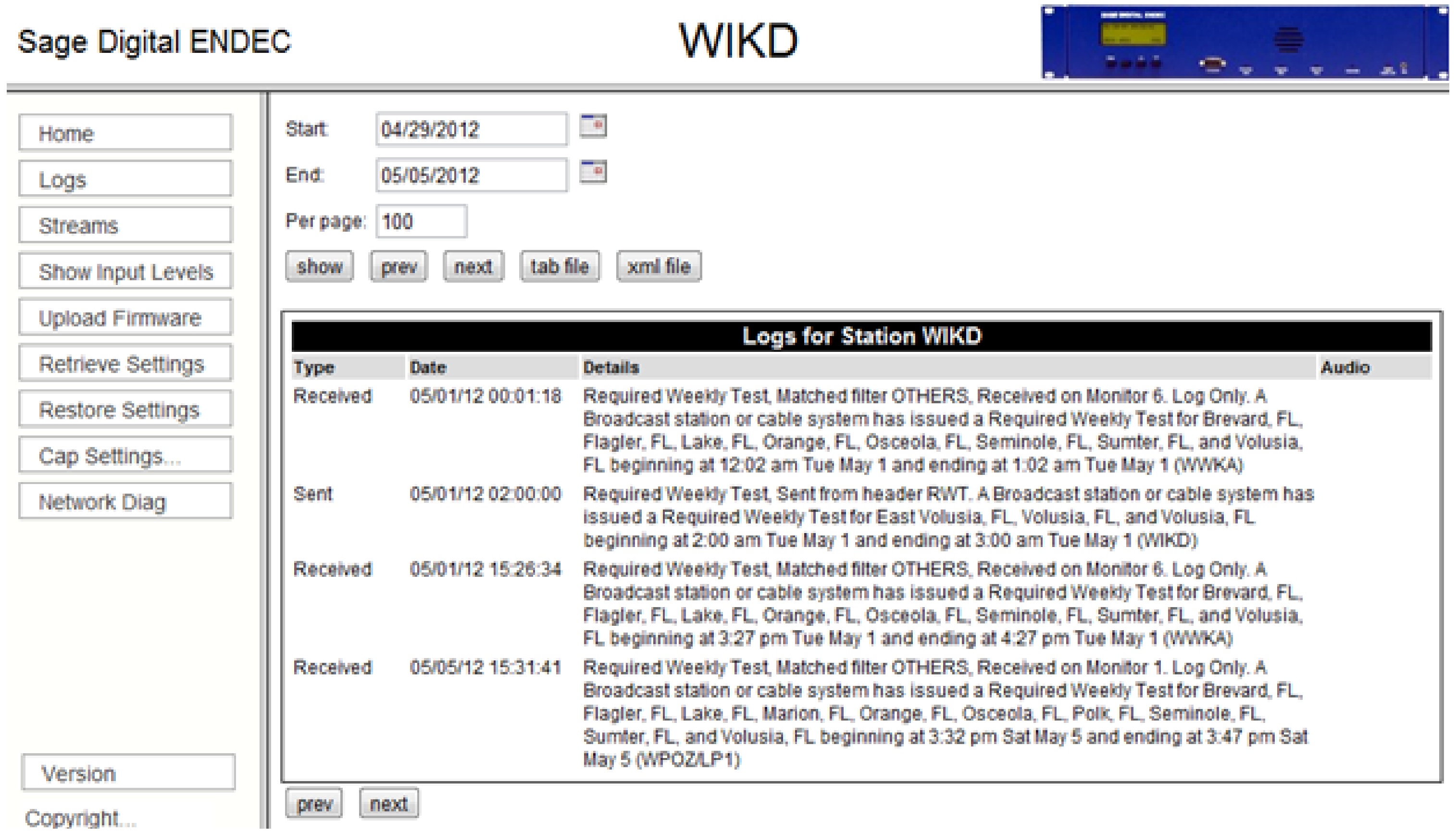
3. Once you are logged into the browser interface click on the button labeled “Logs” located in the left toolbar.

4. Adjust the date range to the current week. Weeks span from Sunday to Saturday. On a normal week without any specific alerts or Monthly tests you should see five entries. One is a log of the test we send, one is a log of the test from our LP1 (88.3MHz WPOZ), one is a log of the test from our LP2 (92.3WWKA), and one is a log of WWKA’s rebroadcast of WPOZ’s weekly test. Lastly, there should be a weekly test from the CAP servers.

