

# **AIR AURA X1**

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## **FM & HD DIGITAL AUDIO PROCESSOR**

### **TECHNICAL MANUAL**

 *Wheatstone Corporation*

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600 Industrial Drive, New Bern, North Carolina 28562 (tel 252-638-7000 / fax 252-637-1285 )

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### TECHNICAL MANUAL

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Wheatstone Corporation  
December 2016



WHEATSTONE AUDIO PROCESSING

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# AirAura X1 Technical Manual

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## General Information

### Introduction

Welcome to the Wheatstone X1 FM and HD Digital Audio Processor, an exciting part of Wheatstone's third generation of audio processing. The X1 expands on technologies previously invented by Wheatstone, incorporates several new technologies, and delivers that "sought after" sound to the FM and HD mediums.

The X1's processing algorithms have evolved from our highly popular FM-55 broadcast audio processor. The X1 package builds on the great sound and simple setup of the FM-55 and adds a second processing path for HD as well as an insert point for a ratings encoder.

The X1 is the first part of a brand new approach to processing from Wheatstone that we call UNIFIED PROCESSING. Unified Processing means the entire processor is self-aware of all the processing stages, how they are working, and what decisions the individual blocks have to make to generate a unique and consistent sound from source to source, no matter what the dynamic range or level.

The X1's unified algorithms include a brand new program aware 10-band limiter (never before offered at this level in any broadcast processor) that takes the effortless sound of the FM-55 platform to a new level. Coupled with our proven iAGC program density controller, specialized low distortion, low IMD clipper for FM and codec aware HD processing, the X1 stands above all other processors in its price class.

Remote control of X1 is via Windows-based Graphical User Interface (GUI) software and can be accomplished via 100BaseT wired Ethernet. The X1 GUI incorporates two layers of controls... one for expert users and one for basic users. This means that even inexperienced users can have confidence in setting up and adjusting the processor.

Eight rear panel General Purpose Inputs (GPI) and four General Purpose Outputs (GPO) are provided to allow hardware-level interfacing of the X1 to station systems.

The X1 provides the user with the ability to insert the ratings encoder and any associated gear directly into the processing path. This wraps the encoder around the audio processor to potentially increase the chances of ratings credit. The front end of the X1 thereby acts as a program density controller for the ratings encoder, ensuring that there is always ample program level to embed the watermark. Previously, one had to place the ratings encoder BEFORE the audio processor and run the risk of the multiband section of the audio processing skewing the watermark tones. With the X1, placement is AFTER the bulk of the multiband section, leaving the watermark intact and with the potential for higher recognition by the meter.

The X1 utilizes an internal base sampling rate of 192 kHz. Far higher sample rates are used where it is beneficial to do so, such as within the X1 clipper. A 24-bit, 144dB internal dynamic range signal path allows audio to be processed with extreme precision. The single front panel touchscreen and control knob allows users easy operation of most key controls to set up and adjust the processor.

All Wheatstone products are designed and manufactured by Wheatstone in its Corporate Headquarters in New Bern, North Carolina, USA. This large, state of the art facility affords complete control over product quality during every step of the manufacturing process. In order to maintain the high product quality standards that Wheatstone is known for for over 40 years, no offshore manufacturing is utilized.

If during installation or setup you find that you need assistance or advice, please feel free to contact our technical support folks at (252) 638-7000 or email us at: [techsupport@wheatstone.com](mailto:techsupport@wheatstone.com).

## X1 Feature Overview

The X1 digital audio processor has been designed to separately process audio for conventional analog FM and digital HD radio signals. Sharing the four-band parametric equalizer and intelligent five band leveler with iAGC, the FM and HD paths are equipped with individual “program aware” 10-band limiter sections and peak controlling processors which have been optimized for the respective broadcast mediums.

The X1’s five band leveler incorporates Wheatstone’s specialized iAGC (IntelligentAGC) algorithm. and operates through a proprietary technique called Density Compensation™. The Density Compensation™ algorithm utilizes real-time analysis of the pre and post-processed audio, and data from the processing controls as the user has adjusted them, to invisibly manage the user set spectral balance and program density. The output of the five band leveler can then be patched to an external ratings encoder device. Upon return to the processor, the audio is then split into two signal paths, FM and HD, each equipped with our 10-band “program aware” limiter.

The FM signal path is equipped with a newly refined FM clipper. Borrowing a little from our FM-55 audio processor, this new FM clipper works with the 10-band limiter to automatically determine the best course of action to maintain loudness and clarity. The result is a higher average level of audio without the traditional tradeoffs you would find in other designs that were “trying” to be loud.

A digitally perfect stereo generator follows the clipper and provides two multiplex composite stereo outputs and two SCA inputs.

The X1 is also fully compatible with exciters that accept composite audio delivered over an AES connection. Wheatstone’s baseband192 system includes audio and subcarriers up to and including 67kHz in an AES format. This system eliminates the need to transition back to analog when using the built in stereo generator in your audio processor. It also maintains a complete digital path without sacrificing loudness or having to use the stereo generator built into the exciter.

The HD signal path is equipped with its own 10-band limiter followed by a smart look-ahead final limiter with adjustable operating parameters which allow program density and peak control to be set as desired. The HD path’s digital and analog outputs may be swapped left for right as well as phase reversed if needed.

The X1 is equipped with Wheatstone's network-based ACI protocol, allowing complete control of not only presets, but any parameter that is assigned to a user control (please contact the factory for details on this). The ACI also allows the X1 to receive time alignment commands from external devices to correct for alignment errors in the processor. For more information on HD time alignment products that are compatible with the X1, please visit our website at <http://www.wheatstone-processing.com>.

Dozens of factory presets are provided with X1 making a wide range of on air sounds available to the user. A fully routable headphone monitoring path allows the audio in various parts of the processing chain to be auditioned.

The System Menu of the GUI provides access to a pair of flexible preset scheduling utilities. As well as the usual short-term scheduler, the X1 is equipped with a "long-form" scheduler feature that can be programmed for automatic preset changes well into the future. This scheduler is useful for changing presets weeks or months after a preset is prepared, such as for a special event, and without having to edit the daily preset schedule in order to accommodate these "special" presets.

GPI (General Purpose Inputs) provide easy access to preset changes via external events such as contact closures and automation control.

## Rack Mounting

The X1 is designed to be mounted into an industry standard 19" equipment rack and requires three rack units (5.25 inches / 13.335cm) of vertical space. If using only two rack screws always use the bottom two screws to prevent twisting of the front panel and other undue forces from harming the processor chassis.

The X1 does not need nor does it have top or bottom cover ventilation holes. Cooling is accomplished via cooler air drawn into vertical slots positioned lower in the side panels which allows latent heat rising by natural convection to exit slots in the top of the rear panel.

A small fan is located inside the unit which stirs the air to even out the internal temperature. This fan has an 80,000 hour rated minimum lifetime and has no role in "removing" heat. In the unlikely event that the fan fails, its failure will not compromise the operation of the X1 as long as the unit is being operated in a reasonable environment (below 104F or 40C).

The X1 may be mounted between other devices in the equipment rack; however, in accordance with good engineering practice it should not be mounted directly above devices that generate a significant amount of heat (such as power amplifiers or power supplies). If such a location is unavoidable, then it is advisable to utilize an extra 1RU blank rack panel between the X1 and devices immediately above and/or below it.

### **WARNING!**

*With very few exceptions, the X1 chassis does not need to be opened in the field.*

*Please be advised that the unit contains high voltage power supply circuits operating at voltages well above AC line potential.*

*These voltages are not only hazardous but also potentially deadly if accidentally contacted.*

*There are no user-serviceable parts inside the unit and it should be returned to Wheatstone Corporation under a Return Authorization should repair ever become necessary.*

## X1 Installation Tips

### Grounding

Establish a low impedance common ground in the facility and try to route all equipment grounds to that point, using conductors with the largest possible surface area while keeping those leads as short as possible. The X1's ground reference (its chassis) should be connected to the station ground. Such a connection is especially important when the X1 is operated in a high RF environment because it helps minimize differential voltages between the processor's chassis and other pieces of equipment.

### Surge Protection

Always place surge protection circuits as close as possible to the device being protected. AC power line surges should be handled in a way that keeps instantaneous potential differences *between* the power line hot, neutral, AC grounding conductor, the station ground and the processor chassis as low as possible. Likewise, measures should also be taken to keep the instantaneous potential difference between the audio cable shields and the processor chassis as low as possible (this applies to *all* audio equipment, not just the X1), particularly when the equipment is located within the electrically hostile environment of a station's transmitter facility.

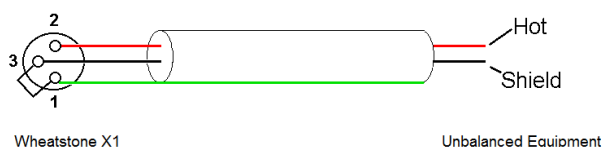
### UPS/Power Conditioning

Choose the best power conditioning/UPS units that your budget will allow, focusing on the most important features and options that you actually need. Some questions to ask while reviewing features are:

- How does the unit handle AC power that is not exactly 60Hz, such as when the facility is on its backup generator?
- If the unit has onboard surge protection, what kind of surge capability does it have and where are those surges directed to?
- Is the unit equipped with remote monitoring capability?
- Does it have onboard monitoring and alarms to signal problems such as batteries with low reserve?

### Analog Audio Input Connections

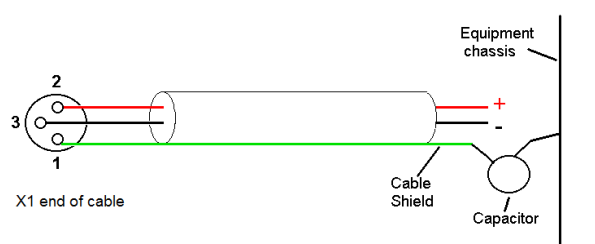
Balanced audio *input* sources should be connected to the X1 using standard two-conductor shielded audio cable such as Belden 8451 or 9451.



*Unbalanced* input audio connections should be made with shielded *two conductor* cable. At the unbalanced source's output connect its "+" output to the X1's "Hi" input (XLR Pin 2) and connect the shield wire to X1's "Lo" input (XLR Pin 3). If the cable's shield

is used (recommended) connect it at the X1 end only (XLR Pin 1) to prevent AC ground loops.





If RF interference is an issue, the far (floating and ungrounded) end of the cable shield can be experimentally bypassed to RF ground via a 0.01 $\mu$ F, 250V AC *rated* capacitor. Suitable capacitors are:

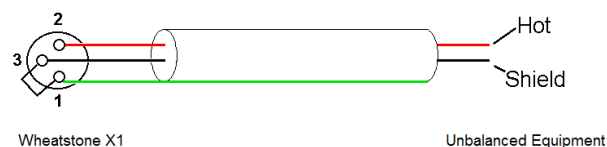
TDK	CS17-F2GA103MYGS
Murata	DE2F3KH103MA3B
Panasonic	ECK-ATS103MF
AVX	65N103MBLCP

## Analog Audio Output Connections

Balanced audio *loads* should be connected to the X1's outputs using standard two-conductor shielded audio cable. *Unbalanced* audio loads should be avoided, but if they can't they should be connected using shielded *two conductor* cable such as Belden 8451 or 9451 (as if connecting a balanced source).

The X1 is equipped with an active balanced output stages that behave like a transformer. Because of this behavior the wiring may be different than expected.

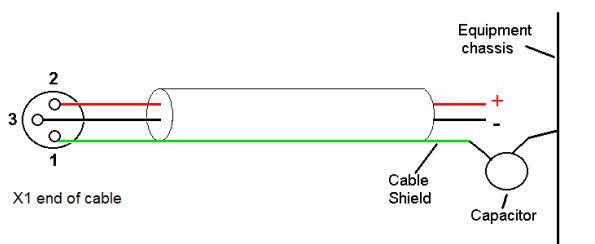
The unbalanced load's "Hi" lead should be connected to XLR connector Pin 2 ("Hi").



The unbalanced load's shield should be connected to the X1's output XLR Pin 3 ("Lo"). Then, and to enable the X1's balanced output amplifier to operate correctly when driving the unbalanced load, it is recommended that the X1's XLR output Pin 1 (ground/shield)

also be connected to Pin 3 ("Lo"), noting that this **MUST BE DONE** at the X1 output connector (see the diagram above).

Note that it is *not* advisable to connect Pin 1 and Pin 3 conductors together at the far end of the cable as doing so can induce external noise and crosstalk on the output amplifier's "load sense" lead which is XLR Pin 3 when Pin 2 is being used as the "Hot."



As in the input case, if RF interference is an issue the far (floating) end of the cable shield can be experimentally bypassed to RF ground through a 0.01 $\mu$ F 250V AC *rated* capacitor to see if it helps. Please refer to the listing above for capacitors known to be effective in this task.

## Digital Audio Connections

For digital audio connections always use a good quality digital audio cable (or twisted pair Category 5E/6 Network cable) having a characteristic impedance of 110 ohms. This cable should be shielded where possible, and in the case of multi-pair cable, each pair should be individually shielded. Foil shielding is recommended for permanent installations, and a cable with foil shield plus an overall braid should be used in applications where frequent flexing of cables might occur.



Generic “audio” cable such as Belden 8451 and 9451 *may* sometimes be used for interconnecting AES3 digital audio devices as long as the cable is short. The actual cable length that will work satisfactorily is determined by *many* factors and may include the error correction and jitter tolerance of the AES3 receiver, the characteristics of the digital cable driver, and the characteristics of the specific cable being used and its length. “Generic” analog audio cables typically has much higher capacitance than digital cable, which can slow down the rise time of digital data signals impairing the ability of the AES3 receiver to accurately recover the digital signal without errors. This can result in increased jitter, dropouts, or at the extreme, no audio at all.

## Where to Install the X1

The best location to install the X1 is at the transmitter site. This requires that a discrete Left/Right STL, either analog or digital, be involved in the signal path. The major benefit of a transmitter site installation is that it enables the use of the X1’s built-in lab-grade stereo encoder which allows much tighter control of modulation peaks.

A transmitter site location for the processor has the additional benefit of allowing the use of the X1’s highly oversampled composite processor for an additional loudness advantage. The Wheatstone composite clipper algorithm is much cleaner and more forgiving than those in other products, and in combination with the X1’s tight pilot and SCA protection filters can create additional loudness without the audible grunge that traditional composite clipper designs typically create.

We recommend that whenever there is a choice between using the AES output of the processor or the composite output to feed your exciter that the composite output should be chosen to maximize loudness, peak control and reap the benefit of the purpose built stereo generator in the processor over the “add on” stereo generator in the exciter. While the exciter’s AES3 digital input may be “clean” and it may be “digital”, it also precludes the ability to gain additional loudness through the use of X1’s intelligent oversampled composite clipper. Also, depending on factors including the sample rates being used the exciter’s AES digital input can exhibit inferior peak control compared to the exciter’s composite input.

When X1 is located at the studio and an STL is being used to send the program material to the transmitter site there are several issues to consider:

### Digital STL

There are two categories of Digital STL’s on the market – those with codec-based audio compression and those using uncompressed linear audio.

When the digital STL employing codec-based audio compression is being used, X1 should be located at the transmitter site which places it *after* the codec. The reason for this is that most codecs will sound better when presented with *unprocessed* studio audio instead of highly processed and pre-emphasized audio from the processor’s output. Further, the encoding schemes used in such STL’s cannot accurately pass the well-defined peak levels created by X1 which can create a modulation (loudness) disadvantage.

Installing X1 at the studio end of a “compressed” STL brings with it at least two caveats:

- The stereo generator and composite clipper inside X1 will not be available. Many digital exciters offer stereo generator and composite clipper functions, but their clippers have historically been quite crude compared to X1’s exceptional clipper.

Therefore, “exciter-hosted” composite clippers are *not* the optimum choice when the station’s ultimate sound *quality* is important.

- Compressed STL’s do not perform well when presented with competitively processed audio, especially when that audio is pre-emphasized. The reason for this is because codecs do their work by examining the audio for opportunities to *remove* content that *shouldn’t* be audible to the average human ear. When densely processed audio is presented to a codec there are fewer “opportunities” for it to remove redundant audio information and most importantly, then *mask* that removal from our hearing so that we don’t notice it. When handling heavily processed (limited dynamic range) material, codec operation is usually much more obvious – even to the point of being objectionable – than when the processing is located *after* the codec where the “masked” artifacts are only occasionally and, usually, minimally unmasked by “processing gain”.

Uncompressed (linear) digital STL’s have only one major limitation – placing X1 at the studio end of the STL will preclude the use of X1’s stereo generator and composite clipper.

**TIP:** If using X1 at the studio be certain that any clippers in the stereo generator at the transmitter site are properly set up to complement the settings in X1. This will prevent gross distortion and potentially large modulation overshoots.

Pre-emphasis should always be applied by the audio processing, not the exciter. Modern FM audio processors have highly sophisticated technology to deal with the challenges presented by FM pre-emphasis curve and can provide very tight modulation control with very low perceived distortion. FM exciters do not have this technology.

The best location for the audio processor from an overall performance standpoint is always at the transmitter.

## Analog Left/Right STL

Older analog discrete left/right STL’s can suffer from an inability to control audio peaks because of inadequate bandwidth in their IF circuits and/or poor low frequency and phase performance. Individual left/right STL’s rarely have identical group delay and this will adversely affect stereo separation when the signal is finally converted to the multiplex composite domain. Such STL’s can also suffer from AFC bounce when handling highly processed low frequency material and this can rob modulation capability and therefore reduce on-air loudness.

## Composite Analog STL

A high quality analog composite STL can advantages over an analog left/right STL and typically have broader audio bandwidth and better audio performance than analog discrete STL’s. Most even have the capability to also piggyback subcarriers such as SCA and RDS along with the composite audio. This means that most SCA and RDS generators may also be located at the studio end of the STL, which, along with the audio processor, makes for a very convenient setup. With a modern composite STL and properly engineered point-to-point path, the audio can be nearly as transparent as a digital STL.

## Analog Phone Lines

Discrete left/right analog “phone line” STL’s are not recommended because of the inability of most Telco service providers to meet the flat frequency response and phase matching requirements of the pair of audio circuits. Furthermore, in many countries wide-band analog circuits have become unavailable or their cost is prohibitive. On the other hand if the “wired” STL is a dedicated (and equalized if necessary) pair of circuits that is under the station’s full control it may be acceptable.

## Arbitron People Meter (PPM)<sup>™</sup>

Field experience has been that Wheatstone processing algorithms favorably pass the data watermarking scheme used in the Arbitron People Meter rating service technology, regardless of the aggressiveness of the processing being performed. The X1 algorithms are no exception. The X1 hardware also adds the additional benefit of inserting the PPM watermark and associated hardware towards the back of the processing for both the HD and FM audio paths. This eliminates the need of having an external AGC to feed consistent audio to the PPM encoder. You can now use the internal AGC in the X1 to do this while allowing the PPM encoder to insert the watermark AFTER most of the processing.

## AC Power Considerations

Please note that in order to enhance its long-term reliability X1 has no power switch (switches usually become intermittent over time without regular use).

X1 accepts AC line input voltages between 90 and 260 VAC, 50 or 60Hz. Power consumption is under 100VA.

Although aggressive AC input filtering is utilized on the AC power input it is always advisable to use external surge protection and an uninterruptible power supply (UPS) wherever possible, especially where the AC power quality can be in question, such as at a remote transmitter site.

Power conditioning, surge suppression, and even power backup devices are wise investments when using sensitive modern electronic devices. X1 is, after all, a highly specialized “computer.”

The use of a UPS (uninterruptible power supply) is usually recommended and will protect X1 from short duration power interruptions and glitches which might otherwise signal it to reboot. When X1 reboots there will be a loss of audio for approximately 15 seconds.

## Rear Panel Connections

The image below shows the rear panel of the X1 and the location of various connectors associated with an installation:



AirAura X1's rear panel connectors from left to right are:

### Bottom Row

Analog Left Channel In  
 Analog Right Channel In  
 AES Insert Loop Output (Send)  
 AES Insert Loop Input (Return)  
 FM/HD Analog Left Channel Out  
 FM/HD Analog Right Channel Out  
 AES Digital Input  
 FM AES Digital Out (L/R or baseband192)  
 HD AES Digital Out  
 GPI Input  
 Ethernet/WheatNet-IP Interface  
 IEC Standard male AC Power Input

### Top Row

SCA-1 Input  
 SCA-2 Input  
 TX 1 Out  
 TX 2 Out  
 GPO Status Output

The listed connections are made via various connectors mounted on the X1's rear panel.

- Nine XLR connectors are provided for analog and digital audio input and output connections, plus an insert loop for ratings encoders.
- Four BNC connectors are provided for SCA inputs and TX (transmitter) out, or Composite Multiplex (MPX) connections.
- One RJ-45 connector is provided for Ethernet connections. This connector may be used to interface X1 to a WheatNet-IP audio network or to connect it to a Windows® PC running the Wheatstone X1 GUI.
- Two DB-9 connectors provide 8 GPI inputs (for selecting presets in the first eight storage slots) and four pre-assigned GPO outputs.

The pinout drawings on pages 1-17 through 1-20 summarize wiring connections.

## Audio Inputs

The X1 accepts three types of audio input sources (excluding the external loop):

- Balanced analog line level left/right audio;
- Digital AES3-compliant left/right audio with sample rates between 32kHz and 96kHz;
- WheatNet-IP via 100BaseT Ethernet connection to a WheatNet-IP audio network.

Input audio can be applied to any or all inputs simultaneously with the caveat that the WheatNet-IP input and AES3 inputs share a *common* digital path into the internal processing.

Automatic audio failover from analog to digital or vice versa is supported. Automatic failover from AES3 or WheatNet-IP to analog is instantaneous and based on invalid or missing bits in the AES3 stream, or after 30 seconds of “silence” (level below -48dBFS).

Automatic failover from analog to AES3 or WheatNet-IP is based on silence sense responding to audio on both channels being below -48dBFS for more than 30 seconds.

Failover capability is *not available* between the AES and WheatNet-IP inputs.

### Analog In – XLR-F

Pin 1 XLR LT SH – LINE LT IN SH  
Pin 2 XLR LT HI – LINE LT IN HI  
Pin 3 XLR LT LO – LINE LT IN LO

Pin 1 XLR RT SH – LINE RT IN SH  
Pin 2 XLR RT HI – LINE RT IN HI  
Pin 3 XLR RT LO – LINE RT IN LO

### AES In – XLR-F

Pin 1 XLR SH – AES IN SH  
Pin 2 XLR HI – AES IN HI  
Pin 3 XLR LO – AES IN LO

### SCA In – BNC

Pin 1 BNC 1 HI - SCA 1 IN HI  
Pin 2 BNC 1 SH - SCA 1 IN SH

Pin 1 BNC 2 HI - SCA 2 IN HI  
Pin 2 BNC 2 SH - SCA 2 IN SH

## Audio Outputs

### FM Path

Output audio for the FM path is available as:

- Balanced analog left/right stereo, pre-emphasized via the switchable FM/HD analog output.
- Balanced analog left/right stereo, de-emphasized according to pre-emphasis in use via the switchable FM/HD analog output.
- AES3 digital, either pre or post diversity delay, and/or de-emphasized according to pre-emphasis in use via the switchable FM AES output jack.

- Baseband192 AES composite signal via the switchable FM AES output jack (must interface with an FM exciter capable of receiving the baseband192 signal).
- Unbalanced composite stereo on two BNC female connectors.
- Left/Right stereo via the WheatNet-IP audio network.

## HD Path

Output audio for the HD path is available as:

- AES3 digital left/right stereo via the HD AES output jack.
- Left/Right stereo via the WheatNet-IP audio network.

## Insert Loop Out – XLR-M

Pin 1 XLR INSERT LOOP OUT SH – INSERT LOOP OUT SH  
 Pin 2 XLR BAL LOOP OUT HI – INSERT LOOP OUT HI  
 Pin 3 XLR BAL LOOP OUT LO – INSERT LOOP OUT LO

## Insert Loop In – XLR-F

Pin 1 XLR INSERT LOOP IN SH – INSERT LOOP IN SH  
 Pin 2 XLR BAL LOOP IN HI – INSERT LOOP IN HI  
 Pin 3 XLR BAL LOOP IN LO – INSERT LOOP IN LO

## FM/HD Analog Out – XLR-M

Pin 1 XLR FM/HD ANALOG OUT LT SH – FM/HD ANALOG LT OUT SH  
 Pin 2 XLR FM/HD HI – FM/HD ANALOG LT OUT HI  
 Pin 3 XLR FM/HD LO – FM/HD ANALOG LT OUT LO

Pin 1 XLR FM/HD ANALOG RT SH – FM/HD ANALOG RT OUT SH  
 Pin 2 XLR FM/HD ANALOG RT HI – FM/HD ANALOG RT OUT HI  
 Pin 3 XLR FM/HD ANALOG RT LO – FM/HD ANALOG RT OUT LO

## FM AES / baseband192 Out – XLR-M

Pin 1 XLR FM SH – AES FM / BASEBAND192 OUT SH  
 Pin 2 XLR FM HI – AES FM / BASEBAND192 OUT HI  
 Pin 3 XLR FM LO – AES FM / BASEBAND192 OUT LO

## HD AES Out – XLR-M

Pin 1 XLR HD SH – AES HD OUT SH  
 Pin 2 XLR HD HI – AES HD OUT HI  
 Pin 3 XLR HD LO – AES HD OUT LO

## TX Out – BNC

Pin 1 BNC 1 HI – TX 1 OUT HI  
 Pin 2 BNC 1 SH – TX 1 OUT SH

Pin 1 BNC 2 HI – TX 2 OUT HI  
 Pin 2 BNC 2 SH – TX 2 OUT SH

## Headphone Monitoring

An overload protected stereo headphone amplifier drives the ¼" stereo headphone jack located on the right side of the X1 front panel. The audio source feeding the headphones



may be chosen from several signal points within the processing algorithms, including both analog and digital inputs, even if those inputs have *not* been selected to feed the audio processing chain. The System menu of the GUI hosts the headphone router selector. The front panel volume knob above the jack controls the audio level.

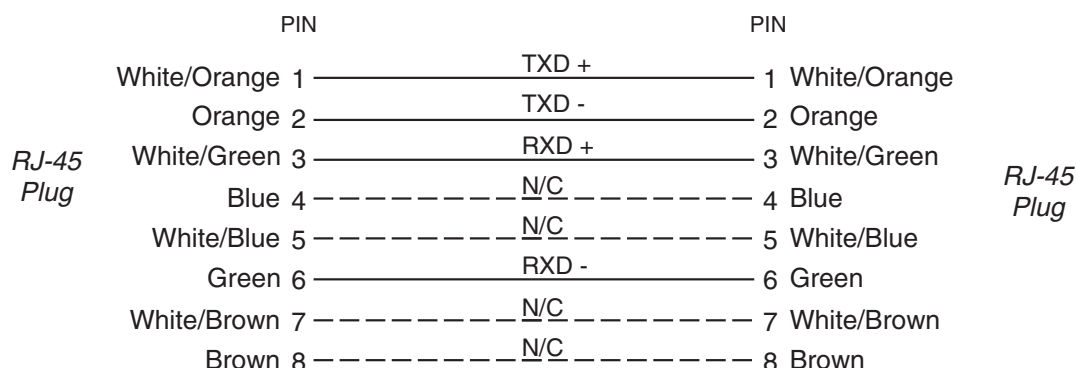
## Network Connections

X1 is equipped with a wired Ethernet connection via the 100Base-T Ethernet port on the rear panel. The port is completely independent and Auto-MDIX allowing them to support straight through and crossover cables in any combination. The wired Ethernet interface can support up to four simultaneous connections to remote GUI.

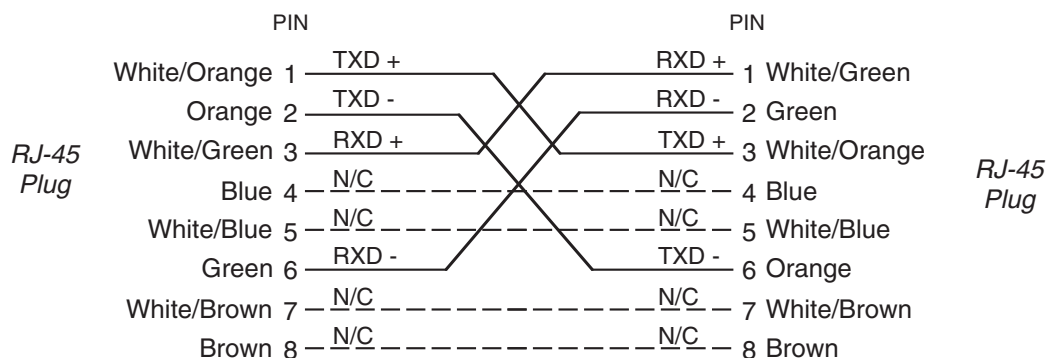
### Ethernet – RJ-45

Pin 1 – TXD +  
 Pin 2 – TXD -  
 Pin 3 – RXD +  
 Pin 4 – N/C  
 Pin 5 – N/C  
 Pin 6 – RXD -  
 Pin 7 – N/C  
 Pin 8 – N/C

### Typical Straight-Through Cable



### Typical Crossover Cable



## General Purpose Interface (GPI/GPO)

AirAura X3 is equipped with eight General Purpose Input (GPI) input ports on one rear panel female DB-9 connector and four General Purpose Output (GPO) tally outputs on a second female DB-9 connector. All GPI and GPO connections are optically-isolated from X1's internal circuitry to prevent external ground loops and to prevent dangerous voltages from being introduced into X1. Further, the GPI inputs accept DC voltages of either polarity, easing interfacing in the field.

The eight GPI inputs are hard-coded in software to activate the first eight preset slots.

The four GPO ports on X1 are hard coded in software to provide the following status outputs:

- GP Output #0 – Becomes enabled on Analog Audio Failure.
- GP Output #1 – Becomes enabled on Digital Audio Failure (either AES3 or WheatNet-IP).
- GP Output #2 – Becomes enabled if the CPU Temperature reaches 50 deg. C (122 deg. F).
- GP Output #3 – Becomes enabled upon a General System Failure.

### GPI

The GPI interface provides eight separate General Purpose Inputs, sharing a common return. The connector is a female DB-9 connector with the following pin assignments. Note that all pins are isolated from X1 internal circuitry!

Pin 1 – GPI COM  
 Pin 6 – GPI 1 IN  
 Pin 2 – GPI 2 IN  
 Pin 7 – GPI 3 IN  
 Pin 3 – GPI 4 IN  
 Pin 8 – GPI 5 IN  
 Pin 4 – GPI 6 IN  
 Pin 9 – GPI 7 IN  
 Pin 5 – GPI 8 IN

### GPO

The pin assignment for the rear panel DB-9 GPO connector is as follows:

Pin 1 - N/C  
 Pin 2 – GPO 1 Return  
 Pin 6 – GPO 1 Source  
 Pin 3 – GPO 2 Return  
 Pin 7 – GPO 2 Source  
 Pin 4 – GPO 3 Return  
 Pin 8 – GPO 3 Source  
 Pin 5 – GPO 4 Return  
 Pin 9 – GPO 4 Source



## Starting With a Preset

The X1 comes equipped with several factory presets and can hold a combination of 160 factory and user presets in its onboard memory. Presets may be saved within X1's onboard memory until all preset storage slots are full. An unlimited number of presets can be stored on the PC that hosts the Windows-based remote control GUI software. From the default system page, tap the green button that says PRESETS. This page will then display. Use the *Prev List* and *Next List* arrows to move the preset list up and down. To select a preset, tap on the name and the hardware will load that preset.



## Processing Presets

In order to prevent clicks and pops when presets are changed, preset parameters are slewed between the current parameter values and those of the new preset whenever a new preset is taken. Because of this, when changing presets, especially between presets that are tuned quite differently, it may take several seconds for the new settings to completely settle in. It is important to remember this concept; X1's preset switching is designed to make it as unobtrusive as possible and therefore parameter changes are not instantaneous. This factor must be taken into consideration whenever switching between presets in order to compare them!

When a preset has been recalled and has not been modified the preset's name is displayed in green text within the GUI's current preset window. If changes to the preset have been made its name will be displayed in red text instead of green. Once the modified settings have been saved back to X1's hardware the preset name will again be displayed in green.

Factory pre-sets can be readjusted and saved to new names in order to create a completely different air sound. The factory presets are write-protected and changes made to them cannot be written back to the same memory location. Factory presets that have been modified are considered by the system to be "user" presets and therefore must be saved as a new name and in a new preset storage slot.

Our advice is to start with a factory preset that has the on air sound that is closest to what you believe you need. If changes are necessary, the best approach is to make small changes, one or two at a time, and then listen for quite a while before deciding that more changes are necessary. A consultant friend of ours advises: *“Tweak small ... Listen large”*.

**NOTE:**

User presets that have been stored in preset slots 1 through 8 are logically assigned to the eight remote GPI functions. Examples of presets that might be stored here are:

- A preset that has all processing turned off and/or has special input/output level calibrations (such as for testing).
- A preset that has all processing enabled but has the Stereo Pilot turned off (a Mono Preset).
- A preset that changes the input or output source or level calibrations (note: “System Settings Change with Preset Takes” must be enabled in the System screen of the GUI in order for presets to control I/O settings).

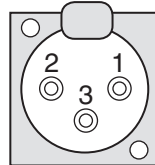
Preset storage is as follows:

- Preset slots 1 - 8 are for user presets that can be selected by the optoisolated GPI interface.
- Factory presets are installed beginning with a BYPASS stored in slot #9. Factory presets cannot be written over or deleted.
- User presets, other than the GPI selectable presets in slots 1 through 8, are stored above the highest-numbered factory preset. The number of available slots for user presets depends on how many factory presets were installed which can vary with software version. The total number of presets on X1’s hardware cannot exceed eighty (80). The storage space available for presets on the GUI’s host PC (because of a preset’s tiny file size) is virtually unlimited.
- User presets may be locked at the user’s discretion to prevent inadvertent changes. Any user can unlock another user-locked presets.

## Analog XLR Connections

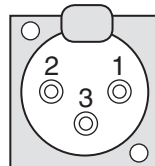
### Analog IN - XLR-F

#### XLR LT-F



PIN 1 - LINE LT IN SH  
PIN 2 - LINE LT IN HI  
PIN 3 - LINE LT IN LO

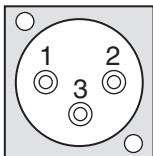
#### XLR RT-F



PIN 1 - LINE RT IN SH  
PIN 2 - LINE RT IN HI  
PIN 3 - LINE RT IN LO

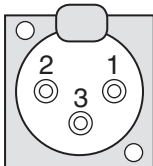
### INSERT LOOP - XLR

#### XLR OUT-M



PIN 1 - INSERT LOOP OUT SH  
PIN 2 - INSERT LOOP OUT HI  
PIN 3 - INSERT LOOP OUT LO

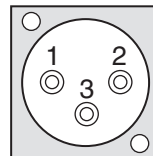
#### XLR IN-F



PIN 1 - INSERT LOOP IN SH  
PIN 2 - INSERT LOOP IN HI  
PIN 3 - INSERT LOOP IN LO

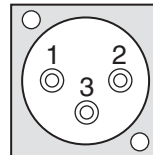
### FM/HD ANALOG OUT - XLR-M

#### XLR LT-M



PIN 1 - FM/HD LT OUT SH  
PIN 2 - FM/HD LT OUT HI  
PIN 3 - FM/HD LT OUT LO

#### XLR RT-M

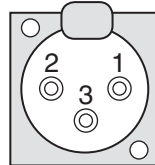


PIN 1 - FM/HD RT OUT SH  
PIN 2 - FM/HD RT OUT HI  
PIN 3 - FM/HD RT OUT LO

## Digital XLR Connections

### AES IN - XLR-F

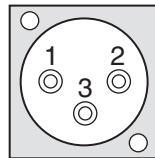
#### XLR-F



PIN 1 - AES IN SH  
PIN 2 - AES IN HI  
PIN 3 - AES IN LO

### AES FM Out/baseband 192 - XLR-M

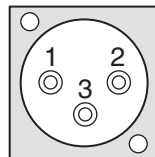
#### XLR FM-M



PIN 1 - AES FM/BASEBAND192 OUT SH  
PIN 2 - AES FM/BASEBAND192 OUT HI  
PIN 3 - AES FM/BASEBAND192 OUT LO

### AES HD Out - XLR-M



#### XLR HD-M





PIN 1 - AES HD OUT SH  
PIN 2 - AES HD OUT HI  
PIN 3 - AES HD OUT LO

## ***BNC Connections***

### ***BNC - SCA***

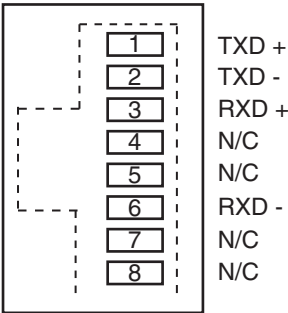
- 1  PIN 1 - SCA 1 IN HI - CENTRAL PIN  
PIN 2 - SCA 1 IN SH - SHELL
- 2  PIN 1 - SCA 2 IN HI - CENTRAL PIN  
PIN 2 - SCA 2 IN SH - SHELL

### ***BNC - TX***

- 1  PIN 1 - TX 1 OUT HI - CENTRAL PIN  
PIN 2 - TX 1 OUT SH - SHELL
- 2  PIN 1 - TX 2 OUT HI - CENTRAL PIN  
PIN 2 - TX 2 OUT SH - SHELL

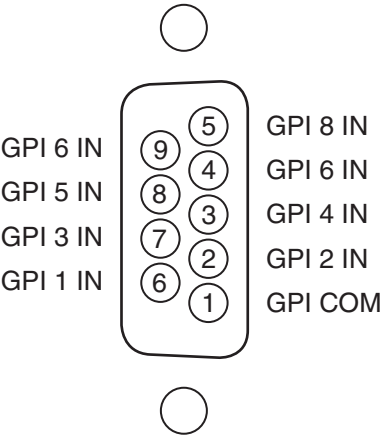
# Ethernet - RJ-45

## RJ-45 ETHERNET

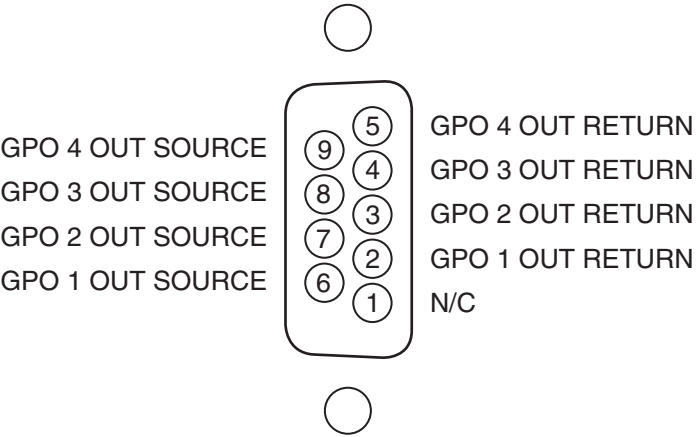


# General Purpose Interface - DB-9

## GPI - DB-9



## GPO - DB-9



# Front Panel GUI

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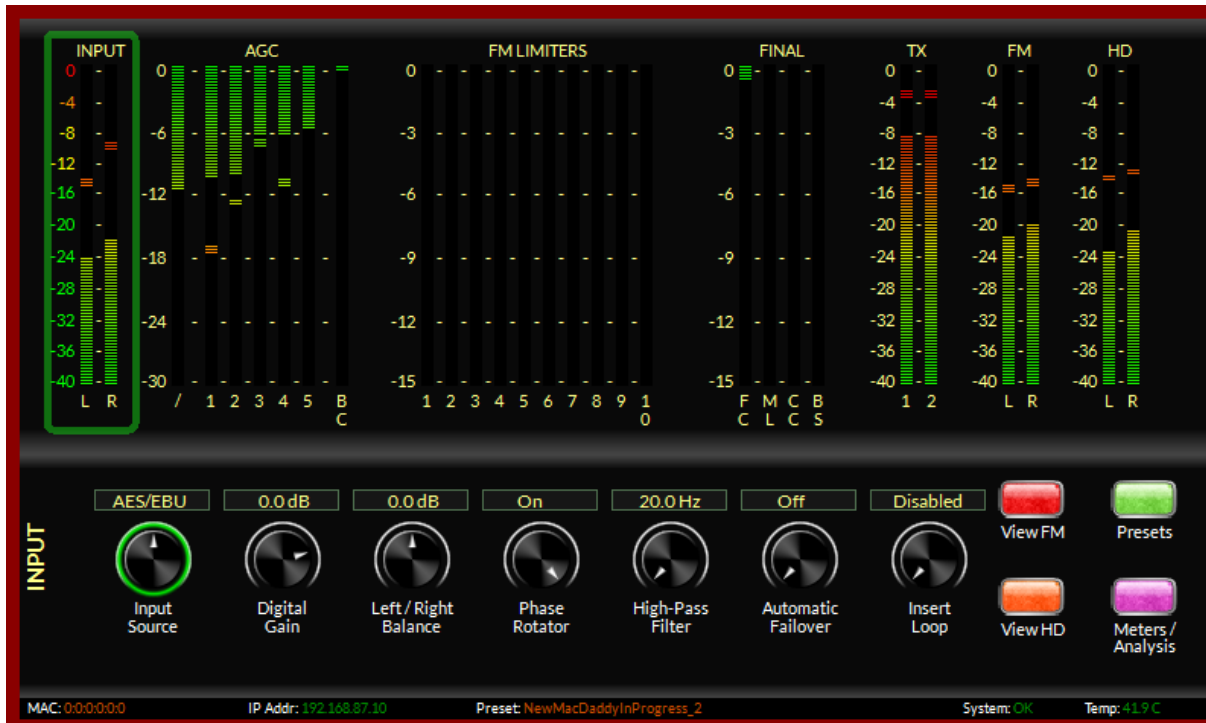


# Front Panel GUI

In addition to comprehensive metering and analysis displays, the front panel also hosts our highly-regarded “Audio Processing Guru®” user interface. These are the key controls governing each section of the audio processor that makes it easy to set up and adjust the sound. For more comprehensive controls, you will need to connect to the hardware using the Windows GUI interface.

## Input Screen

Touching the INPUT meters on the front panel touchscreen brings up the input controls. This allows you to select the input source, digital gain, left/right balance (to correct for left/right level discrepancies), phase rotator, high pass filter, automatic failover (analog to digital OR digital to analog in the event of audio loss) and activation of the insert loop.



## Input Source

To change the input source, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to scroll thru the options available. There are three input source options. Analog, AES/EBU or Wheatnet. Once the correct input source has been selected, push the knob in to activate.

**NOTE:** The source will not change unless you push the knob in to confirm your choice. The control will time out and default to the current source after 15 seconds if no other selection is made.



## Digital Gain

Offsets the input level of the AES or Wheatnet signal. This control is visible only when AES/EBU or Wheatnet is selected as the input. To change the digital gain, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to increase or decrease the gain. The input level will change in real time as you move the knob. Once you have reached the desired level, push in the knob to confirm the new value.

*Note: the digital gain control adjusts the input level for both the Wheatnet-IP source as well as the AES source.*



## Analog Gain

Offsets the input level of the analog signal. This control is visible only when analog is selected as the input. To change the analog gain, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to increase or decrease the gain. The input level will change in real time as you move the knob. Once you have reached the desired level, push in the knob to confirm the new value.



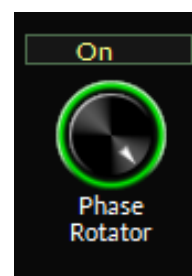
## L/R Balance

Offsets the left/right balance of the selected input. To change the left/right balance, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to favor gain to the left or the right channel. The balance will change in real time as you move the knob. Once you have reached the desired level, push in the knob to confirm the new value.