5. Click the button that says “xml file” and open the downloaded file in Microsoft Excel. A dialog box will pop open, make sure that “as an XML table” is selected.

6. Press Ctrl+Shift+x. This will run a macro that automatically formats the XML log. Otherwise you will need to delete Columns A&B(dateStart and dateEnd), and word wrap the text in Colum C (details).

7. Copy and paste the logs into the appropriate week of the FCC EAS Logs document. (WIKD-NAS>MUSIC>ENGINEERING>FCC).



REMEMBER!

Every month WPOZ is required to send out a Monthly Test (RMT). WWKA will relay it to us and we are to relay it is as well. If you check the logs and see that we did not receive a RMT from either WPOZ or WWKA, or that we did not send it after receiving it, immediately notify the Chief Engineer and General Manager.

Likewise, if you notice that we did not receive a RWT from WPOZ or WWKA, or that we did not send one, notify the Chief Engineer and General Manager. Once an explanation for the failure has been found you will need to make note of it on that week’s log. (RWT are not required in a week for which an actual alert, such as an Amber alert, or a RMT has been sent).

THE EAS LOGS

In addition to the SAGE Digital Endec, located in the transmitter shed, WIKD employs a Trilithic EASyPlus EAS Unit located in the studio. This EAS unit only monitors WIKD, and its sole purpose is to provide a verification that all alerts were actually aired.

1. To access the Trilithic you will need to be using DJ CPU. Open the Start Menu, and click on the red EASyPlus icon.

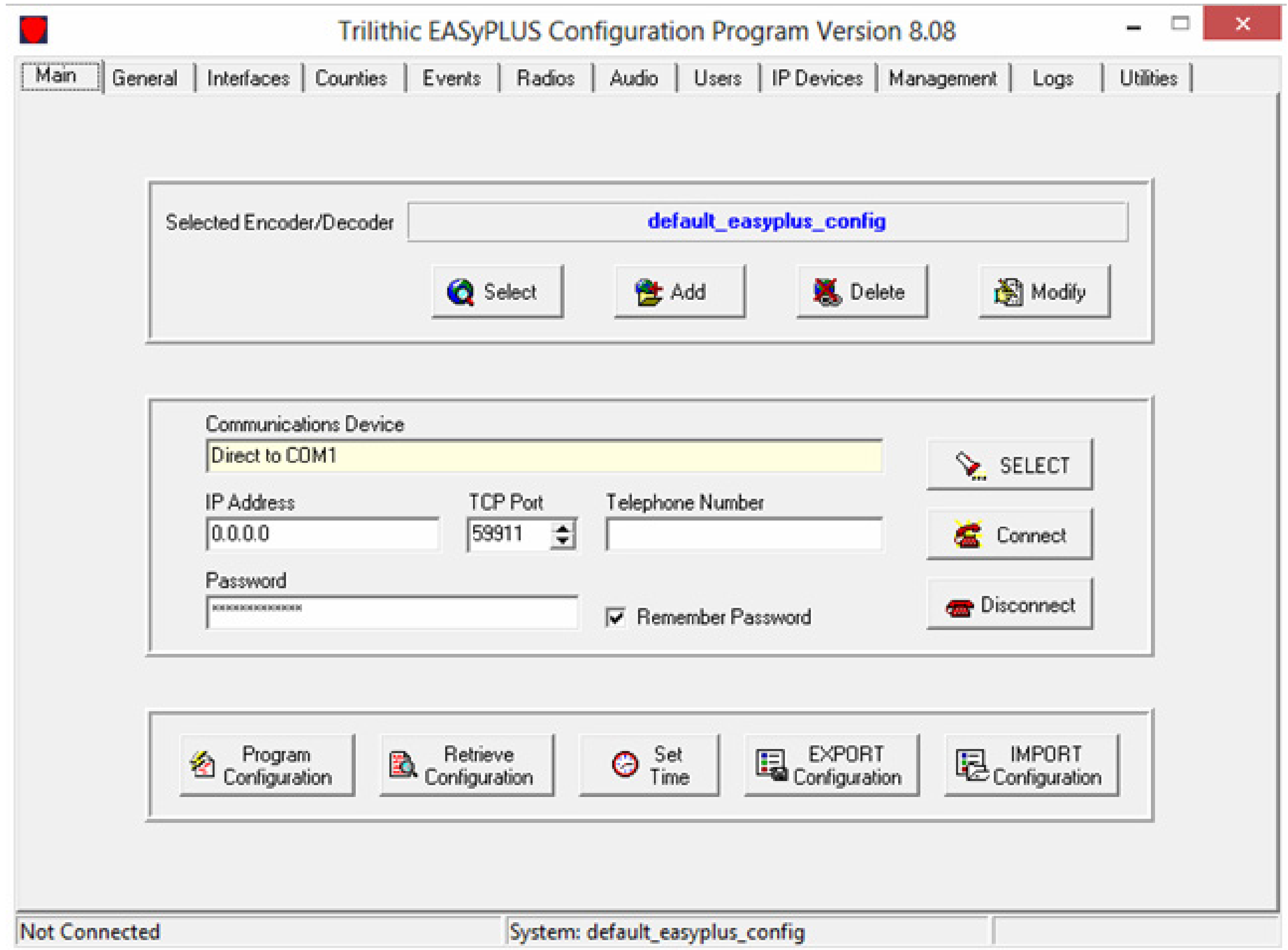
2. Once it has opened, click Connect.

3. Once you have connected to the device, click on the Logs tab.

4. Click the Clear button to erase any old logs. Make sure Word Wrap is checked, and select the Alert RX/TX filter to minimize the amount of logs downloaded. Once everything is in order download and save the logs.

8. Open up the log file and copy and paste the entries for the current week into the FCC EAS Log document immediately following the Sage logs. Make sure they are on their own separate page.

9. Print off the log for that week, sign and date it, and then insert it into the EAS binder. It is good practice to highlight the stations as well as any RMT for easier viewing. See the log book for examples.



OPERATIONAL LOGS

As mentioned earlier, LPFM stations are required to maintain a Station Log which must contain Tower Light logs, EAS Logs, and Operational Logs. The Operational Logs contain records of any station outages due to equipment malfunction, servicing, or replacement, as well as any operation that occurs outside the parameters of the station's license.

Ideally, it will be a rare occasion that you have to make an entry into the operational log. However, ANYTIME the station goes off the air, you must make an entry detailing exactly what happened, and what steps were taken to fix it.

In addition, anytime you install major new equipment, or anything that affects that transmission signal (a new stero generator, EAS encoder, transmitter, etc.) you must make a notation in the log.

Small things matter as well. For example if the PD forgets to include soft syncs in the music log and the legal ID's air a half hour late, then you need to make a notation.

SAMPLE OPERATIONS LOG

WIKD OPERATIONS LOG

INCIDENT REPORT

DATE: JULY 21, 2013

ISSUE:

Needed to add a new device server to AIR1-HOST's configuration, which required shutting down the Workstation interface. Mark shut down automation as soon as the next song began playing (WOAFR will continue playing the current song even if the interface has been closed), made the required changes, and then re-launched automation. However, once the current song finished, automation then proceeded to skip through the remaining events in the playlist while returning a "not authorized error". Mark then shut down automation on AIR1 and launched automation on PROD1. PROD1 worked fine, and was then placed on-air. There was approximately one minute of dead air.

RESOLUTION:

When remotely accessing either AIR1-HOST or PROD1-HOST using either Teamviewer or Remote Desktop Connection they must be configured to continue playing back audio on the remote computer, not the local computer. This will not cause an issue if automation is already running, but if automation is launched during a remote session with the remote computer configured to play audio through the local computer, automation will start failing through songs, generating either a "not authorized error", or a "audio server error".

OPERATOR SIGNATURE:

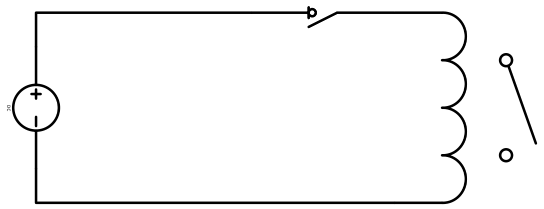
Alexander Reynolds

LOGIC SIGNALS

In the same way that there are audio signals, there are also logic signals. These are primarily used to control equipment. For example, a logic signal might be used to connect a CD player to a mixing console. Whenever the CD player's channel is turned on, the console will send a logic signal to the CD player which starts playing the disc.

The actual sending of logic signals is somewhat complicated, but it all begins (generally) with a simple relay.

A relay is nothing more than an electrically activated switch. A mechanical relay looks something like this:



When the switch at the top is closed, it allows electricity to pass through the inductor (the coil of wire on the right). When the inductor is energized it creates a magnet field that closes the switch on the right.