## Phase Rotator

The phase rotator helps correct asymmetrical audio often found in human voice. It is recommended to leave the phase rotator "ON" especially if you plan to process aggressively. To turn the phase rotator on or off, tap the button. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select on or off. Once the correct mode has been selected, push the knob in to confirm.



## High Pass Filter

The High Pass Filter is used to remove inaudible and unnecessary subsonic energy from the audio signal prior to it being processed. By removing this energy, processing is cleaner and modulation energy is not wasted by transmitting sounds that would not be perceptible to a listener.

To change the high pass filter frequency, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to set the desired frequency. The range is 20Hz to 300Hz. Once the desired frequency has been reached, push in the knob to confirm the change.



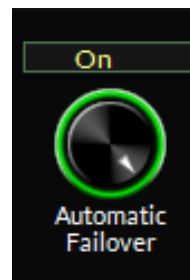
## Automatic Failover

Automatic failover from AES/EBU or Wheatnet-IP to analog is instantaneous and based on invalid or missing bits in the AES3 stream, or after 30 seconds of “silence” (level below -48dBFS).

Automatic failover from analog to AES3 or Wheatnet-IP is based on silence sense responding to audio on both channels being below -48dBFS for more than 30 seconds.

Failover capability is *not available* between the AES and Wheatnet-IP inputs.

***Before you set the automatic failover, make sure you are on the input source that will be your primary audio source.*** To set the automatic failover, tap the knob. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select on or off. Once the correct mode has been selected, push the knob in to confirm.



## Insert Loop Option

The insert loop option allows the user to insert a ratings encoder watermark or any other external processing gear into the middle of the X1’s “airchain”. This allows the user to have the X1’s iAGC and five band processing appear before any ratings encoder so that the encoder is fed a consistent level.

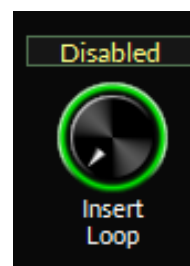
Prior to this option, many users resorted to external AGC’s to feed the ratings encoder prior to the main processor. Adding an additional gain control device to the airchain had the tendency to cause audible fighting between the external AGC and the internal AGC in the main processor. Defeating the AGC in the main processor was not always an option.

The correct solution is to provide an insert point in the main processor’s audio path to accommodate the ratings encoder. This way, not only do we have one AGC that controls the input level to the processor AND the ratings encoder, we can now place the ratings encoder towards the back of the processing so the watermark is less likely to be altered by any previous multiband stages.

To turn on the insert loop, tap the knob. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select on or off. Once the correct mode has been selected, push the knob in to confirm.

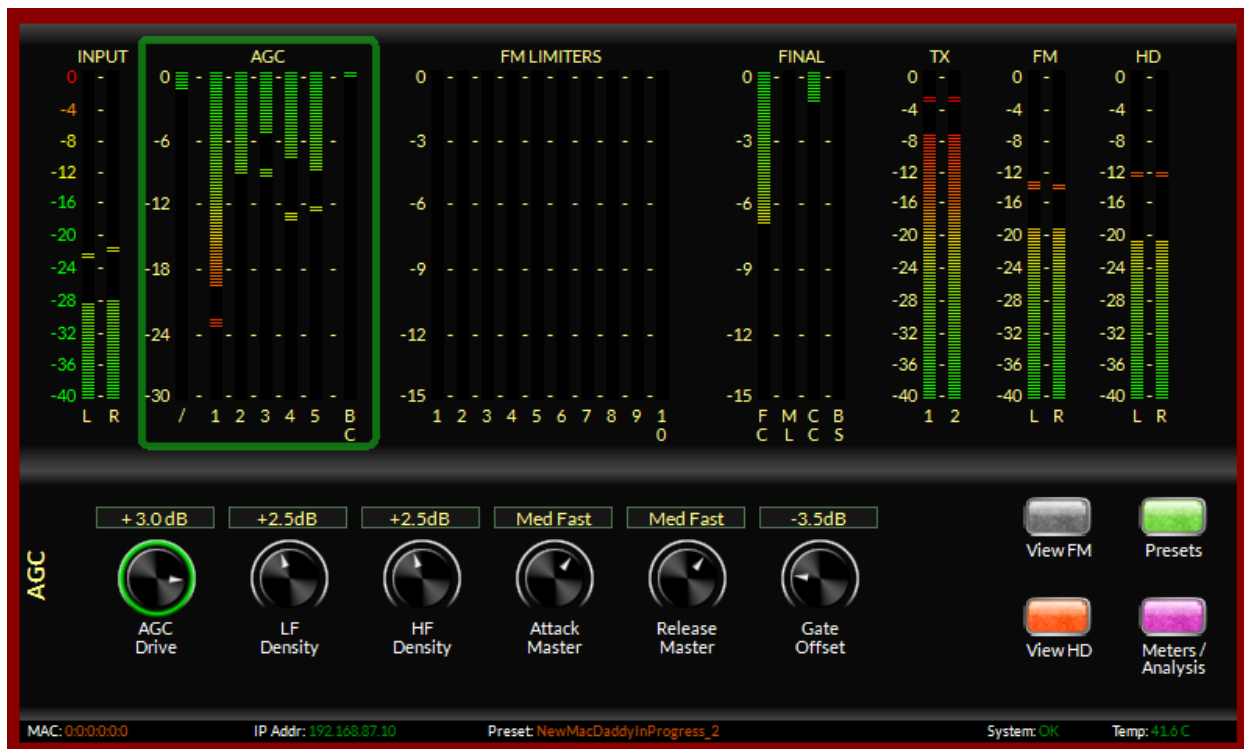
***NOTE: If there are no devices in the insert loop, audio will be interrupted if you enable the insert loop. The insert loop accepts AES/EBU signals only. The default sample rate is 48kHz. If you desire the sample rate of the loop to be 44.1kHz, you will need to feed a 44.1kHz signal to the main AES input of the X1. As of this writing, there is no audio failover for signal loss in the insert loop.***

The insert loop is send and return levels are fixed. The send level is normally within a target of -24dBFS to -20dBFS, depending on how aggressive the processing is set up in the 5 band. If you are only looping the ratings encoder, the return audio should be at the same amplitude as when it left the processor. If you are adding any other devices into the insert loop, refer to their documentation for proper audio level line up procedures.



## 5 Band iACC

Touching the 5 band iAGC meters on the front panel touchscreen brings up the controls for the 5 band processing. This allows you to adjust AGC drive, density, attack and release and gate offset.



### AGC Drive

Offsets the drive level to the 5 band processing. The range is -12dB to +6dB. To change the AGC drive, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to increase or decrease the gain. The level will change in real time as you move the knob. Once you have reached the desired level, push in the knob to confirm the new value.



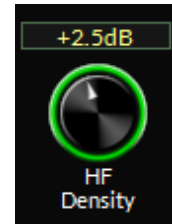
### LF Density

Increases density in the bass frequencies. The range is 0 (open) to +6 (aggressive). To change the LF density, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to increase or decrease the gain. The processing effect will change in real time as you move the knob. Once you have reached the desired level, push in the knob to confirm the new value.



### HF Density

Increases density in the mid and high frequencies. The range is 0 (open) to +6 (aggressive). To change the HF density, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel



knob to increase or decrease the gain. The processing effect will change in real time as you move the knob. Once you have reached the desired level, push in the knob to confirm the new value.

## Attack Master

Governs how quickly the 5 band processing corrects audio. There are nine positions in this control. Eight of the nine are manual controls... that is, they attack based on the user setting (slowest, slower, slow, med slow, med fast, fast, faster, fastest). The center position is AUTOMATIC or AUTO. When in this position, the attack time is program dependent.

To change the master attack, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select the type of attack. The processing effect will change in real time as you move the knob. Once you have reached the desired setting, push in the knob to confirm the new value.



## Release Master

Governs how quickly the 5 band processing recovers after correcting audio. There are nine positions in this control. Eight of the nine are manual controls... that is, they release based on the user setting (slowest, slower, slow, med slow, med fast, fast, faster, fastest). The center position is AUTOMATIC or AUTO. When in this position, the release time is program dependent.

To change the master release, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select the type of release. The processing effect will change in real time as you move the knob. Once you have reached the desired setting, push in the knob to confirm the new value.



## Gate Offset

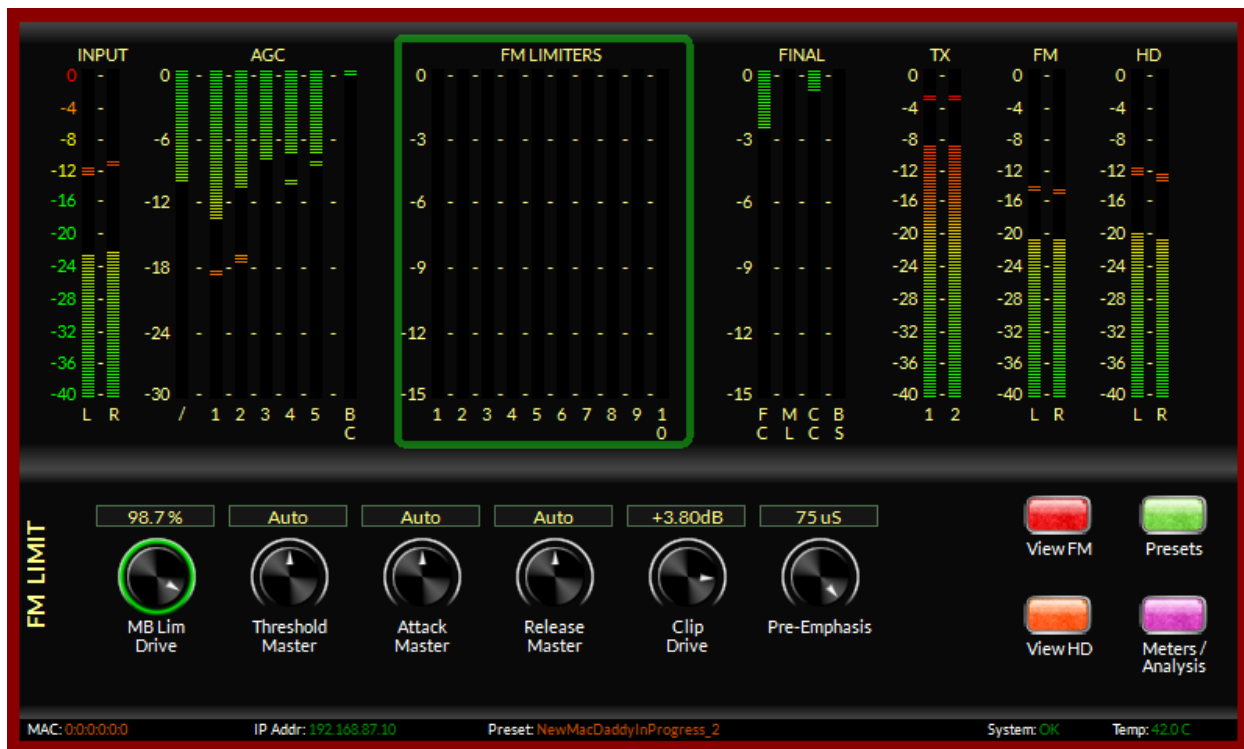
Offsets the default gate value in the preset. Positive numbers tighten up the gate, while negative numbers allow for more leveling of the audio before the gate is applied.

To change the gate offset, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to increase or decrease the gain. The processing effect will change in real time as you move the knob. Once you have reached the desired level, push in the knob to confirm the new value.



## 10-Band FM Limiter

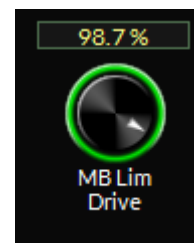
Touching the FM limiters metering on the front panel touchscreen brings up the controls for the 10-band limiter. This allows you to adjust limiter drive, threshold, attack, release, clipper drive and pre-emphasis.



### MB Lim Drive

Determines the amount of input signal applied to the 10-band limiter — less drive creates a more gentle sound while still controlling peaks. Higher settings make the sound more aggressive by increasing short term density. The range is 80-100%.

To change the MB Lim drive, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to increase or decrease the gain. The level will change in real time as you move the knob. Once you have reached the desired level, push in the knob to confirm the new value.



### Threshold Master

Determines the threshold of the 10-band limiter. The range is -6dB to +6dB with the center position being AUTO (program dependent).

To change the threshold master, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select the type of release. The processing effect will change in real time as you move the knob. Once you have reached the desired setting, push in the knob to confirm the new value.





## Attack Master

Governs how quickly the 10-band limiter corrects for peaks. There are nine positions in this control. Eight of the nine are manual controls... that is, they attack based on the user setting (slowest, slower, slow, med slow, med fast, fast, faster, fastest). The center position is AUTOMATIC or AUTO. When in this position, the attack time is program dependent.

To change the master attack, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select the type of attack. The processing effect will change in real time as you move the knob. Once you have reached the desired setting, push in the knob to confirm the new value.



## Release Master

Governs how quickly the 10-band limiter recovers after correcting audio. There are nine positions in this control. Eight of the nine are manual controls... that is, they release based on the user setting (slowest, slower, slow, med slow, med fast, fast, faster, fastest). The center position is AUTOMATIC or AUTO. When in this position, the release time is program dependent.

To change the master release, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select the type of release. The processing effect will change in real time as you move the knob. Once you have reached the desired setting, push in the knob to confirm the new value.



## Clip Drive

Governs how much drive there is to the final peak clipper. This control can make extreme changes to the audio, so care should be given as to how hard you drive this control. Sometimes, a little can go a long way.

To change the drive control level, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select the type of release. The processing effect will change in real time as you move the knob. Once you have reached the desired setting, push in the knob to confirm the new value.



## Pre-Emphasis

To change the pre-emphasis value, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to scroll thru the options available. There are three pre-emphasis options. FLAT, 50uS and 75uS. Once the correct pre-emphasis has been selected, push the knob in to activate.

**NOTE: North America uses 75us pre-emphasis, while other regions use 50us. Refer to your local governing body as to which pre-emphasis should be used if you are unsure.**



The analog outputs of the audio processor have an option to apply a de-emphasis filter to match the pre-emphasis selected here. The same can also be applied to the AES/EBU left/right output. Pre-emphasis is not applied to any HD output options.

## Where and When to Use Pre-Emphasis and De-Emphasis

Pre-emphasis was designed to help reduce the background noise of FM broadcasts. By raising high frequencies in the transmission system and applying a complimentary filter in the receiver, we gain additional signal to noise ratio for the listener.

Most processors have many options for applying pre-emphasis and de-emphasis. It can get confusing. So we will try to cover all the basics here.

### If You Are Using the Analog Composite Outputs in the X1

Pre-emphasis should be applied. If you are using the analog or AES output to feed headphones, you should de-emphasize these outputs or they will sound too “bright”.

### If You Are Using baseband192 (AES Composite) to Feed Your Exciter

Pre-emphasis should be applied. If you are using the analog output to feed headphones, you should de-emphasize these outputs or they will sound too “bright”.

### If You Are Using an AES/EBU Connection to the Exciter (where the exciter performs the stereo generator function)

Pre-emphasis should be applied AS WELL AS de-emphasis to the AES signal. This act will apply the proper peak control when pre-emphasis is re-applied later in the exciter. If you set the X1's pre-emphasis to FLAT and apply pre-emphasis at the exciter, you will be prone to gross overshoot as many exciters do not have overshoot filters to control pre-emphasis. The best (and not always available) option if you must use an AES/EBU connection that is not composite is to perform pre-emphasis at the processor before a linear STL path and NOT perform any further emphasis in the exciter.

### Using the AES FM Output to Feed a Codec

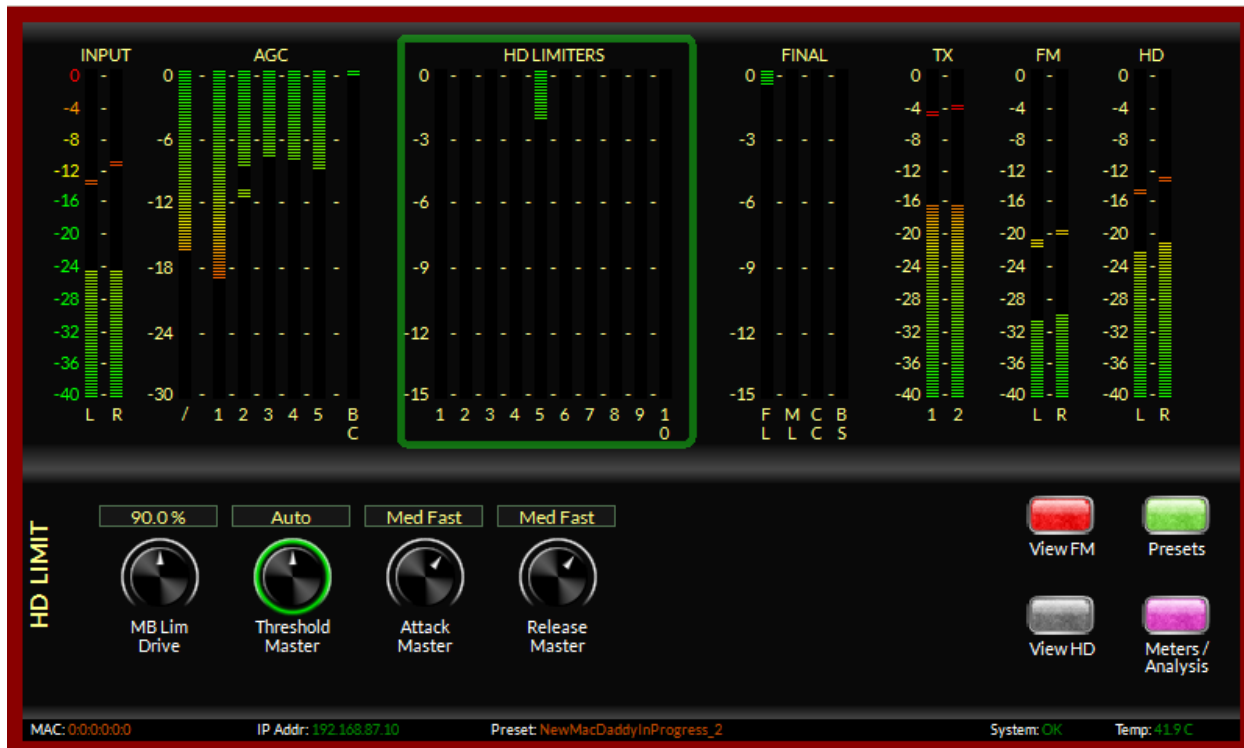
Pre-emphasized audio (with or without de-emphasis) should never be applied to an audio codec. If you need processing for streaming from your X1, the HD AES/EBU output or the analog outputs (set for HD) should be fed to your stream. Feeding audio managed for FM to an internet stream will unveil artifacts. If you must feed your stream with audio processed for FM, you should not expect great results.

***NOTE: For more information on setting de-emphasis for the outputs on the X1, please see page 2-20.***



## 10-Band HD Limiter

To access the HD limiter metering on the front panel touchscreen, tap “View HD”. The FM limiter controls will be replaced by the HD limiter controls, brings up the controls for the 10-band limiter. Now you can adjust limiter drive, threshold, attack and release.



### MB Lim Drive

Determines the amount of input signal applied to the 10-band limiter — less drive creates a more gentle sound while still controlling peaks. Higher settings make the sound more aggressive by increasing short term density. The range is 80-100%.

To change the MB Lim drive, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to increase or decrease the gain. The level will change in real time as you move the knob. Once you have reached the desired level, push in the knob to confirm the new value.



### Threshold Master

Determines the threshold of the 10-band limiter. The range is -6dB to +6dB with the center position being AUTO (program dependent).

To change the threshold master, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select the type of release. The processing effect will change in real time as you move the knob. Once you have reached the desired setting, push in the knob to confirm the new value.



## Attack Master

Governs how quickly the 10-band limiter corrects for peaks. There are nine positions in this control. Eight of the nine are manual controls... that is, they attack based on the user setting (slowest, slower, slow, med slow, med fast, fast, faster, fastest). The center position is AUTOMATIC or AUTO. When in this position, the attack time is program dependent.

To change the master attack, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select the type of attack. The processing effect will change in real time as you move the knob. Once you have reached the desired setting, push in the knob to confirm the new value.



## Release Master

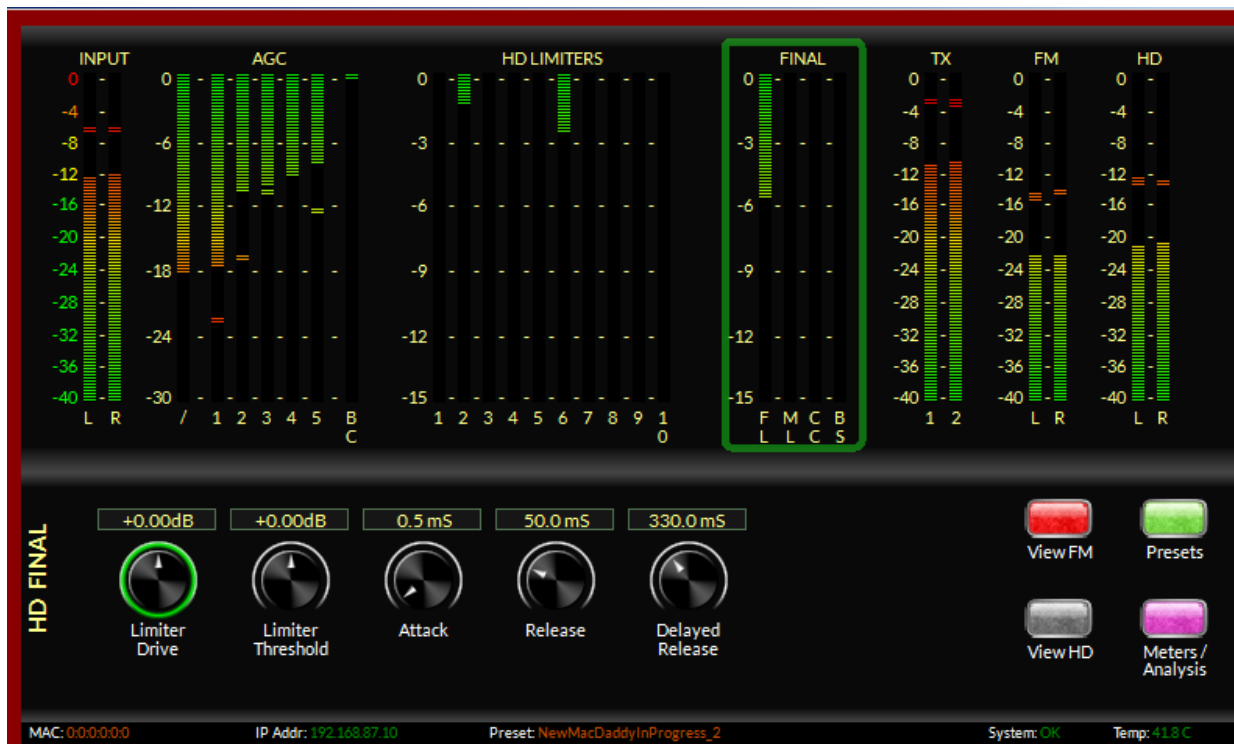
Governs how quickly the 10-band limiter recovers after correcting audio. There are nine positions in this control. Eight of the nine are manual controls... that is, they release based on the user setting (slowest, slower, slow, med slow, med fast, fast, faster, fastest). The center position is AUTOMATIC or AUTO. When in this position, the release time is program dependent.

To change the master release, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select the type of release. The processing effect will change in real time as you move the knob. Once you have reached the desired setting, push in the knob to confirm the new value.



## HD Final Limiter

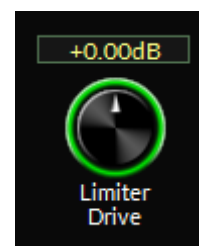
To access the HD final processing on the front panel touchscreen, tap “View HD”. The FM final controls will be replaced by the HD final controls.



### Limiter Drive

Determines the amount of input signal applied to the oversampled lookahead limiter — less drive creates a more gentle sound while still controlling peaks. Higher settings make the sound more aggressive by increasing short term density.

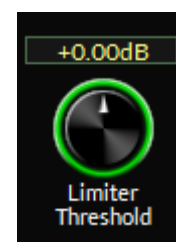
To change the final Limiter drive, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to increase or decrease the gain. The level will change in real time as you move the knob. The range is -6dB to +6dB. Once you have reached the desired level, push in the knob to confirm the new value.



### Limiter Threshold

Determines the threshold of the lookahead limiter. The range is -6dB to +6dB.

To change the limiter threshold, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select the type of release. The processing effect will change in real time as you move the knob. Once you have reached the desired setting, push in the knob to confirm the new value.



## Attack

Governs how quickly the lookahead limiter corrects for peaks. The timing range is 0.2mS to 100mS.

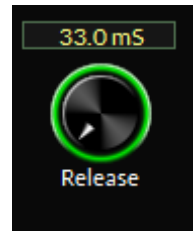
To change the attack, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select the type of attack. The processing effect will change in real time as you move the knob. Once you have reached the desired setting, push in the knob to confirm the new value.



## Release

Governs how quickly the lookahead primary gain control recovers after correcting audio. The timing range is 33mS to 300mS.

To change the release, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select the type of release. The processing effect will change in real time as you move the knob. Once you have reached the desired setting, push in the knob to confirm the new value.

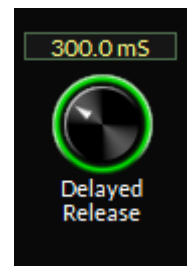


## Delayed Release

Governs how quickly the lookahead secondary gain control recovers after correcting audio. The timing range is 100mS to 1000mS.

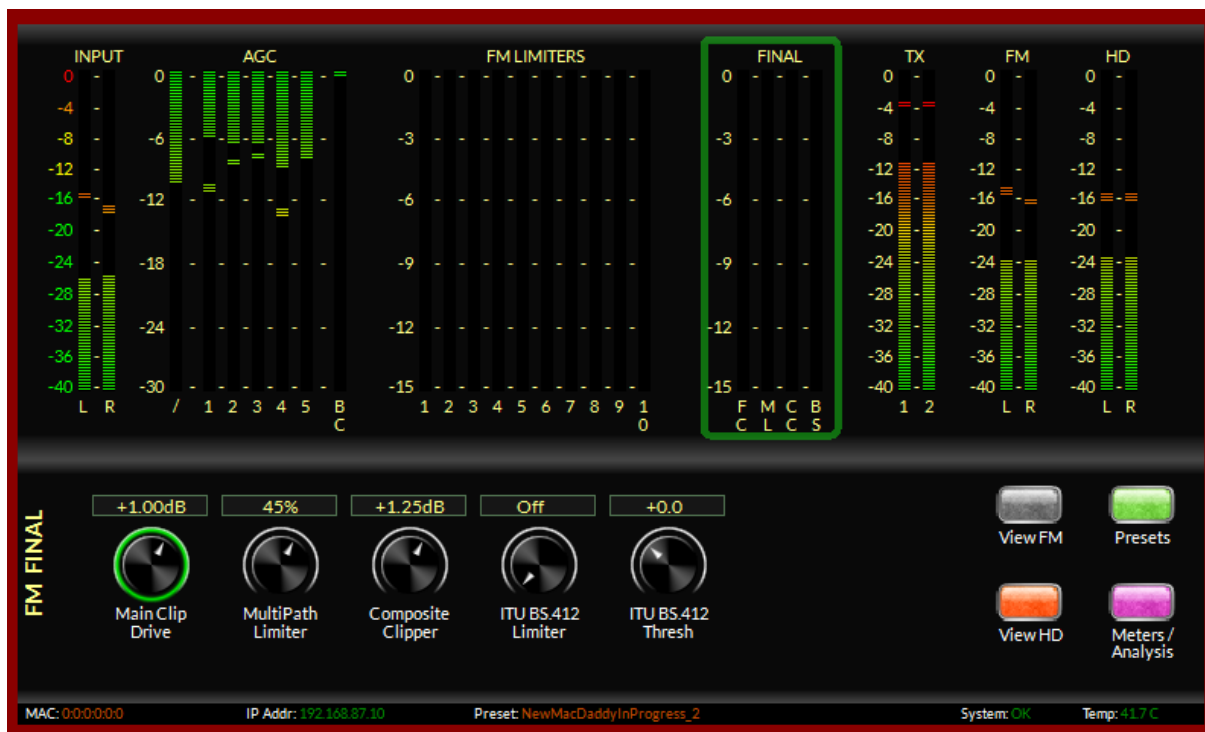
The secondary delayed release control creates a slower platform for the faster look-ahead limiter to release to. This prevents unwanted pumping or grunge from more conventional designs where instantaneous peak control is needed without clipping.

To change the release time, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select the type of release. The processing effect will change in real time as you move the knob. Once you have reached the desired setting, push in the knob to confirm the new value.



## FM Final Clipper

Touching the FINAL metering on the front panel touchscreen brings up the controls for the FM clipper.



## Main Clipper Drive

Determines the amount of drive to the distortion-cancelled final clipper. The normal range for most presets is between 0dB and +3dB. Higher numbers will increase loudness but have the potential to add distortion.

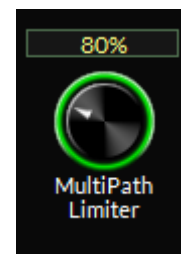
To change the main clipper drive, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select the type of release. The processing effect will change in real time as you move the knob. Once you have reached the desired setting, push in the knob to confirm the new value.



## Multipath Limiter

In order to make stereo enhancement “play nice” with the majority of stereo receivers in real-world listening environments with all types of program material, it is preferable to have some sort of controlling mechanism in place to ‘manage’ the amount of L-R energy present in the transmitted signal as a function of program material. This is precisely what the Multipath Limiter does.

When active, the multipath limiter will restrict the L-R information in the audio signal to a percentage of the overall L+R, this control is calibrated in %. In almost all cases, we recommend starting with a setting of 80%. If your radio station is prone to receiver blend, reduce the control to 70, 60 or 50% and listen for improvements.



To change the multipath limiter control, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select the type of release. The processing effect will change in real time as you move the knob. Once you have reached the desired setting, push in the knob to confirm the new value.

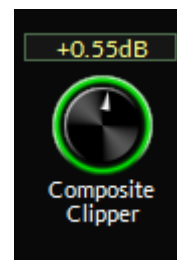
There are eleven operational choices available under the Stereo Width Limiter control settings: 10% to 100% in 10% percent steps, and an Off setting at either end of the control.

## Composite Clipper Drive

Determines the amount of drive to the X1's highly oversampled composite clipper. The composite clipper in the X1 has the ability to add legitimate loudness to the overall signal while minimizing audible distortion and subcarrier contamination.

Most presets have the composite clipper drive between 0 and +1.25dB. Settings above +2.0dB can start to cause audible distortion with high e'sses on human voice.

To change the composite clipper control, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select the type of release. The processing effect will change in real time as you move the knob. Once you have reached the desired setting, push in the knob to confirm the new value.



## ITU/BS.412 Controller

The Wheatstone BS 412 MPX Power Controller has a range from -3dB to +6dB in 0.1dB steps.

### TURNING ON THE BS 412 MPX POWER CONTROLLER

**IMPORTANT! The MPX Power Controller's sole purpose is to reduce loudness and program density. If you are not required to use the BS-412 Controller it should be left in the OFF position.**



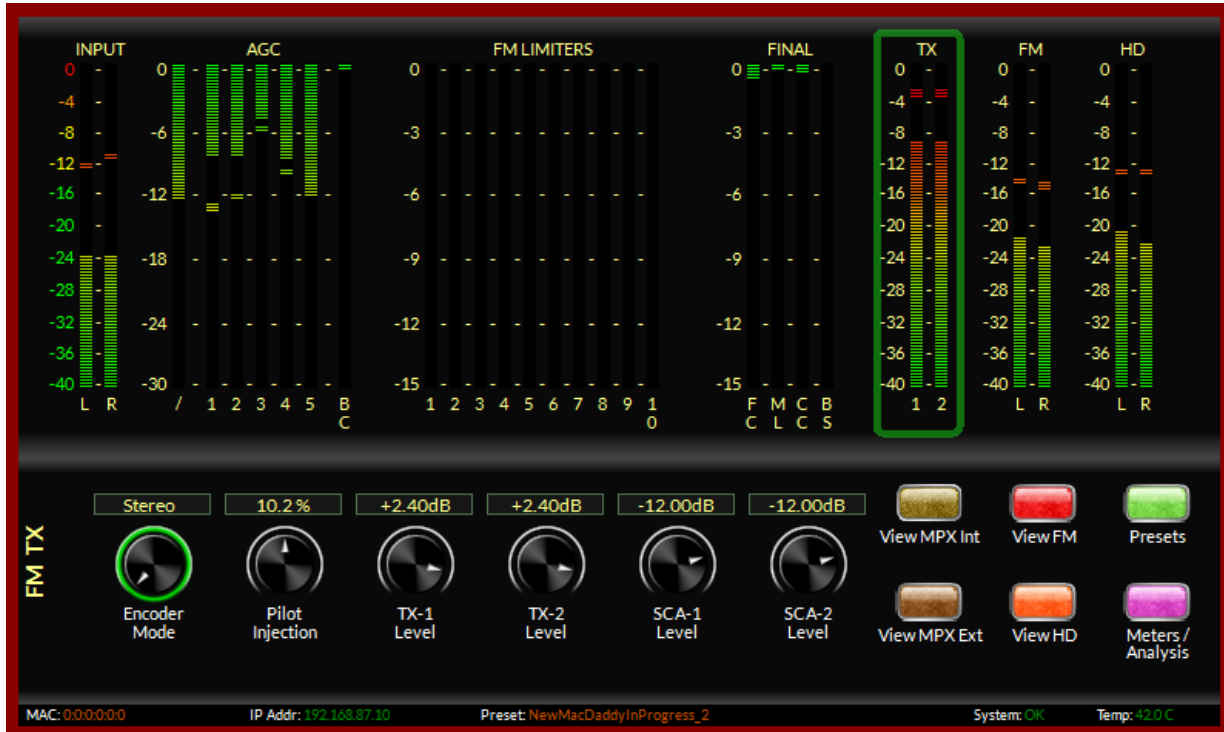
When the BS-412 control is switched to the "ON" position the MPX Power controller is engaged and the algorithm begins measuring and controlling the processor's total MPX power according to the ITU-R BS.412-7 standard. Because the measurement is essentially an average value integrated over a very long time the MPX Power controller will not make quick adjustments to the MPX Power when first engaged.

As the controller integrates the MPX energy the drive to the clippers will be modified until the measured MPX power satisfies the reference level as set by the Stereo Encoder menu's BS-412 control. The control's "0dB" setting conforms to the current ITU-R BS.412-7 Multiplex Power standard.

To turn on the controller, tap the ITY BS.412 Limiter knob. Use the front panel knob to select the mode. Once you have reached the desired setting, push in the knob to confirm the new value. Next, tap the ITU BS.412 Threshold control. Use the front panel knob to increase or decrease the threshold. Push in the knob to confirm the new value.

## FM TX Controls

Touching the TX metering on the front panel touchscreen brings up the controls for the FM composite outputs and SCA levels.



### Encoder Mode

The stereo encoder can be operated in Stereo or Mono by selecting the desired operating mode with this control

To change, tap the knob. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select mono or stereo. Once the correct mode has been selected, push the knob in to confirm.



### Pilot Injection

The pilot injection control adjusts the 19kHz stereo pilot amplitude and is adjustable from 0 (Off, to cause the station transmit in Mono) to 20% of the main composite signal amplitude in 0.1% steps.

Normal operation of this control should be at 9%. When adjusting, you should have a modulation monitor set up with the ability to read the pilot injection level to ensure this control is properly adjusted.

To change the pilot injection amount, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select amount. Your modulation monitor should indicate an increase or decrease in pilot level. Once you have reached the desired setting, push in the knob to confirm the new value.





## TX 1 and 2 Level

The TX1 & 2 LEVEL controls govern the composite output signals as they appear on the rear TX1 and TX2 BNC connectors. Normally these controls operate at +2.40dB for unity gain.

When adjusting, you should have a modulation monitor set up to ensure these controls are properly adjusted to match the modulation regulations of your area.

To change the level on either the TX1 or TX2, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select amount. Your modulation monitor should indicate an increase or decrease in modulation. Once you have reached the desired setting, push in the knob to confirm the new value.



**NOTE:** TX2 can also be switched to be a 19kHz pilot reference. This operation can be performed in the Windows based PRO GUI by navigating to the MPX and Outputs page, selecting the outputs tab and setting the TX2 output option from MPX out to Pilot.

## SCA 1 and 2 Level

The SCA 1 & 2 LEVEL controls govern the SCA injection amount from 3rd party subcarrier generators (such as an RDS generator) as a percentage of the main composite signal amplitude.

Normal operation of this control depends on a number of factors. You should set this control to -12.00dB to start. Have a modulation monitor capable of measuring the subcarrier you are adjusting ready to go with an RF sample or strong off air signal applied. Insert the 3rd party subcarrier generator into either the SCA 1 or SCA 2 port. If possible, **adjust the trim control on the 3rd party device** to match the proper injection level as displayed on the modulation monitor.

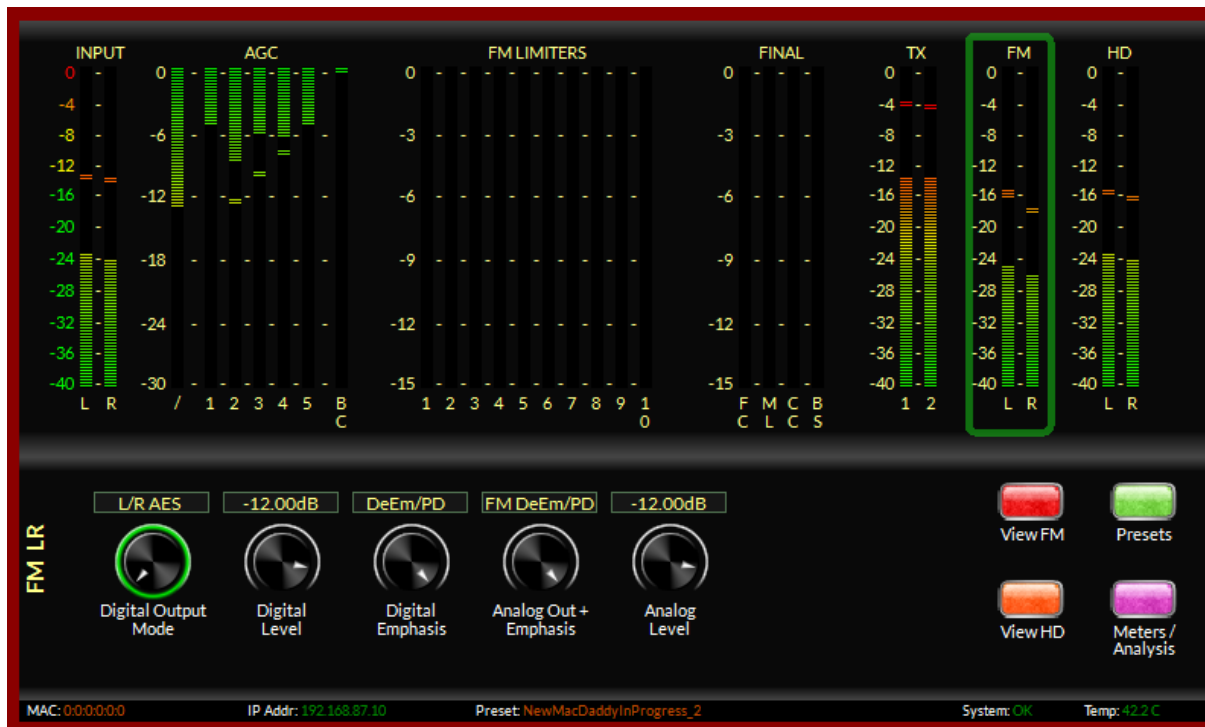


If there is no option to control the subcarrier injection level from the 3rd party device, you can offset the injection using the X1's SCA gain controls. To change the level on either the SCA 1 or SCA 2 injection levels, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select amount. Your modulation monitor should indicate an increase or decrease of the injection level. Once you have reached the desired setting, push in the knob to confirm the new value.



## FM L/R Output Controls

Touching the FM metering on the front panel touchscreen brings up the controls for the FM left/right outputs and emphasis options.

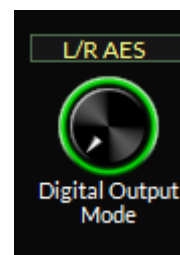


## Digital Output Mode

Changes the FMAES output from Left/Right AES to baseband192 (composite over AES).

To change, tap the knob. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select L/R AES or BB192. Once the correct mode has been selected, push the knob in to confirm.

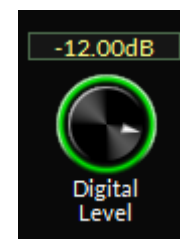
**NOTE: The baseband192 (BB192) option can only be used with an exciter that is capable of ingesting the baseband192 signal (composite AES). If you are unsure if your exciter can accept this signal, please contact the manufacturer of your exciter for more details.**



## Digital Output Level

Changes the FM AES output level. If you are feeding an exciter or STL with this output, you should have a modulation monitor set up to ensure this control is properly adjusted to match the modulation regulations of your area.

To adjust, tap the control. You will see a green ring indicating the control is active and ready to be changed. You will want to have use the front panel knob to select amount. You should see the modulation increase or decrease on your monitor. Once the correct mode has been selected, push the knob in to confirm.



## Digital Emphasis

There are four options for this control that governs the type of de-emphasis is placed on the FM AES output. You can select the output to be after the diversity delay and FLAT, before the diversity delay and FLAT, after the diversity delay and DE-EMPHASIZED or before the diversity delay and DE-EMPHASIZED (it should be noted that “PD” means PRE-DELAY and “D” means DELAY).

To adjust, tap the control. You will see a green ring indicating the control is active and ready to be changed. Turn the knob to select the option you desire for that output. Once the correct selection has been made, push in the knob to confirm

**NOTE:** For more information on which option to choose, please see page 2-10 “Where and When to Use Pre-Emphasis and De-Emphasis”.

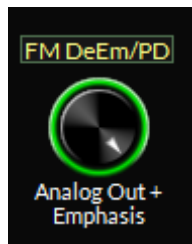


## Analog Out + Emphasis

This control governs the analog utility L/R output jacks on the rear panel of the X1. They can be assigned as FM outputs with various options for de-emphasis, or as another HD output. The options are HD, FM after the diversity delay and FLAT, FM before the diversity delay and FLAT, FM after the diversity delay and DE-EMPHASIZED or FM before the diversity delay and DE-EMPHASIZED (it should be noted that “PD” means PRE-DELAY and “D” means DELAY).

To adjust, tap the control. You will see a green ring indicating the control is active and ready to be changed. Turn the knob to select the option you desire for that output. Once the correct selection has been made, push in the knob to confirm.

**NOTE:** For more information on which option to choose, please see page 2-10 “Where and When to Use Pre-Emphasis and De-Emphasis”.