In a "momentary" relay, the switch on the right is attached to a spring, which keeps the switch in the "open" position. The switch will only be closed for as long as the inductor is energized.

In a "latching" relay, as soon as the switch is moved to the closed position it will stay there until the inductor is energized with the opposite polarity.

Simple enough right?

Let's make it slightly more complicated.

The MIKA armstands in the studio include an "On-Air" light and use a 5 pin XLR style output connector. This connector includes both the audio signal and power for the lamp.

The lamp in the armstands requires between 12 and 24v in order to work, and has a maximum current drain of 40mA.

The Logitek Audio Engine has 15 GPO's, or General Purpose Outputs. GPO's are essentially a fancy way of saying relays. The relays within the Logitek are rated for a maximum of 500mA at 50v, which means they are more than capable of handling the power neccessary for the On-Air lamps.

However, the Logitek Relays cannot supply any power. This is where the Henry SuperRelay comes in. The Henry has a single GPI (General Purpose Input), six GPO's, and one AC Relay. It also includes a 12v power source.

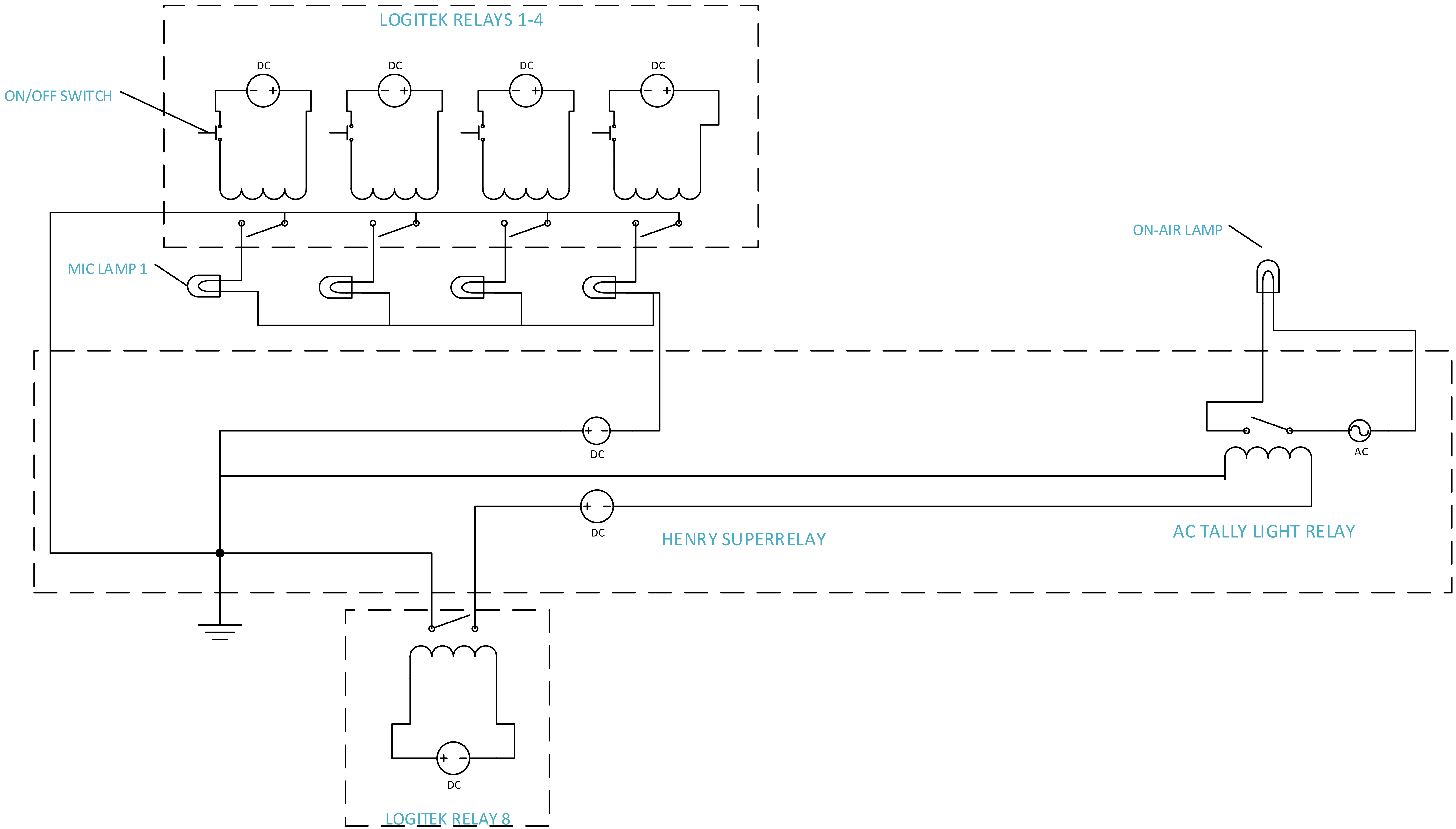
The Logitek is programmed to close Relay 8 anytime a microphone is turned on. Relay 8 is connected to the GPI on the Henry. When Relay 8 is closed it causes the Henry to closes all of its relays. This results in power being supplied to the AC Socket which the On-Air lamps outside the studio and office are connected to.