## Analog Output Level

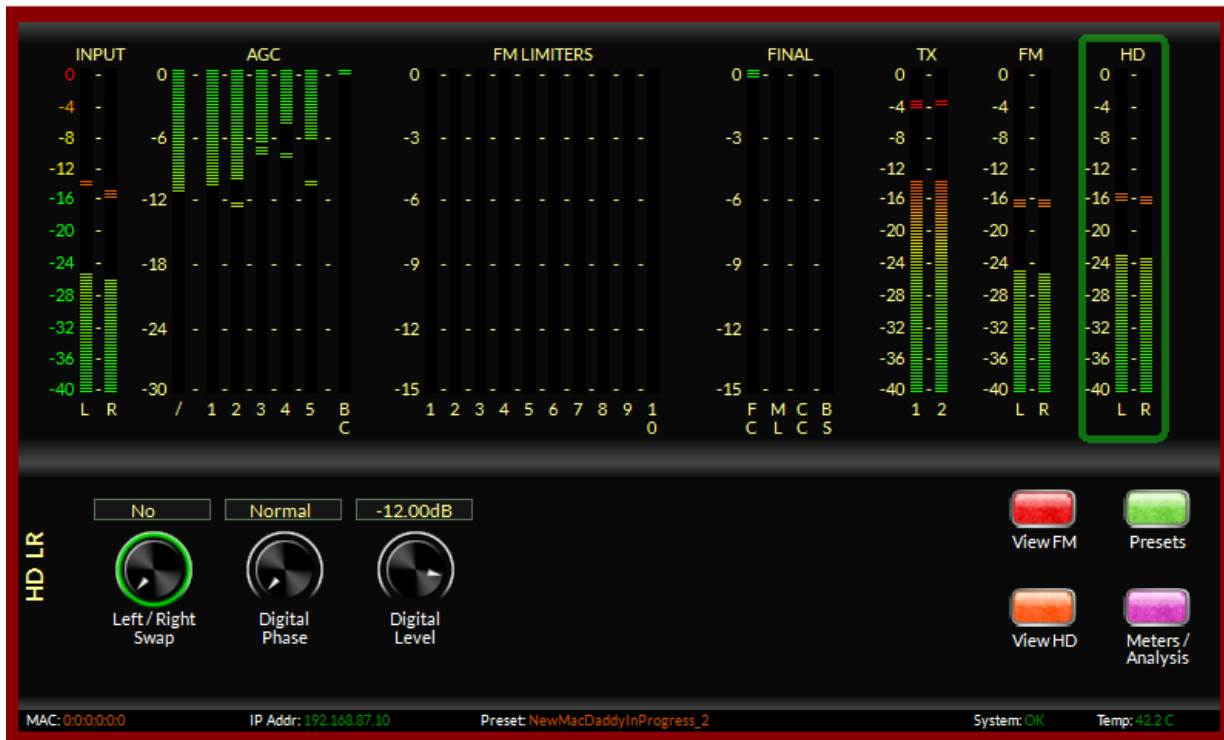
Changes the L/R analog output level. If you are adjusting this output to feed an STL or exciter, you should have a modulation monitor set up to ensure this control is properly adjusted to match the modulation regulations of your area.

To adjust, tap the control. You will see a green ring indicating the control is active and ready to be changed. You will want to have use the front panel knob to select amount. You should see the modulation increase or decrease on your monitor. Once the correct mode has been selected, push the knob in to confirm.



## HD Output Screen

Touching the HD metering on the front panel touchscreen brings up the controls for the HD outputs and options.



## Left / Right Swap

If needed, the left/right swap helps to match the analog left/right channel assignments. In most cases, you would not use this option.

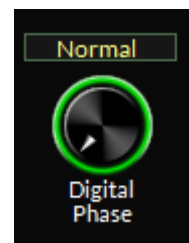
If, however, you do need to swap the Left and Right channels on the HD output, tap the control. You will see a green ring indicating the control is active and ready to be changed. The options are YES (swap the left and right channels) or NO (do nothing). Once the correct option has been selected, push the knob in to confirm.



## Digital Phase

If needed, the digital phase option matches the phase between the FM and HD audio signals.

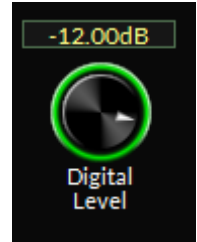
To change the polarity, tap the control. You will see a green ring indicating the control is active and ready to be changed. The options are NORMAL and REVERSED. Once the correct option has been selected, push the knob in to confirm.



## Digital Level

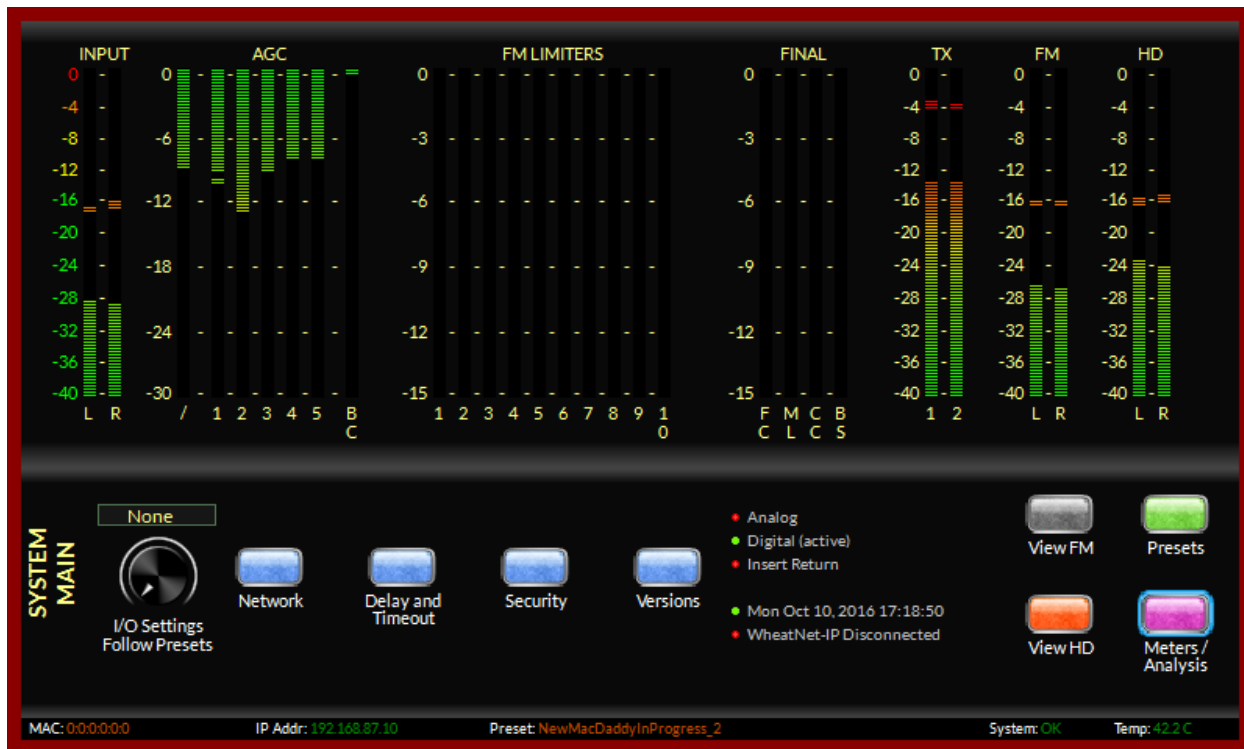
Changes the HD AES output level. If you are adjusting this output to feed an STL or exciter, care should be taken to monitor levels on the input of the STL or exciter.

To adjust, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select amount. You should see the modulation increase or decrease on your monitor. Once the correct mode has been selected, push the knob in to confirm.



## System

Hitting the ESC (escape) button at any time on the front panel brings up the system output page.



## I/O Settings Follow Presets

### Input/Output Settings Change With Preset Takes

When this option is OFF, recalling presets ignores the settings of the input and output level controls and other “system” controls associated with those functions. The following are considered to be X1 SYSTEM, or global parameters, which are saved with presets but not restored when presets are “taken,” unless one of the “Input/Output Settings Change With Preset Takes” options is chosen.



**These options are changed if the “INPUT” or “BOTH I/O” option is selected**

- Audio Source - Analog or AES
- Analog Gain (Analog)
- AES Gain (Digital)
- Balance (common to both Analog and Digital Inputs)
- Phase Rotator

**These options are changed if the “OUTPUT” or “BOTH I/O” option is selected**

- All DeEmphasis options
- All PreDelay options
- Digital Output

- Pilot
- BS412
- SCA 1
- SCA 2
- TX 1
- TX 2
- Pilot Only options
- Analog - L/R, L/R Deemph, and MPX output style choices

The option allows the X1 to modify its input and output gain settings whenever presets are recalled. Input and output gain settings are always stored with the presets when processing values are stored, but those settings will be ignored upon preset recall when the “System Settings Change with Preset Takes” checkbox is unchecked.

To adjust, tap the control. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to select the option you want (None, Input, Output, Both I/O). Once the correct mode has been selected, push the knob in to confirm.

## Network

Tapping the blue NETWORK button brings up this window:



This page allows you to assign the IP address for the front panel and the hardware. Because of how the system works (and because the front panel is essentially a GUI interface that connects to the hardware), your X1 requires two IP addresses to live on your network and function properly. Failure to properly assign an IP address will result in the inability to connect via the front panel or Windows based GUI to adjust sound. It will also interrupt WheatNet-IP audio I/O. If you are using analog or AES input and output signals, the processor will still function even with incorrect IP addresses.

There are five assignment knobs and a button to apply changes made to the address.

The first knob selects which parameter is to be changed. The options are FRONT PANEL ADDRESS (FP ADDR), HARDWARE ADDRESS (HW ADDR), SUBNET and GATEWAY. Tapping on the first button and using the knob to scroll thru the options allows you to access the various network settings.

Once you find the parameter you want to set, push the knob in. Now, tap on the 2nd knob. This will highlight the first of the four dot decimal numbers. Using the front panel knob, you can increase or decrease the value. When the desired value is found, press the knob in to confirm. Repeat these steps for the next three numbers and for changing other parameters.

As you change the IP Addresses, you will see the address list to the right of the knobs reflect the new addresses you are entering.

**NOTE: The actual IP address will NOT change until you are ready to reboot. You should reboot after all network settings are made and confirmed.**

Once your network settings have been assigned, press the APPLY ALL CHANGES button. You will need to reboot the processor for the new address to validate.

**NOTE: The Windows-based GUI and WheatNet-IP connectivity will happen on the Hardware (HW) IP address.**

## Front Panel IP Address (FP)

This is the reserved address that the front panel (not the remote GUI on the PC!) uses to communicate with the X1 system board. In a perfect world, this address should ALWAYS be configured to be one number higher than the IP address that you'll assign to the X1's "Hardware" address, but on the majority of networks should never end in "0" or "255."



## Hardware IP Address (HW)

This is the address of the X1's DSP array and is the address that remote GUIs running on host computers will use to connect to and control X1 over your network.

## Subnet Mask

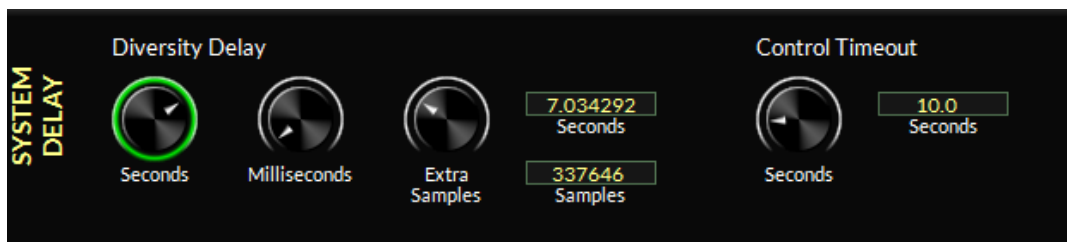
The subnet mask entry must be appropriate for the network being used. Please see your local friendly IT person if this is a bit confusing.

## Gateway

This is the Gateway Address that the X1 will use to reach connections outside of its own internal network. If you are on a local LAN segment or connected to the X1 by a direct Ethernet connection, the Gateway address is not usually necessary.

## Diversity Delay

Tapping the DELAY and TIMEOUT blue button on the system page brings up the menu to delay the analog FM signal in order to match the HD transmission delay.





It is advised that you have an HD modulation monitor capable of measuring the time alignment offset before you adjust. While many have been able to get it close tuning by ear, this method is not only time consuming, it can be very public with constant tweaking of the FM delay. The result will be audible “skipping” on the air.

The best unit we have found to measure the diversity delay offset (as of this writing) is the Belar FMHD-1. You can find more information about the Belar by visiting [www.belar.com](http://www.belar.com)

It should also be noted that the best way to manually set these controls is to use the Windows based GUI. In that environment, it is possible to double click on either the SECONDS or SAMPLES window and type in the value you want, rather than the tedious task of dialing it in. If you are using the Belar FMHD-1, the offset can be displayed in samples, which is a much more accurate measurement than the time sample.

The Belar unit mentioned above is also capable of automatically maintaining FM and HD audio sync once the alignment has been performed. The FMHD-1 talks with the X1 over the Ethernet connection (both units need to be on the same network) and corrects for drift. The error rate is better than 3 samples... most systems are within 1 sample.

If you need to make adjustments from the front panel, there are three controls for the diversity delay. Seconds, Milliseconds and a fine tune knob to align the FM and HD audio transmissions via samples.

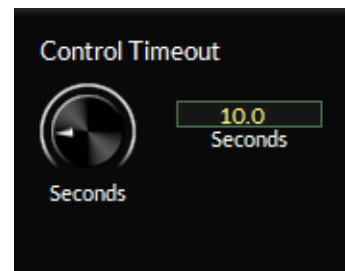
Tap on parameter you want to adjust. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to increase or decrease the delay. Pushing in the knob confirms the changes.

**NOTE:** *You will hear the delay change in real time on the FM broadcast.*

## Control Timeout

The control timeout governs the amount of time the X1 will wait to revert to the previous setting of a control if no confirmation is given when changing the control by pressing the knob.

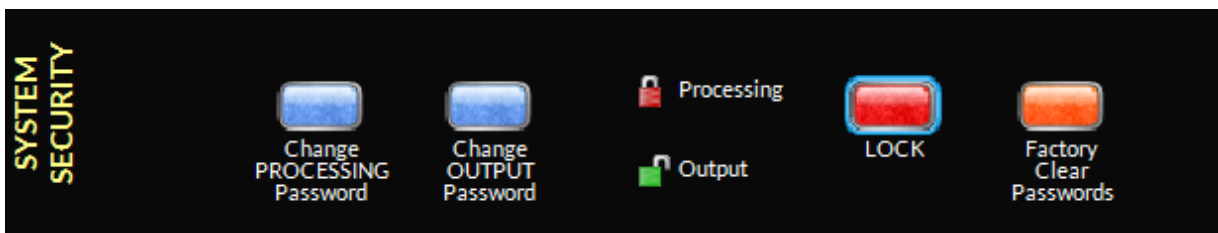
Tap on the knob. You will see a green ring indicating the control is active and ready to be changed. Use the front panel knob to increase or decrease the delay. Pushing in the knob confirms the changes.



## Security

The X1 is equipped with separate passwords that govern the processing and the output controls of the hardware. In this way, you can set different passwords and allow some users access to the sound controls and not the output controls, vice versa or both.

Tapping on the blue SECURITY button on the SYSTEM page brings up the options to set/change the PROCESSING Password or the OUTPUT Password.





Choose one of the options to password protect by clicking on it. You may enter any alpha-numerical password you wish up to EIGHT characters (the password IS case sensitive). The system will ask you to verify the password before accepting it. REMEMBER YOUR PASSWORD! Your processor will be locked out without it and you will need to contact Wheatstone to reset it.

Once the password(s) have been entered, you will see a new option to LOCK the processor.

Pressing LOCK will restrict access to the processor parameters that have assigned passwords. In the example above, the PROCESSING settings are locked, but the OUTPUT settings are unlocked.

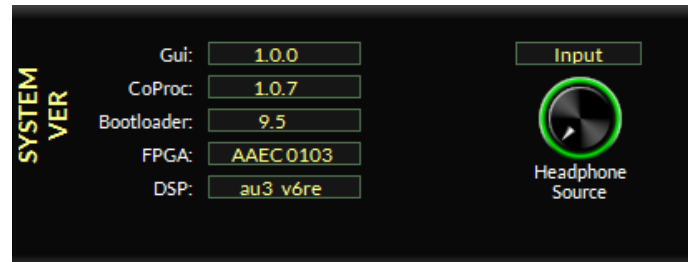
If at any time you lose the password, click on the FACTORY CLEAR PASSWORDS option. Follow the instructions.

***PLEASE NOTE: In order to clear the password, you will need to take the code generated by the processor when you tap the FACTORY CLEAR PASSWORDS option and contact the factory. Wheatstone cannot reset your passwords without the code.***

## Versions

Tapping the blue VERSIONS button brings up the information about the current software running on the processor. This may be useful when contact support to discuss issues.

Also included here is the option to change the source feeding the front panel headphone. The options are ACTIVE INPUT, ANALOG INPUT, DIGITAL INPUT, HD OUTPUT, FM PRE-DELAY and INSERT LOOP RETURN.



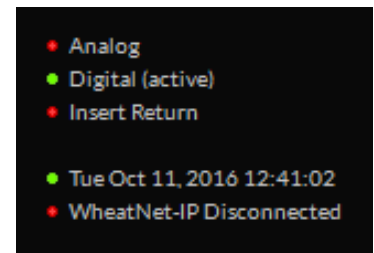
To adjust, tap the control. You will see a green ring indicating the control is active and ready to be changed. Turn the knob to select the option you desire for that output. Once the correct selection has been made, push in the knob to confirm.

## Status

The front panel also shows the STATUS of the processor. Green indicators show when a feature is active, red shows audio loss or an inactive feature.

Here, the ANALOG input has no audio (Red status indicator). The Digital AES/EBU input has audio (Green status indicator) and is also active (as seen by the "active" status next to the word "Digital").

- There is no audio or AES sync at the Insert Return
- The time of day is updated as noted by the green indicator
- WheatNet-IP is disconnected (red indicator)



## **Saving Your Work**

Once you have the processing dialed in, you can save the preset. From any of the screens, select the green PRESET button. You will see an alpha-numeric keyboard. On the lower right, select SAVE SETTINGS AS PRESET. The keyboard will illuminate and you can enter the name of your preset (64 characters maximum).

Once you have assigned a name, click SAVE SETTINGS AS PRESET again. The preset will be saved in the next empty slot.

The X1 comes with 160 preset slots available. Approximately 20 are used for factory presets (this number will change depending on software versions).

# X1 Pro GUI

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# X1 Pro GUI

The Windows based X1 PRO GUI is an advanced user interface which reveals all of the controls existing within the FM and HD processing structures. This GUI is designed and intended for advanced processing users, those who are quite familiar with audio processing terms and understand audio processing concepts, the interactions of processing controls, and how to apply that knowledge to achieve the individual sound they desire.

The X1 Pro GUI allows you to adjust the processor and see, in real time, how it is reacting to the audio being processed.

The X1 Pro GUI software may control an unlimited number of X1's via a standard Ethernet network. Utilizing TCP and UDP protocols, the Pro GUI can be used to control the processor from anywhere in the world as long as there is an Internet connection. This includes controlling it behind firewalls, NAT routers and VPN tunnels.

## Getting Started

The X1 Pro GUI is a Windows™ based program and is supplied on the product CD as an ex- excitable program. Though the GUI may be installed to any folder on the host computer, the default path is: “C:\Program Files\Wheatstone\AirauraX1.” After installation, the GUI can be started by clicking on the X1 Pro item in the Windows Start menu.

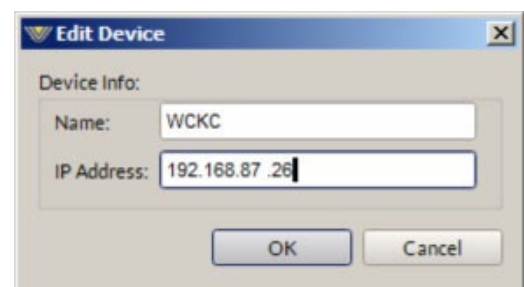
The GUI is designed to run on all PC running Windows XP, Vista, 8 or 10. The GUI may not be compatible with tablet based Windows operating systems. At this time, the GUI is not supported on Apple based platforms, however it has been known to work on some versions of virtual Windows running on Apple platforms.

## Connecting With The Pro GUI

The X1 and the PC running the X1 Pro GUI program can be straightforwardly connected together over a standard Ethernet Local Area Network (LAN).

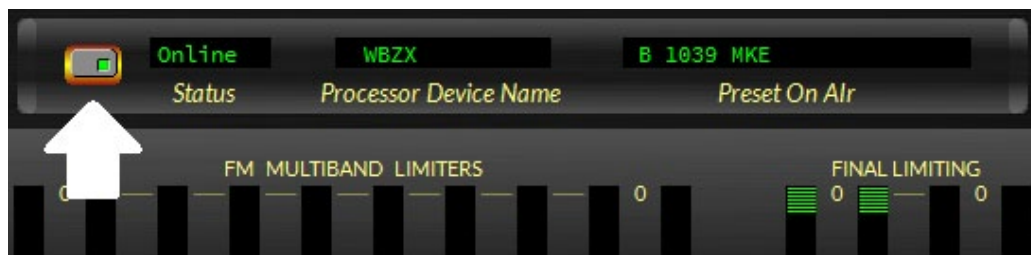
The X1 should be installed, powered-up, and verified to be operating normally. A CAT5 Ethernet cable connected to the rear panel 100baseT LAN port should be connected to the Ethernet port on the host PC. The X1 is Auto-MDIX so either a straight through or crossover cable may be used.

You will need to know the HARDWARE IP address you assigned to the hardware. This is the IP address you will use to connect with the GUI. Assuming the GUI is now installed, start the program to bring the GUI on the screen. Next, on the lower right side of the GUI, under HARDWARE, locate the Devices button and click on it. This will open the Edit Device dialog which is used to tell the GUI what IP address and name the X1 may have.



The Name field can be left blank or used to give the X1 a unique name in the Devices display box at the top of the Pro GUI – this is the device that the GUI is either currently connected to, or configured to connect to if it is not yet connected. In the IP Address field enter the “HARDWARE” address that was previously assigned to X1 and then click OK.

Highlight the new X1 device in the Devices list and click on the Select button. Now you can place the GUI online by clicking on the virtual button to the left of the Status message.

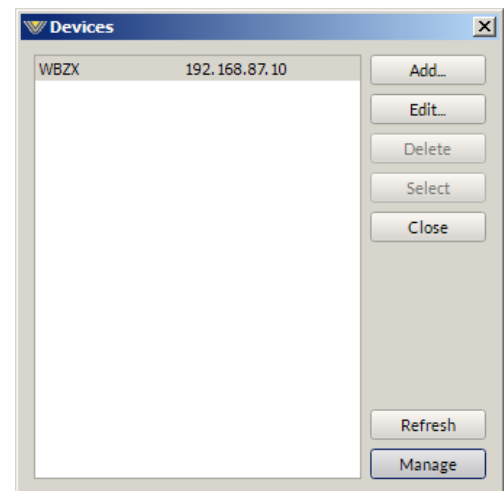


The virtual button will be green when the X1 connects to the hardware indicate that the GUI is now communicating with the X1. Once the GUI has been made aware of the X1's existence in this manner, it will always appear in the list of X1 devices and it will be instantly accessible from anywhere, any time.

While connected to an X1, the status bar will indicate Online and the “Devices” text will show the name of the X1 that it is currently communicating with (in this example, the X1 has been named WBZX). The status bar will also display the name of the preset that the X1 you're connected to is currently running.

When the Preset name is showing as green text it means that a factory or user preset is running and that no changes have been made to it. If the Preset name is showing in orange text, it means that the preset has had some changes made to it or is a work in progress that has not yet been saved to the processor.

Status Bar with preset in orange, indicating that the preset has been altered and not saved.



## Connecting Directly Without a LAN

You can work without a LAN by connecting the X1 and the PC Ethernet ports together with a standard Ethernet cable – either a straight through or crossover cable may be used. As mentioned previously X1 has auto-sensing, or Auto-MDIX Ethernet ports.

***NOTE: The controlling PC and the X1's network settings must be configured to place them both on the same subnet! This is also why the front panel GUI and remote hardware GUI must be in the same network address space (usually one IP address from each other).***



## The X1 and Internet Security Concerns

The X1 does not act as a “web device” or “web server,” nor does it support open ports to the Internet. Therefore there is no worry that someone could “hack” into the X1 and use it as a pathway to the rest of the network to which it is connected.

## About DHCP and The X1

The X1 does not utilize DHCP – Dynamic Host Configuration Protocol (or automatically assigned network addressing) – and needs to be assigned a “STATIC” network unique IP address. Most DHCP servers assign addresses starting at the bottom of the group 192.168.0.xxx. Therefore choosing an address high in that group, say 192.168.0.200, as a static IP address will likely keep it out of the way of the busiest DHCP addressing.

However, it’s always wise to check with your network administrator to determine what static IP addresses may be open and safe to use for AirAura. Sometimes the local network warrants the Subnet and Gateway values to be vastly different, but our suggested starting values for the Subnet and for the Gateway usually suffice for all but the most complex situations.

## AirAura Network Protocols and Ports Used

The X1 uses both TCP and UDP protocols to communicate with the remote GUI. TCP is used from GUI to the X1 because its high reliability ensures that all control changes sent to the X1 will be received with 100% accuracy. The UDP protocol is used by X1 to send and update metering and other real time data back to the GUI. The TCP protocol uses port 55899. The UDP Protocol uses a port in the range between 60000 and 60010. It tries 60000 first and if it is busy it tries 60001 next. This process repeats if necessary until an idle UDP port is found. Note that X1 requires these ports to be open through any firewalls for successful connectivity.

## Using The X1 Pro GUI



The GUI may be positioned on the host computer's screen by left-clicking-and-holding the "Wheatstone" logo, and dragging the GUI to the desired position. In normal "Windows" fashion, the GUI may be minimized on the desktop (taken off the screen) and/or closed by way of the familiar controls at the extreme top right.



The X1 Pro GUI is a windows based application that connects to the processing hardware and expands on the controls featured on the front panel of the X1. The GUI is divided into two parts. The top half shows metering and the bottom half contains the adjustment controls.

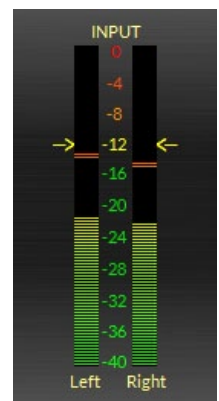
## Metering

This section of the manual will first dive into the metering displays, then show the various controls that govern the sound of the audio processor

### Input Meters

The input meters show the level of the active audio source (Analog, WheatNet-IP or AES). The meters are scaled in dBFS with average and peak levels indicated. The arrows at -12dBFS indicate the maximum recommended peak level. Average program level should be between -24dBFS and -20dBFS.

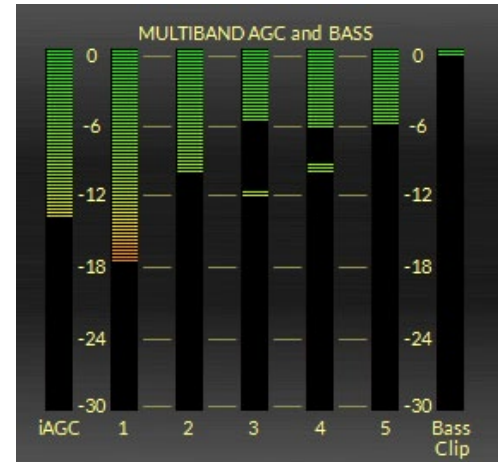
The X1 has a dynamic range of >120dB, thus there is no need to run the input "hot" in order to maximize the signal-to-noise ratio.



## Multiband AGC and Enhance Meters

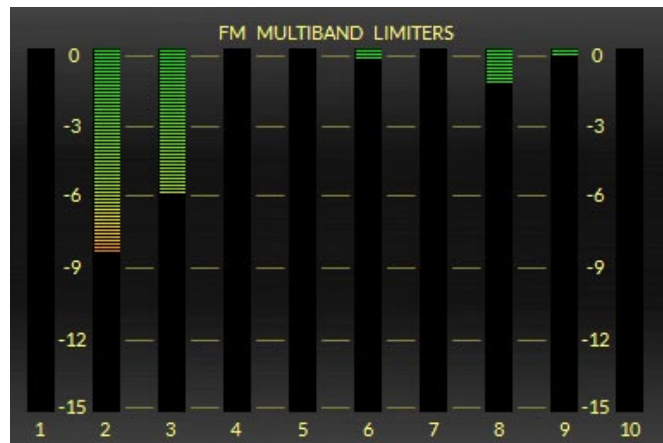
To the right of the input meters are seven meters at the heart of the processing. From left to right they are **iAGC Depth**, the five band section which includes **Low Bass Gain Reduction**, **Mid Bass Gain Reduction**, **Midrange Gain Reduction**, **Presence Gain Reduction**, **High Gain Reduction**, and **Bass Clipper Depth**.

The iAGC Depth and Bass Clipper display a constant bar that shows the amount of gain reduction. The five band metering has two different segments. The constant bar shows total gain reduction in any band, while the “peak lines” show additional compression applied to that particular band. If more aggressive processing is used, the more frequent those “peak lines” will be seen.



## 10-Band Limiter Metering

The output of each of the five bands in the multiband AGC/compressor section is peak controlled by two limiters for precise yet inaudible control. Because the limiters work in conjunction with the settings in the five band section as part of a system we call **UNIFIED PROCESSING**, there is no need for any type of audio gating which is found in more traditional processors. Generally, the 10-band limiter catches only peak overshoots from the five band AGC/Compressor by a few dB. Under normal operation, the gain reduction meters of each band should **RARELY** indicate more than 3dB of gain reduction. In some presets that are not aggressive, it is quite possible not to see **ANY** gain reduction at times.



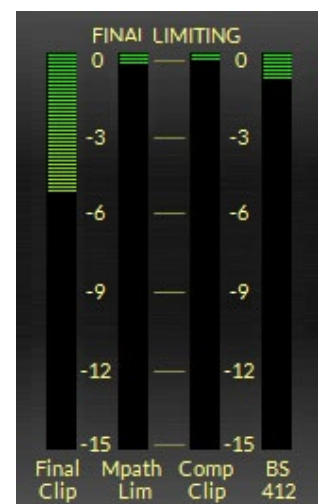
The default display is for the FM multiband limiter. To view the HD limiter, there is an option to switch the metering located below the HD Output meters on the right side of the GUI. When in View HD, the 10-band HD limiter will be shown.

## Final Limiting Meters

The final limiting meters show the gain reduction depth of the **Final Clipper** (or **HD Limiter** if View HD is checked), **Multipath Limiter** (Stereo Width), **Composite Clipper**, and **BS.412 Loudness Manager**.

It is normal to see between 6dB and 9dB of gain reduction in the final clipper as well as up to 3dB on composite clipping on aggressive audio when in View FM mode. To see the final HD limiter metering, switch to View HD. The option to switch between View FM and View HD is located below the HD output meters on the right side of the GUI.

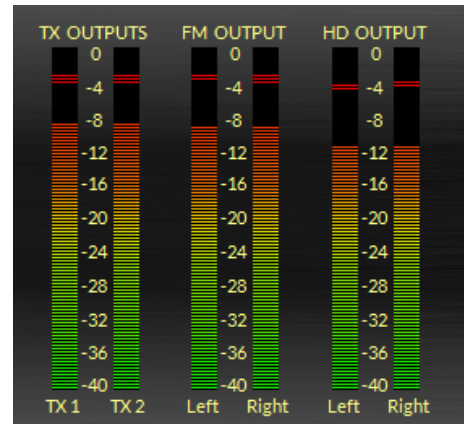
The multipath limiter is a dynamic stereo controller and the meter shows how much reduction of the stereo sound field is occurring in dB at any given time.



The BS.412 meter shows how much gain change is being made by the loudness manager. The BS.412 loudness manager should only be used in regions that require such a system. If your governing body does not require the BS.412 loudness manager, turning it on will reduce overall loudness in the long term.

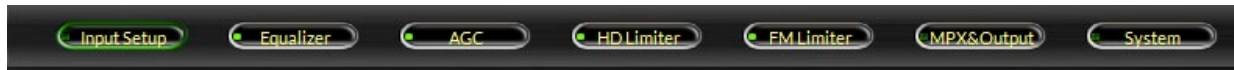
## Output Metering

The final three sets of meters on the right side show the output levels of each of the outputs options. The first set are the TX1 and TX2 outputs. These are relative and the processor output should be set by observing a modulation monitor. The next set is the L/R output level of the FM AES output, followed by the L/R HD AES output.



## GUI Adjustment Controls

Between the metering and the GUI controls lie a row of seven buttons. These buttons open the menus to the various stages of processing for you to adjust. They also provide a sort of signal flow chart for the processor. The menu options are **Input Setup**, **Equalizer**, **AGC**, **HD Limiter**, **FM Limiter**, **MPX & Output**, and **System**.



This section of the manual will explore the different menus, in order, and explain the controls and options contained within each menu.

## INPUT SETUP MENU



The X1 input menu contains all of the input source, level and conditioning filters for the processor. Each menu section also contains a column to the far right that, at any time, allows you to access the preset list and preset library, switch between the FM and HD metering view as well as launch the analysis window or device manager. We will cover those options later in this manual as this section pertains only to the controls that govern the input settings.



## Source Selection and Automatic Failover

The Input Source can be selected as either Analog or Digital. There is one analog input source via the X1's balanced audio inputs on its rear panel. There are two types of Digital sources; standard AES/EBU and WheatNet-IP.

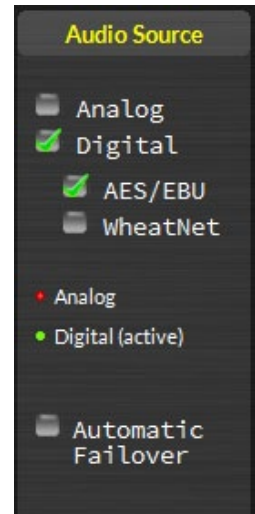
Below the source selection checkboxes are a pair of signal presence indicators. The Analog indicator is green if there is a signal present on both left and right channels and it is higher than -24dBu. The AES indicator turns green when there is a valid AES signal present at the digital input and it is above -48dBFS.

In the event the currently selected input source fails and the Input Failsafe checkbox is checked, an alternate source of audio which has been connected to the alternate audio input can automatically be put on the air.

If the primary source was digital (AES/EBU or WheatNet-IP), the analog input will be selected immediately if there are invalid bits in the data stream or missing audio data. Also, the analog input will be selected after 30 seconds of a valid digital data stream having signal levels below -48dBFS.

If the primary source was analog, then a silence sense timeout of 30 seconds must elapse before the unit switches to the digital input. The audio failure sense threshold is fixed at -24dBu, and this combination with the 30 seconds timeout is suitable for virtually all program types.

**NOTE:** While the audio failover is available between the analog input and one of the digital input options (AES/EBU or WheatNet-IP), audio failover is NOT available between digital modes. For instance, if you are using WheatNet-IP as your primary input source, you cannot have the processor fail back to AES/EBU as the backup source. It must fail back to analog.



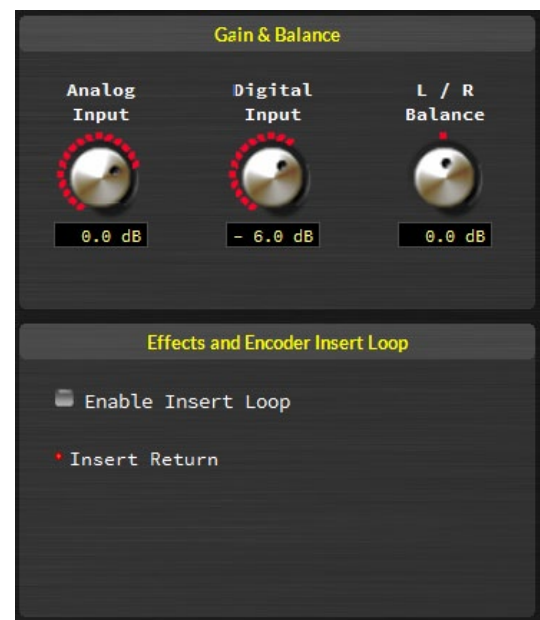
## Input Gain, Balance, and Input Loop

### Analog Input Gain

The Analog audio input gain can be adjusted by clicking on the knob and dragging the mouse or touch-pad to the right or left. The Analog Input Gain level can be adjusted over a +/- 24dB range. To adjust the gain, left click on the knob and use the mouse wheel to increase or decrease the gain. You can also double click on the current value and type in the new value.

### Digital Input Gain

The Digital audio input gain can be adjusted by clicking on the knob and dragging the mouse or touch-pad to the right or left. The Digital Input Gain control affects both AES3 and WheatNet-IP and gain can be adjusted over a +/- 24dB range. To adjust the gain, left click on the knob and use the mouse wheel to increase or decrease the gain. You can also double click on the current value and type in the new value.



## L/R Balance

Static level errors in Left/Right channel balance can be corrected by using the L/R Balance knob. The Left/Right balance control affects all input sources and can be adjusted over a +/-12dB range. Usage of this control should be restricted to short term “band-aid” use only since Left/Right channel balance is best corrected upstream of AirAura if it is out of balance. To adjust the gain, left click on the knob and use the mouse wheel to increase or decrease the gain. You can also double click on the current value and type in the new value.

## Enable Insert Loop

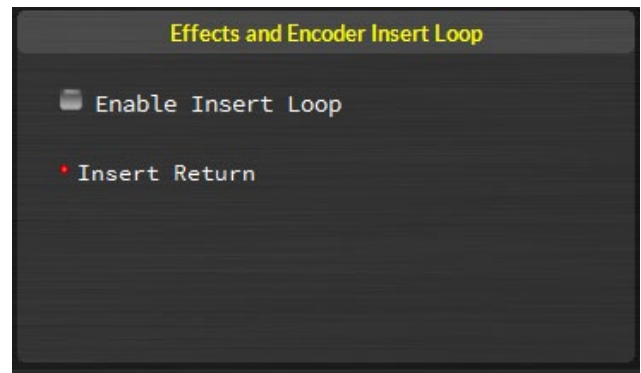
Enabling the insert loop breaks the audio connection between the output of the five band processor and the input to the FM and HD limiter. Instead of directly connecting the five band output to the FM and HD limiter, the audio from the 5 band is sent to the insert loop OUT jack on the rear panel of the X1. This allows you to insert a ratings encoder watermark (or other gear). The output of the watermark encoder (and associated gear) is then returned to the IN jack. At that point, the audio signal continues to the FM and HD limiters, peak controllers and stereo generator (on the FM side).



Using the insert loop allows the user to have the X1's iAGC and five band processing appear before any ratings encoder so that the encoder is fed a consistent level.

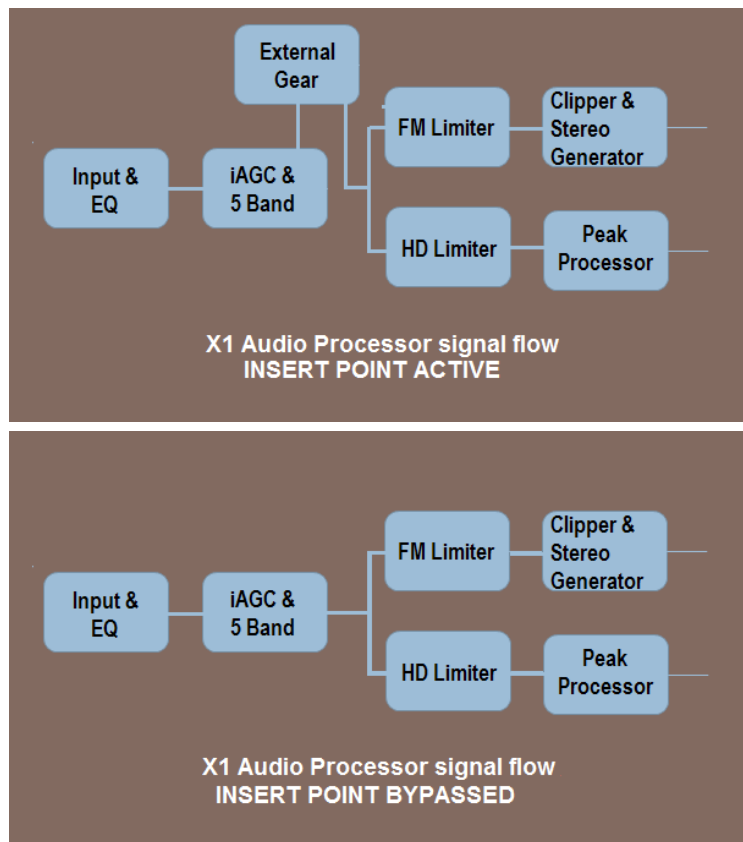
Prior to this option, many users resorted to external AGC's to feed the ratings encoder prior to the main processor. Adding an additional gain control device to the airchain had the tendency to cause audible fighting between the external AGC and the internal AGC in the main processor. Defeating the AGC in the main processor was not always an option.

The correct solution is to provide an insert point in the main processor's audio path to accommodate the ratings encoder. This way, not only do we have one AGC that controls the input level to the processor AND the ratings encoder, we can now place the ratings encoder towards the back of the processing so the watermark is less likely to be altered by any previous multiband stages.



**NOTE: Make sure you have your audio devices properly set up before enabling the insert loop or you will have an on air audio failure. The INSERT RETURN indicator will change to green if two conditions are met... AES lock and audio present.**

**The insert loop accepts AES/EBU signals only. The default sample rate is 48kHz. If you desire the sample rate of the loop to be 44.1kHz, you will need to feed a 44.1kHz signal to the main AES input of the X1. As of this writing, there is no audio failover for signal loss in the insert loop.**



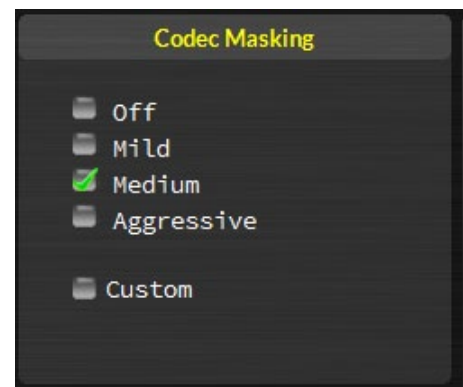
The insert loop is send and return levels are fixed. The send level is normally within a target of -24dBFS to -20dBFS, depending on how aggressive the processing is set up in the 5 band. If you are only looping the ratings encoder, the return audio should be at the same amplitude as when it left the processor. If you are adding any other devices into the insert loop, refer to their documentation for proper audio level line up procedures.

The insert loop option is preset dependent IF the option for “Input Settings Follow Preset Takes” is selected. If not, the input loop option is not preset dependent. For a list of input options that can follow preset takes, see Chapter 2, page 2- 23.

## Codec Masking

This feature can help the overall sound of the radio station when lossy audio in stereo is being presented as the source on a consistent basis (either thru compressed audio files or if the programming is via satellite or some other non-linear transmission mode). While this is not a “cure all”, it can help to make the transition from lossy to linear less noticeable. For most situations, the mild or medium settings are usually sufficient. Aggressive should only be used in extreme conditions.

The setting that is right for you will be determined after critical listening for some time. It should be noted that in the Aggressive setting, some stereo separation will be sacrificed due to the nature of how the codec mask algorithm works, so Aggressive should best be left for talk formats delivered over satellite.



The custom option is normally not active. This option is only selected when a Wheatstone Processing technician has made a custom setting for a customer in the codec mask. Once a custom setting is made, the custom option will be active. The only way to go back to the user presets is to re-take a factory preset .

## High Pass Filter

The High Pass Filter is used to remove inaudible and unnecessary subsonic energy from the audio signal prior to it being processed. By removing this energy, processing is cleaner and modulation energy is not wasted by transmitting sounds that will not be perceptible to a listener.

The High Pass Filter also removes subsonic energy that could upset the operation of equipment later in the audio chain, such as the AFC loop in the STL or FM exciter.

The control range is 20Hz to 300Hz. To adjust the value, left click on the knob and use the mouse wheel to increase or decrease the frequency. You can also double click on the current value and type in the new value.