At the same time, the Logitek relay associated with the specific microphone closes and completes the circuit between the 12v power supply and the microphone lamp. In order for power to pass from the Henry to the Mic Lamps, the relay on the Logitek side must be closed.

This way, anytime any microphone is turned on, the On-Air lamps outside the studio and office will be turned on, but only the lamp associated with the microphone will turn off.

Slightly more complicated right?

It's cool. There's a picture on the next page. ☺


LOGITEK RELAYS 1-4
DC
DC
DC
DC
ON/OFF SWITCH
MIC LAMP 1
ON-AIR LAMP
DC
AC
DC
HENRY SUPERRELAY
AC TALLY LIGHT RELAY
DC
LOGITEK RELAY 8

NETWORKING
PART 11
BASICS

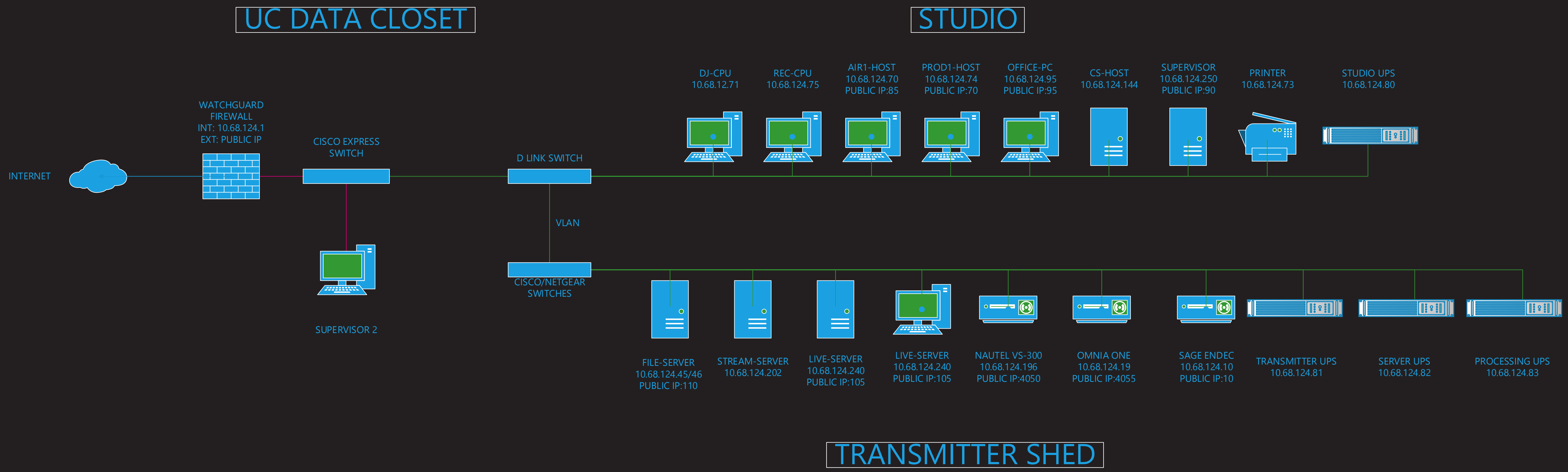
Yeah. There's not much here. Write it your own damn self.

THE
B A S I C S

If you don’t already have a very basic understanding of how computer networking, then I would suggest going through Juniper’s free web course on Networking Fundamentals, which you can find here. At a minimum, you should have an understanding of what IP and MAC addresses are, as well as a rough idea of what firewalls, routers, and switches do.

This part of the guide is really a glorified glossary of the equipment and software we use at WIKD. If you want to learn more, then I suggest looking into getting a Cisco Networking certification, such as the CCENT (Entry Networking Technician).

Below is a diagram of the WIKD Network, including all computers and devices.



THE GLOSSARY

DHCP: DYNAMIC HOST CONFIGURATION PROTOCOL
A protocol used to dynamically assign IP addresses to computers and devices. LIVE-SERVER hosts a DHCP server for WIKD. It will assign IP addresses in the range of 10.68.124.70 to 10.68.124.90.

DNS: DOMAIN NAME SYSTEM
A protocol which translates hostnames (www.google.com) into IP addresses (74.125.137.138). LIVE-SERVER hosts a DNS server for WIKD. We use an alternate DNS of 4.2.2.2.

FIREWALL
A device that controls the flow of packets between an exterior network (usually the internet), and an internal network. It can restrict the flow of packets based on which ports they wish to access. WIKD uses a Watchguard X750e firewall that also serves as our default gateway. In addition to performing standard firewalling, it also performs Port Address Translations that allow access from outside WIKD to computers and devices within WIKD.

FTP: FILE TRANSFER PROTOCOL
A network protocol which is used to transfer files over the internet. WIKD hosts an FTP Server on FILE-SERVER using Bulletproof FTP. This allows you to access files on the Engineering, NAS, Music, and Operations directories from anywhere in the world.

IPv4 ADDRESS
A 32 bit address assigned to computers and devices of the form: xxx.xxx.xxx.xxx, where x is a number between 0 and 255. All computers and devices in WIKD have an IPv4 address that begins with 10.68.124.

PAT: PORT ADDRESS TRANSLATION
PAT means that instead of having to have a separate public IP address for each computer or device within WIKD, we can simply assign them a specific port of the public address. Based on the configuration of the Watchguard, the firewall can then route data sent to a specific port to a specific device within WIKD.
For example, WIKD employs a public IP address of 155.31.216.30. If you access port 8000 (155.31.216.30:8000), the Watchguard firewall will direct you to the Shoutcast v2 server that is hosted on LIVE-SERVER. The internal address of the Shoutcast server is 10.68.124.240:8000.

PRIVATE IP ADDRESS
An address which has been set assigned to only be used in networks that are not connected to the internet, or perform some form of Network Address Translation. The following are private IP Address spaces:
10.0.0.0 – 10.255.255.255
172.16.0.0 – 172.31.255.255
192.168.0.0 – 192.168.255.255
WIKD employs 10.68.124.0 – 10.68.124.255 as our private IP address space.

PoE: POWER OVER ETHERNET
A standard which allows power to be passed over Ethernet cabling. PoE is how the Avaya VoIP phones receive power without needing separate power adapters. PoE does require specialized equipment to implement. The Cisco 3560 switch in the transmitter shed is a PoE enabled switch, which is why we can use an Avaya VoIP phone there.

PUBLIC IP ADDRESS
An address which can be reached on the internet. These are typically expensive and hard to get. WIKD employs a public IP address of 155.31.216.30.

AUTOMATION

PART 12

TROUBLESHOOTING

AND OTHER

HELPFUL NUMBERS

THE WIDE ORBIT SUPPORT HELP LINE

(214) 451-4200

MEMORIZE THE NUMBER.
THEY'RE NICE PEOPLE.

AND HERE'S A LINK TO THE MANUAL.
READ IT.
IT'S HELPFUL.(AND EXCELLENT)

USEFUL CONTACTS

EATON
1 (800) 356-5737

LOGITEK
1 (877) 231-5870

LYNX
(714) 545-4700 x206

NAUTEL
1 (877) 628-8353

OMNIA
(216) 241-7225

RANE
(425) 355-6000

RME
(954) 626-0674

ROLLS
(801) 263-9053

RVR
(305) 471-9091

SAGE
(914) 872-4069

SPINITRON
(617) 233-3115

WIDEORBIT
(214) 451-4200

JIM HOGE
WPOZ ORLANDO
FLORIDA EAS CHAIRMAN
(407) 869-8000
jim.hoge@zradio.org

KIM SILVA
FCC ENFORCEMENT BUREAU
TAMPA FIELD OFFICE
(813) 348-1602

FLORIDA ASSOCIATION OF BROADCASTERS
(850) 681-6444

BROADCASTERS GENERAL STORE
(352) 622-7700

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** Due to a series of unfortunate events (PTEK Transmitters suck) WIKD has been banned from calling PTEK Technical Support.*

Not that they were going to help you anyways...