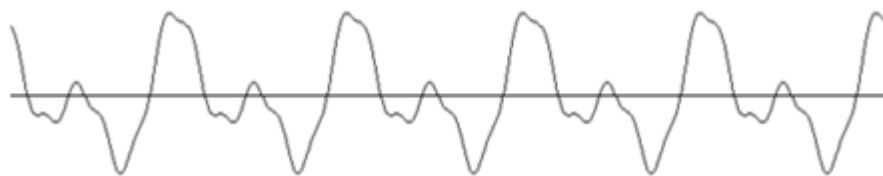
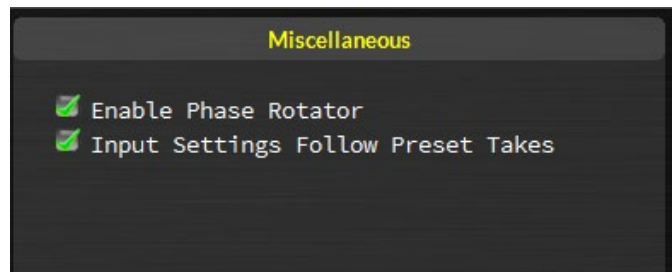


## Miscellaneous

The final two options in the INPUT menu are the Phase Rotator and the option for input settings to follow preset takes.

### Phase Rotator

Human voice is usually asymmetrical by nature, which means that it is usually “peakier” in one polarity than the other, hence asymmetrical. What the Phase Rotator does is fix these asymmetrical peaks, and the way it works is by changing the phase of signal harmonics compared to their fundamental frequencies. This action reduces the peak to average ratio and enables an increase in apparent loudness with minimal audible detriment. We do this because processing the audio for a symmetrical medium like FM is greatly simplified if the waveforms are symmetrical.





## Input Settings Follow Preset Takes

By default, when the X1 changes presets, its input settings (levels, input source, balance) are static. If you wish to have special input levels for different presets, or change any of the following, you will need to check this option and then save your current preset with the adjusted input settings.

The following settings are allowed to change when this special option is checked:

- Current Audio Source (Analog, Digital or WheatNet-IP)
- Analog Gain
- Digital Gain
- Balance (common to both Analog and Digital Inputs)
- Phase Rotator
- Insert Point Loop

The X1 also has a set of output controls which can be dependent on preset takes. The option for that can be found in the MPX& Output menu and a list of controls that can be changed when that option is selected can be found on page 2-23, Chapter 2.

## EQUALIZER MENU



The four band equalizer has four independent bands. Band 1 and 4 are shelving filters while Bands 2 & 3 are fully parametric. The two parametric bands can be adjusted in the following way:

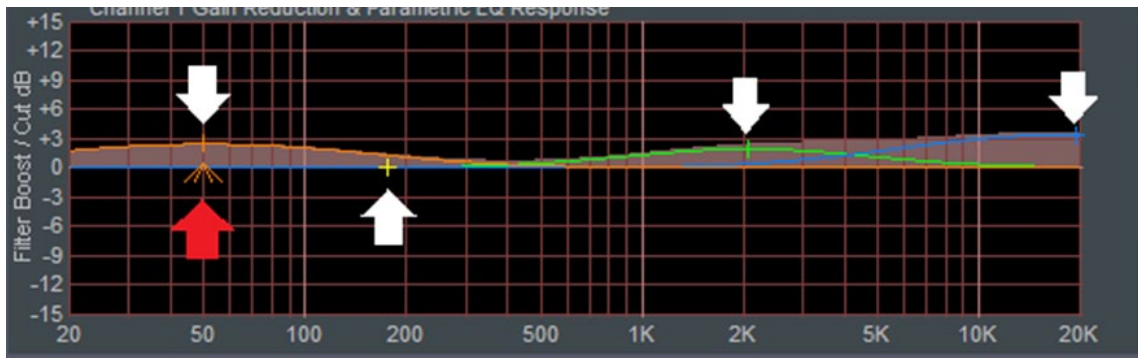
- Center Frequency** – 20 Hz to 20 kHz.  
**Bandwidth** – 0.2 to 3.0 octaves.  
**Boost/Cut** – +/- 14.0dB.

The Band 1 Shelving EQ has a frequency range of 20-500Hz and can have a boost or cut of 14dB. The Band 4 Shelving EQ has a frequency range of 2kHz to 20kHz and can have a boost or cut of 14dB.

The controls of the equalizer may be manipulated in three ways:

1. The first is by directly clicking on and dragging the control sliders in the parametric equalizer control screen. As the controls are manipulated, numerical representations of the control settings appear in the boxes below the band that is being adjusted.

- The second method of adjusting the parametric equalizers is by directly manipulating the actual curves on the graphical screen below using the mouse. Double clicking on the small crosshair at the top (or bottom if it's below the line) of the curve and holding the left mouse button down as you drag the mouse will move the curve to any frequency (left or right move) as well as adjust the curve's height above or below the 0dB reference line (up or down move). If you are adjusting the two center parametric bands, you can also click on the small "tent" underneath (see red arrow in the example below), and holding the left mouse button down while sliding the mouse left or right allows changes the bandwidth, or "Q," of the equalizer section.



- The third method is by direct value entry. By double clicking on the value boxes underneath each slider, a dialog opens up which allows keyboard entry of the operating values for each equalizer section. Once the value has been typed into the entry box, clicking the OK button or hitting the keyboard Enter key will make the value active. Note that the direct entry method is controlled by a lookup table of all possible values. If an entry is made that is not in the lookup table, the closest lookup table entry will be made active.

## EQ Enable

This is the master Equalizer In/Out button. When this box is checked the equalizer section "IN".

## iAGC AND COMPRESSOR

The X1 has two sections dedicated to dynamics control – the iAGC and the multiband leveler/compressor. Both are linked by DSP algorithms which logically connect and control them. We call this system “iAGC”.



Navigating the iAGC/Compressor menus is fairly easy. To the left are eight tabs. Each tab hosts the controls for a specific part of the iAGC/Compressor. The controls under each tab are split into two categories: BASIC and EXPERT.

The BASIC controls group the adjustors that make the most impact on the sound in an easy to access location. Most people will not need anything more than the controls in the BASIC category.

The EXPERT controls are for more subtle adjustments that should only be touched in unique situations where a specific sonic texture is desired.

It should be noted that the output of the iAGC/Compressor section directly feeds the multiband limiters of both the HD and FM audio paths. This is the last section of the processing that is shared between the two paths.

The output of the iAGC/Compressor can also be sent to the INSERT LOOP SEND XLR jack on the rear of the X1 so that a ratings encoder and associated gear can be inserted after iAGC and spectral control but before limiting on both the HD and FM audio paths.

This section of the manual will take each tab and describe the adjustors within. The tabs for Bands 1-5 will be in one section as the controls duplicate for each of those tabs.

## iAGC

Because of its nature, the iAGC only sports the BASIC control set. Much of the iAGC decision making is in real time based on measurements of amplitude, density and user settings in each of the five band compressors. Thus, only very broad controls are available to adjust as the actual values of the settings “behind the curtain” are constantly adjusting and updating.

### Window Size

Sets the amount of gain change before the iAGC adjusts the audio level. This control is entirely based on the amplitude of the audio and does not affect the dynamics measurement. Lower settings (Counter Clockwise) will make the iAGC work more like a traditional wide-band AGC. Higher settings allow the iAGC to do less dynamics control on changing levels. The range of the control is 0 to +8dB. To adjust the value, left click on the knob and use the mouse wheel to increase or decrease the window size. You can also double click on the current value and type in the new value.



### Attack Speed

Changes the attack speed when the audio falls outside the iAGC window. Faster values mean a quicker change in amplitude. The control range is 2 to 7 seconds. To adjust the value, left click on the knob and use the mouse wheel to increase or decrease the speed. You can also double click on the current value and type in the new value.



### Release Speed

Changes the release speed when the audio falls outside the iAGC window. Faster values mean a quicker change in amplitude. The control range is 2 to 7 seconds. To adjust the value, left click on the knob and use the mouse wheel to increase or decrease the speed. You can also double click on the current value and type in the new value.



### Adapt Mode

Determines if the gain change in the iAGC is initially quick or slow. In the ADAPT mode, a drastic level change will make the iAGC work faster to keep up with the level. In the LINEAR mode, the iAGC will make gain changes uniformly no matter what the input level is. To change the mode, left click on the knob and use the mouse wheel to switch the mode. Left click again to confirm the change.



## iAGC Gate Threshold

Sets the level at which the iAGC will no longer adapt or change the audio level. The range is -60dBFS to -35dBFS. To adjust the value, left click on the knob and use the mouse wheel to increase or decrease the threshold. You can also double click on the current value and type in the new value



## Return To Zero (RTZ)

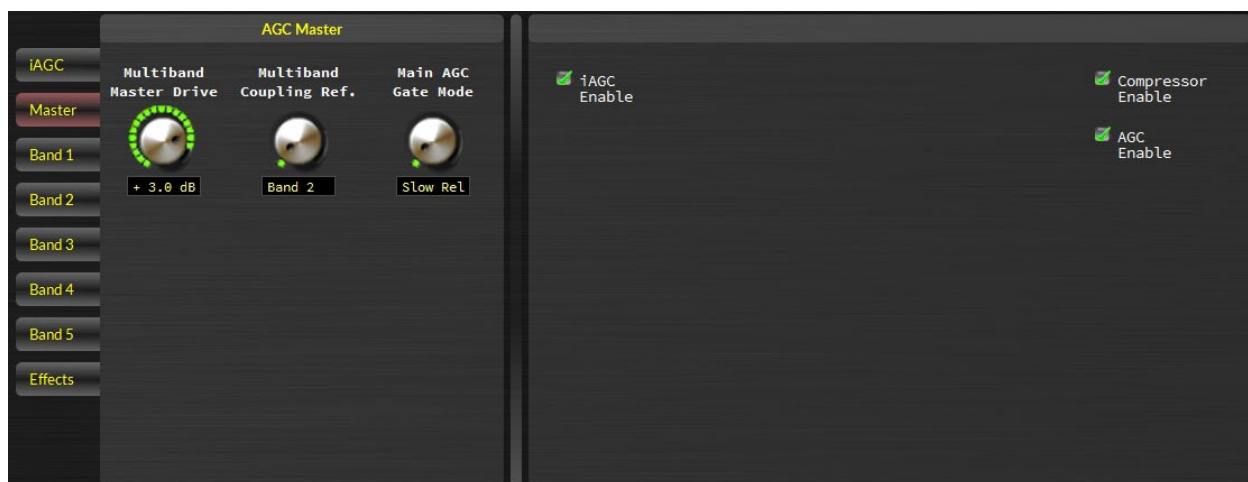
When audio falls below the iAGC gate threshold, this option determines if the iAGC gain leaks off to “zero” (in this case, the iAGC “zero” is actually -12dBFS of gain). To turn this option on or off, left click on the knob and use the mouse wheel to switch the mode. Left click again to confirm the change.



## Master

The MASTER tab shows the controls that apply to ALL five bands of the multiband AGC/compressor section. They include the drive to the five band section, the coupling band reference control and the gate mode for the five band AGC.

To the right of the master controls are the options to enable or disable some or all of the iAGC and five band AGC/Compressor.



## Multiband Master Drive

The Multiband Master Drive control adjusts the amount of signal at the input to the iAGC and Leveler/Compressor which then determines the depth of gain control achieved. Driving the leveler harder (higher drive numbers) results in a more consistent on air sound along with an increased ability to bring up low passages in program material when needed. Less drive creates a more “gentle” on air sound which then has less capability to bring up low passages. The range of this control is -12.0dB to +6.0dB. To adjust the value, left click on the knob and use the mouse wheel to increase or decrease the gain. You can also double click on the current value and type in the new value.





## Multiband Coupling Reference

This control assigns the reference band from which the other bands “follow” when adding or reducing gain. When Band 2 is selected, the audio tends to be warmer with better defined bass. When Band 3 is selected, the audio is more uniform and doesn’t favor one part of the spectrum over another. To change this option, left click on the knob and use the mouse wheel to switch the mode. Left click again to confirm the change.



## Main AGC Gate Mode

There are two different modes available for tailoring the behavior of the AGC gate during periods of silence or low audio levels when the audio is below the currently set Gate Threshold.

When SLOW RELEASE (Slow Rel) is selected and the audio input falls below the Leveler Gate Threshold, rather than the bands “holding” their current gains, their gains will slowly increase towards 0dB.

When Hold is selected and the audio falls below the Leveler Gate Threshold the gains of the five bands will “freeze” at their current values. Their gains will hold at those values until audio is above the Leveler Gate Threshold again. The Hold mode prevents the Leveler from increasing its gain in the absence of audio which minimizes the increase of background noise.

To change gate mode, left click on the knob and use the mouse wheel to switch between the two modes. Left click again to confirm the change.



## Bypass Options

The final controls under this tab are checkboxes to enable the iAGC, five band Compressor and five band AGC. Under normal operations, all three should be checked. Disabling the iAGC allows the X1 to operate like a “traditional” audio processor where the five band AGC and Compressor will increase and decrease gain based only on amplitude and spectral balance, and program density is not measured.



Bypassing the five band AGC ( unchecking AGC ENABLE) will also defeat the iAGC as well and leave just the compressor in line. The result will be very dense audio. Defeating all three options bypasses everything in this section. At that point, unprocessed audio is passed to the 10-band limiter.

Wheatstone suggests using the X1 as intended with all sections engaged. Because the iAGC measures the dynamic range, if any processing is used before the X1, the iAGC will “detect” the reduced dynamic range and adjust accordingly. Thus, even with outboard AGC’s, the X1 still works best when its iAGC is enabled.

## Band Tabs (1-5)

These controls are the heart of the X1's sonic signature. Spectral balance, density and a great deal of level control happens in these five tabs. Each of the five tabs have the same controls (with the exception of BAND 5. BAND 5 does not have a crossover control as it is at the top of the frequency spectrum).



### Density (Basic Control)

The Density control adds more energy to the processing band being adjusted as it is increased. It basically changes the relationship of the audio controlled by the AGC vs the compressor. Lower numbers (0-2dB) let the AGC do more of the audio control for a more open sound. This is the desired range for Classical, Jazz and Public Radio style formats.

Higher numbers (2dB-6dB) place more of the burden on the compressor. In the 2dB-6dB range, the audio is more consistent in both level and EQ, but is also louder and more “competitive”.

The range of the control is 0-6dB in 0.5dB increments. Recommended settings for most formats are +3dB for bands 1 and 2, +2 for Band 3 and 0 or +1 for Bands 4 and 5. To adjust the value, left click on the knob and use the mouse wheel to increase or decrease the density. You can also double click on the current value and type in the new value.



### Band Mix (Basic Control)

The Band Mix allows you to increase the OUTPUT level of the band you are adjusting. It is part of the final output mixer of the entire five band section and away to further shape the equalized sound of the five band AGC/Compressor AFTER processing as been applied.

The band mix control for each of the five bands of processing can be found in two places. Under the tab of the respective band being adjusted and under the EFFECTS tab (which will be discussed later in this section). The two controls track each other and have a boost or cut of +/- 6dB).

To adjust the value, left click on the knob and use the mouse wheel to increase or decrease the gain. You can also double click on the current value and type in the new value.



## Master Attack (Basic Control)

The Master Attack control sets the overall speed of the attack for each of the five bands of processing. Lower numbers indicate the band will make corrections much faster, extremely important if you are processing for competitive audio. Higher numbers will lead to a slower response time to correcting audio and is more suited for formats that don't require aggressive processing. The range of the control is 50mS to 3000mS (or 3 seconds). The recommended setting for most formats is 300mS.

To adjust the value, left click on the knob and use the mouse wheel to increase or decrease the speed. You can also double click on the current value and type in the new value.



## Master Release (Basic Control)

The Master Release control sets the overall recovery speed of each band after gain has been corrected. Lower numbers indicate the band will recover faster (and, thus, be more audible) while higher numbers will lead to a slower recovery time and a more open quality to the audio (at the expense of instantaneous correction).

The range of the control is 100mS to 7000mS (7 seconds). If iAGC is used, the recommended release time is 5000mS-7000mS. If iAGC is disabled, a faster recovery time will probably be desired (2000mS-5000mS).

Even in the most competitive situations, Wheatstone does not recommend running the Master Release control below 1000mS in any band unless you are specifically looking for a certain signature.

To adjust the value, left click on the knob and use the mouse wheel to increase or decrease the speed. You can also double click on the current value and type in the new value.



## Band Threshold (Expert Control)

The Band Threshold control determines at what level processing will begin to take effect in a particular band. Higher threshold settings cause processing to begin at higher levels and lower threshold settings cause processing to begin at lower levels. The range for the X1's threshold controls range is -80 to -50dBFS.

As the band threshold control is an expert control, the recommended setting for this control is preset dependent. Instead of adjusting this control, Wheatstone recommends you adjust the Band Mix control instead if you are looking to add or reduce audio in a particular band.

To adjust the threshold, left click on the knob and use the mouse wheel to increase or decrease the value. You can also double click on the current value and type in the new value.



## Ratio (Expert Control)

The Ratio determines the how tightly the multiband processing in this section controls the audio. The range of adjustment is from 2:1 to 6:1. The Ratio for each band may be set individually, and the Ratio settings apply to both the Leveler and Compressor simultaneously.

You will find that the ratio control is set to around 3.00:1 in most bands in most presets. This is the recommended settings for this control and, under





normal operation, you should not have a need to change this control.

If you find adjustment is necessary, left click on the knob and use the mouse wheel to increase or decrease the value. Double clicking and typing a new value is disabled for this control.

## Band Coupling (Expert Control)

Each band of the five band AGC/Compressor can be offset from its neighbor by using the Band Coupling control.

Each Band Coupling control may be adjusted over the range of 0-6dB of decoupling (Band 1 has the option to decouple by 9dB). When all the coupling controls are set to 0dB, only Band 2 can take on more gain (if Band 2 is the key band, if Band 3 is the key band, that band then is the only band that can add gain). All of the other bands will remain “flat”. Such a setting is useful in formats such as Classical and Jazz that typically would not enjoy the automatic spectral balance “correction” that more processed formats usually prefer.

**In most programming situations the optimum setting of the Coupling controls will likely be somewhere between -1.0dB and -3dB for bands ABOVE the key band and -2.0dB to -6dB for bands BELOW the key band.** Decoupling the LEVELER’s bands beyond -6dB can allow their gains to “wander” too much, causing unnatural spectral buildup on some material. Therefore the maximum decoupling is 6dB (except for Band 1, which can decouple to -9dB for added bass)

Normally, the EQ and iAGC settings in the preset dictate how much band coupling should be allowed, thus Wheatstone recommends not changing this setting. If, however, you find you NEED to adjust the band coupling, left click on the knob and use the mouse wheel to increase or decrease the value. You can also double click on the current value and type in the new value.



## Compressor Attack (Expert Control)

The compressor attack control dictates how quickly the embedded compressor in each band will reduce gain on “peaky” audio that is not corrected by the slower AGC. Faster attack times will yield tighter level control at the expense of a more dense sound. Slower attack times allow more peaky audio to pass on to the limiter stages downstream.

As with the other controls in the expert group, the compressor attack has been carefully set based on a number of other factors. If you find you need to change the compressor attack, left click on the knob and use the mouse wheel to increase or decrease the value. You can also double click on the current value and type in the new value. The range of the control is 10mS to 1000mS (1 second).



## Compressor Release (Expert Control)

The compressor release determines how quickly the embedded compressor recovers after making a correction to the audio. Faster release times add loudness to the audio (at the expense of more audible processing). Slower release time allow for a smoother sound to the audio that tends to be more open.

As with the other controls in the expert group, the compressor release control has been carefully set based on a number of other factors. If you find



you need to change the compressor attack, left click on the knob and use the mouse wheel to increase or decrease the value. You can also double click on the current value and type in the new value. The range of the control is 50mS to 3000mS (3 seconds).

## Crossover Control (Expert Control)

Bands 1 through 4 have an extra control that sets the crossover (turnover) frequency between two bands of processing. The AGC/Compressor is a five band design utilizing phase linear Linkwitz-Riley crossover filters. Careful filter design eliminates unnatural spectral coloration when deep gain reduction is occurring in one or a few bands.

### Crossover Frequency Options

Band 1 to Band 2: 50Hz - 153Hz

Band 2 to Band 3: 223Hz - 687Hz

Band 3 to Band 4: 1.2kHz - 2.0kHz

Band 4 to Band 5: 4.49kHz - 7.55kHz



As with the other controls in the expert group, the crossover frequencies have been carefully set based on a number of factors. If you find you need to change the crossover frequencies, left click on the knob and use the mouse wheel to increase or decrease the value. If you make an incorrect entry, the control will round to the next available value.

## Effects Tab

The effects tab includes the final output mix for each of the 5 AGC/Compressor bands as well as the stereo enhancement options.



The top row of controls on the effects page is the output gains of each band. These controls are ganged to the same output mix controls on the individual band adjustor pages. They are included in both places for easy access. If you're fine tuning a particular band, it is handy to have the band mix control on the same page. On the other hand, if you just want to play with overall equalization, having a second option of the controls in one place is also helpful. Just remember the controls ARE ganged and will track.

The bottom row of controls is the stereo enhancer controls. These help the user increase the stereo sound field in specific frequency bands as opposed to other processing designs which raise the overall L-R (difference) signal to increase stereo enhancement.

The Stereo Enhance controls can be used in combination to satisfy virtually any style of stereo enhancement. To make the stereo sound field big, wide, and warm, one might slightly reduce the Band 5 Stereo Enhance levels while also slightly increasing the Stereo Enhance 2, 3 and 4 Mix levels.

Or, if you just want to have an overall increase in stereo perception, making increases to just Band 3 and 4 will get you there, without having to raise the entire L-R signal. This approach has a benefit in that it gives the same type of stereo enhancement that you'll find in other processors yet it also minimizes receivers' reactions to multipath interference.

To change the mix value, left click on the knob and use the mouse wheel to increase or decrease the value.

***NOTE: When adjusting the mix controls, keep in mind that these outputs either directly feed the 10-band limiters or, if using the insert loop, the Input of your outboard gear.***

If the insert point is NOT used, the stage immediately following is the 10-band limiters. There is one limiter tailored for FM and one for HD. If you ARE using the insert point, you should follow the instructions on page 3-12 to properly set up this feature.



## HD LIMITER



The X1's HD limiter is custom designed to manage audio ahead of the HD codec. Because of the nature of codecs, the HD limiter and final peak processing does not clip the audio, rather, sophisticated lookahead algorithms are able to apply precise peak control while always maintaining a high quality audio signal.

The HD limiter screen is divided into three sections:

The BASIC controls group the adjustors that make the most impact on the sound in an easy to access location. Most users will not need anything more than the controls in the BASIC category. Bass enhancement and setting the 10-band limiter parameters are located in the BASIC control group.

The EXPERT-LIMITER TUNING controls are for more subtle adjustments that should only be touched in unique situations where a specific sonic texture is desired. These controls fine tune the timing of the 10-band limiter and final lookahead limiter away from the factory preset. Care should be heeded when adjusting these parameters as mis-adjustment can cause overshoot.

The EXPERT – BASS TOOLS TUNING controls fine tune the frequency and texture of the bass enhancement. Again, careful tuning is paramount for the best results.

### Multiband Drive (Basic Control)

Drives the level into the 10-band limiter. It's important to remember that the X1 needs only a few dB of limiting to be loud yet amazingly clean! The drive control is calibrated in percentage (%) where higher values equate more processing for the 10-band section.

This drive control has a secondary function in that it also can offset the level coming back to the processor from the insert loop. Most presets will have the multiband drive between 90-100%. If you find that you have to set this control outside this range, or that 100% does not yield enough drive, evaluate the gain structure of the equipment in the insert point.

To change the multiband drive, left click on the knob and use the mouse wheel to increase or decrease the value. You can also double click on the value and type in a new value between the number 80 and 100.



## Limiter Drive (Basic Control)

The drive to the final lookahead peak limiter is AFTER the 10-band limiter. The final lookahead limiter has a program dependent attack setting for precise peak control, but user adjustable release times are available in the expert mode.

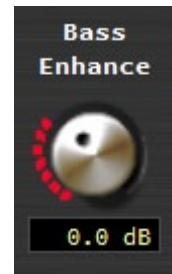
To change the limiter drive, left click on the knob and use the mouse wheel to increase or decrease the value. The drive control is calibrated in 0.5dB steps. You can also double click on the value and type in a new value between -6dB and +6dB. If you make an incorrect entry, the control will round to the next available value.



## Bass Enhance (Basic Control)

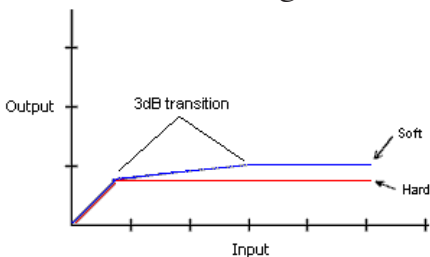
Governs the amount of enhancement driven into the separate specialized bass processor/limiter. The control range is -6dB to +12dB in 0.5dB steps with most presets settling on a value between -3dB to +3dB.

To change the bass enhance drive, left click on the knob and use the mouse wheel to increase or decrease the value. You can also double click on the value and type in a new value. If you make an incorrect entry, the control will round to the next available value.



## Multiband Knee (Basic Control)

Changes the knee of the limiter from soft to hard. When in HARD mode, the limiter thresholds set in the controls in the lower region of the limiter page are absolute... limiting will begin at the control settings. When in SOFT, gain reduction beginning to take place 3dB below a band's actual threshold setting with limiting gradually getting harder as it reaches threshold.



Comparison of Hard and Soft Limiter Modes

In the soft mode, limiting is more obvious but the audio can also be smoother, especially on voice. In the hard mode, limiting is less noticeable but some peaky material may induce clipping faster.

To change Multiband Knee, left click on the knob and use the mouse wheel to switch between soft and hard. Left click again to confirm the change.

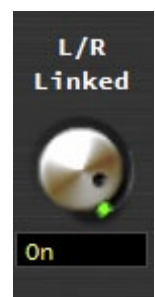


## L/R Linked (Basic Control)

The Lookahead Limiter normally operates with its left and right channels un-ganged, which permits each channel to be peak limited independently of the other. With independent peak limiting program material may be perceptibly louder than when the limiters are linked though the stereo image may not be maintained when one channel limits and the other doesn't.

When the L/R Linked box is checked the left and right channel control signals are made common and the greater of the two controls both channels equally. The linked mode therefore prevents unnatural stereo image shifts when one channel needs to reduce the signal to control peak excursions and the other channel doesn't.

To switch on or off the link, left click on the knob and use the mouse wheel. Left click again to confirm the change.





## Bass Style (Basic Control)

Similar to the 10-band limiter, the bass processor is also a limiter. It can be a hard knee limiter (for tighter bass control) or a soft knee limiter (which allows the bass a fuller and warmer sound). A third option is to turn off the bass processor, which is one of the three options available (OFF, Soft, Hard).

To change the bass style, or switch the bass enhancer off, left click on the knob and use the mouse wheel to choose. Left click again to confirm your choice.



## Master Attack (Expert Control)

The master attack control sets the attack speed for all 10-limiter bands. The normal setting for this control in all presets is AUTO, which means the attack speed is program dependent. Offset from auto are four settings that tell the attack speed to work faster or slower while still factoring in program content.

To change the master attack speed, left click on the knob and use the mouse wheel to increase or decrease the value.



## Master Limiter Release (Expert Control)

The master release control sets the release speed for all 10-limiter bands. The normal setting for this control in all presets is AUTO, which means the release speed is program dependent. Offset from auto are four settings that tell the release speed to work faster or slower while still factoring in program content.

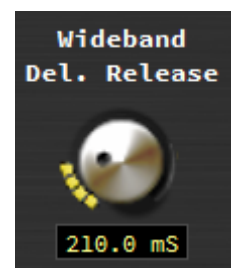
To change the master release speed, left click on the knob and use the mouse wheel to increase or decrease the value.



## Wideband Delayed Release (Expert Control)

The Delayed Release control sets the long term recovery of the lookahead limiter. Delayed Release reduces intermodulation distortion and allows more limiting depth via the Limiter Drive control without incurring a distortion penalty for extra loudness. The control range is 100mS to 1.0 seconds.

To change the delayed release timing, left click on the knob and use the mouse wheel to increase or decrease the value. You can also double click on the value and type in a new value.



## Master Threshold (Expert Control)

The Master Threshold sets the limit threshold for the 10-band limiter. Normally this control should be set to AUTO which allows the threshold to operate in a program dependent way. Changing the threshold will offset the control and make it a static threshold.

To change the threshold, left click on the knob and use the mouse wheel to increase or decrease the value.



### Wideband Threshold (Expert Control)

The wideband threshold for the lookahead final limiter is a static threshold. This control should normally be set to 0dB. Raising the control will allow more overshoot and less peak control. Lowering the control will create tighter peak control at the expense of more audible limiting.

To change the threshold, left click on the knob and use the mouse wheel to increase or decrease the value.



### Filter Frequency (Expert Control)

The filter frequency sets the LPF (Low Pass Filter) for the Bass Enhance Limiter. It is not the “center” frequency for the bass processing, rather all audio below the frequency shown is processed by the Bass Enhancer and Limiter.

Normally there should be no need to change this control. However, if you feel you need to increase or decrease the filter frequency for a different type of bass response, left click on the knob and use the mouse wheel to increase or decrease the value. You can also double click on the shown value and type in a new setting. The range is 40Hz to 400Hz in 10Hz increments.



### Filter Style (Expert Control)

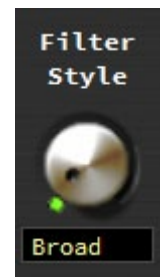
The Filter Style control switches in one of three types of low pass filters for Bass Tools. The style of filter selected can impact the sound of bass.

The BROAD filter shape allows for a very full bass sound. Some bass harmonics are let out into the “main mix” (frequencies above where the Bass Tools frequency control is set) and the broad filter has a texture that is reminiscent of bass enhancers in older digital boxes with the control of a modern limiter/clipper design.

The MEDIUM filter shape is what previous versions of AirAura bass management used and is also found in the VBMS bass enhancement systems found in our other audio processors. The MEDIUM filter shape gives bass some more “heft” without letting as much of the bass harmonics out of the enhancer into the main mix.

The TIGHT filter shape keeps virtually all bass harmonics out of the main mix. It is also the lightest sounding of the three and the bass has a very controlled texture.

To change the filter style, left click on the knob and use the mouse wheel to choose which style sounds best. In order to hear the change, you will have to left click again to confirm the change.



### Bass Clip Threshold (Expert Control)

Sets the threshold level of the bass clipper in the X1. Lower values increase bass clipping and produced very controlled bass. At some point, however, there is a tradeoff between controlled bass and “thin” sounding audio. The bass limiter threshold should be set by ear using content with a good dose of bass material.

To change the control, left click on the knob and use the mouse wheel to increase or decrease the value. You can also double click on the shown value and type in a new setting. The range is 0dB to -12dB. Suggested settings are usually -3dB to 0dB.



## Bass Texture (Expert Control)

Determines the level of upper harmonics of the bass enhancing limiter. Higher percentage (%) values creates more harmonics. Lower numbers produce a smoother sounding bass. This control should be set by ear.

To change the control, left click on the knob and use the mouse wheel to increase or decrease the value. You can also double click on the shown value and type in a new setting. The range is 0dB to -12dB.



## FM LIMITER

The X1's FM 10-band limiter is designed to manage audio ahead of the X1's highly oversampled peak controlling main and composite clippers. Because of the nature of FM broadcast and the need to be highly competitive, the X1 uses some very unique features in this stage to limit audio as inaudibly as possible.



The FM limiter is divided into three sections:

The BASIC controls group the adjustors that make the most impact on the sound in an easy to access location. Most people will not need anything more than the controls in the BASIC category. Bass enhancement and setting the 10-band limiter parameters are located in the BASIC control group

The EXPERT-LIMITER TUNING controls are for more subtle adjustments that should only be changed in unique situations where a specific sonic texture is desired. These controls fine tune the timing of the 10-band limiter and final lookahead limiter away from the factory preset. Care should be heeded when adjusting these parameters as mis-adjustment can cause overshoot.

The EXPERT – BASS TOOLS TUNING controls fine tune the frequency and texture of the bass enhancement. Again, careful tuning is paramount for the best results.

## Multiband Drive (Basic Control)

Drives the level into the 10-band limiter. It's important to remember that the X1 needs only a few dB of limiting to be loud yet amazingly clean! The drive control is calibrated in percentage (%) where higher values equate more processing for the 10-band section.



This drive control has a secondary function in that it also can offset the level coming back to the processor from the insert loop. Most presets will have the multiband drive between 90-100%. If you find that you have to set this control outside this range, or that 100% does not yield enough drive, evaluate the gain structure of the equipment in the insert point.

To change the multiband drive, left click on the knob and use the mouse wheel to increase or decrease the value. You can also double click on the value and type in a new value between the number 80 and 100.



### Clipper Drive (Basic Control)

This control is basically the output of the 10 band limiter to the main clipper. Higher numbers yield louder audio with the traditional “loudness vs clean” tradeoff. The drive should be between +1dB and +3dB for competitive formats and -1dB to +1dB for classical or fine arts/talk programming.

To change the clipper drive, left click on the knob and use the mouse wheel to increase or decrease the value. You can also double click on the value and type in a new value between -6dB and +6dB.



### Bass Enhance (Basic Control)

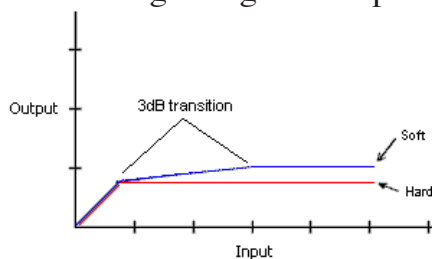
Governs the amount of enhancement driven into the separate specialized bass processor/limiter. The control range is -6dB to +12dB in 0.5dB steps with most presets settling on a value between -3dB to +3dB.

To change the bass enhance drive, left click on the knob and use the mouse wheel to increase or decrease the value. You can also double click on the value and type in a new value. If you make an incorrect entry, the control will round to the next available value.



### Multiband Knee (Basic Control)

Changes the knee of the limiter from soft to hard. When in HARD mode, the limiter thresholds set in the controls in the lower region of the limiter page are absolute... limiting will begin at the control settings. When in SOFT, gain reduction beginning to take place 3dB below a band's actual threshold setting with limiting gradually getting harder as it reaches threshold. In the soft mode, limiting is more obvious but the audio can also be smoother, especially on voice. In the hard mode, limiting is less noticeable but some peaky material may induce clipping faster.



Comparison of Hard and Soft Limiter Modes

To change Multiband Knee, left click on the knob and use the mouse wheel to switch between soft and hard. Left click again to confirm the change.



## Pre-Emphasis (Basic Control)

Sets the pre-emphasis curve to match what is required for your region and transmission system. The options are FLAT, 50uS and 75uS. To change the pre-emphasis curve, left click on the knob and use the mouse wheel to switch between FLAT, 50uS or 75uS. Left click again to confirm the change.

It should be noted that all the output DE-EMPHASIS options on the X1 will follow the PRE-EMPHASIZED curve selected here. If FLAT is chosen, there will be no de-emphasis applied at the selected outputs even if de-emphasis is engaged.



## Where Should Pre-Emphasis Go?

Pre-emphasis should always be applied by the audio processing and never by the exciter IF POSSIBLE. Modern FM audio processors are equipped with highly refined and very sophisticated technology to manage the myriad challenges posed by FM pre-emphasis. They can provide very tight modulation control with very low perceived distortion.

If you are in a situation where pre-emphasis must be applied at the exciter, the appropriate pre-emphasis should be selected in the X1 and the output feeding the exciter should be de-emphasized. This way, audio is peak controlled based on the pre-emphasis curve and overshoot in the exciter from its pre-emphasis will be minimized.

## Bass Style (Basic Control)

Similar to the 10-band limiter, the bass processor is also a limiter. It can be a hard knee limiter (for tighter bass control) or a soft knee limiter (which allows the bass a fuller and warmer sound). A third option is to turn off the bass processor, which is one of the three options available (OFF, Soft, Hard).

To change the bass style, or switch the bass enhancer off, left click on the knob and use the mouse wheel to choose. Left click again to confirm your choice.



## Master Attack (Expert Control)

The master attack control sets the attack speed for all 10 limiter bands. The normal setting for this control in all presets is AUTO, which means the attack speed is program dependent. Offset from auto are four settings that tell the attack speed to work faster or slower while still factoring in program content.

To change the master attack speed, left click on the knob and use the mouse wheel to increase or decrease the value.



## Master Limiter Release (Expert Control)

The master release control sets the release speed for all 10 limiter bands. The normal setting for this control in all presets is AUTO, which means the release speed is program dependent and based on how quickly the master attack speed control is set. Offset from auto are four settings that tell the release speed to work faster or slower while still factoring in program content.

To change the master release speed, left click on the knob and use the mouse wheel to increase or decrease the value.



## Master Threshold (Expert Control)

The Master Threshold sets the limit threshold for the 10-band limiter. Normally this control should be set to AUTO which allows the threshold to operate in a program dependent way. Changing the threshold will offset the control and make it a static threshold.

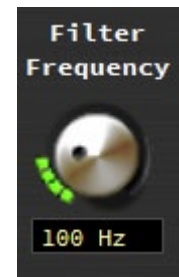
To change the threshold, left click on the knob and use the mouse wheel to increase or decrease the value.



## Filter Frequency (Expert Control)

The filter frequency sets the LPF (Low Pass Filter) for the Bass Enhance Limiter. It is not the “center” frequency for the bass processing, rather all audio below the frequency shown is processed by the Bass Enhancer and Limiter.

Normally there should be no need to change this control. However, if you feel you need to increase or decrease the filter frequency for a different type of bass response, left click on the knob and use the mouse wheel to increase or decrease the value. You can also double click on the shown value and type in a new setting. The range is 40Hz to 400Hz in 10Hz increments.



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## Bass Clip Threshold (Expert Control)

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To change the control, left click on the knob and use the mouse wheel to increase or decrease the value. You can also double click on the shown value and type in a new setting. The range is 0dB to -12dB. Suggested settings are usually -3dB to 0dB.



### Bass Texture (Expert Control)

Determines the level of upper harmonics of the bass enhancing limiter. Higher percentage (%) values create more harmonics. Lower numbers produce a smoother sounding bass. This control should be set by ear.

To change the control, left click on the knob and use the mouse wheel to increase or decrease the value. You can also double click on the shown value and type in a new setting. The range is 0dB to -12dB.



## MPX & OUTPUT

The MPX and OUTPUTS menu contains the controls required to configure the X1's internal DSP-based precision stereo generator and set the various output levels. Along with the output controls are adjustments for the HD time alignment and test tones to calibrate output levels.

There are three menus associated with this page: PROCESS & SCA, OUTPUTS and DELAY & TEST.

### Process & SCA Tab



#### Encoder Mode (Processing)

The stereo encoder may be operated in Stereo or Mono simply by selecting the desired operating mode with this switch. In the Mono mode the stereo pilot is turned off.

To change this setting, left click on the knob and use the mouse wheel to choose. Left click again to confirm your choice.



#### MPX Processing (Processing)

The MPX Processing control sets the input drive level to either the composite clipper or lookahead limiter as selected. We recommend that the Composite Clipper Drive be kept under +2.0dB for best MPX spectrum cleanliness (all composite clippers create MPX spectrum “trash”, even those that claim to create a tremendous amount of L/R demodulation).

To change the control, left click on the knob and use the mouse wheel to increase or decrease the value. You can also double click on the shown value and type in a new setting. The range is -6dB to +6dB. Suggested settings are usually between 0dB and +2dB.





## MPX Process Style (Processing)

The X1 is equipped with two methods of processing the composite stereo waveform to increase loudness:

- **Composite Clipper** – A high ratio distortion managed hard clipper precisely controls peaks without generating high order distortion. This clipper has a brighter sound than the Lookahead Limiter because it creates a higher level of harmonic artifacts.
- **Lookahead Limiter** – This option processes the composite stereo waveform with a highly-oversampled lookahead limiter with fully automatic attack and release times. Its 0.5 millisecond lookahead and extremely high sample rate precisely controls composite waveform peaks on a cycle by cycle basis.



If the MPX Process Style is set to Lookahead Limiter and the MPX Process Drive is set to 0dB or lower all composite processing is defeated.

To change this setting, left click on the knob and use the mouse wheel to choose Clipper or Limiter. Left click again to confirm your choice.

## Multipath Limiter (Processing)

The X1's Stereo Encoder (Stereo Generator) is equipped with a special algorithm called the Automatic Multipath Limiter. This algorithm operates in the sum/difference domain and automatically controls the amount of difference channel (L-R) amplitude as a function of the instantaneous sum channel (L+R) amplitude.

The purpose of this algorithm becomes clear when one realizes that a large amount of L-R is known to exacerbate FM multipath. We said "exacerbate" because a high level of L-R does not cause multipath. Rather, the behavior of a stereo receiver as it blends between stereo and mono becomes much more obvious when there is a large amount of L-R in the decoded audio because of the acoustical summation of L+R and L-R energy in the listening environment. When the L-R signal "goes away" during a receiver blend the perceived audio level drops and the larger the amount of L-R present when such a blend occurs, the more noticeable the multipath "event" will be to the listener.



When stereo enhancement is utilized in the program chain (or in the audio processor) L-R energy is naturally increased – this is the purpose of stereo enhancement – to exaggerate the stereo image width by increasing the level of L-R. When taken to the extreme however the extra L-R energy can cause either unnatural sounding audio on some program material (the hole in the middle effect), odd behavior in many stereo receivers, or both.

Another issue is that in FM broadcasting, each dB of L-R increase results in a like decrease in level in the L+R sum channel and it is this signal that is responsible for mono loudness. This is important to keep in mind when a fair percentage of the listening audience might be listening on mono receivers!

In order to make stereo enhancement "play nice" with the majority of stereo receivers in real-world listening environments with all types of program material, it is preferable to have some sort of controlling mechanism in place to "manage" the amount of L-R energy present in the transmitted signal as a function of program material. This is precisely what the Automatic Multipath Limiter does.

There are eleven operational choices available under the Multipath Limiter control settings: 10% to 100% in 5% percent steps, and an Off setting. Off is the setting immediately above the setting for 100%.

When in the Off position the Stereo Width Limiter is completely out of the circuit. When the control is at the 100% position the L-R may be permitted to achieve 100% of the instantaneous L+R audio level. At lesser percentages the amount of L-R energy is constrained to that percentage of the L+R level at that instant.

A recommended setting for this control is 70%, which constrains the L-R to about 3dB below the L+R. With the Multipath Limiter at this setting it will almost never touch “normal” stereo program material – even when stereo enhancement is used. This is in contrast to a control in another audio processor that has static settings for Stereo, -3dB, -6dB, and Mono. In that product, setting the control to -3dB causes a static 3dB reduction in stereo separation on all program material regardless of whether it needs it or not. We believe our approach is better because it is intelligent. It does not decrease stereo separation on program material that does not need it.

To change the control, left click on the knob and use the mouse wheel to increase or decrease the value.

### Pilot Level (Processing)

The Pilot Level control adjusts the level of the 19kHz stereo pilot and is adjustable from 0 (Off) to 20% of the main composite signal amplitude in 0.1% steps.

To change the control, left click on the knob and use the mouse wheel to increase or decrease the value.



### BS.412 Processing (BS.412 MPX Power Control)

Turns on or off the BS.412 Power Controller. The BS.412 MPX Power Controller's sole purpose is to reduce loudness and program density as required in certain European countries. **If you are not required to use the BS.412 Controller do not turn this feature on.** Enabling the BS.412 MPX Power Limiter can result in up to a 5dB loss in loudness.

To change this setting, left click on the knob and use the mouse wheel to turn on or off. Left click again to confirm your choice.



The X1's BS.412 MPX Power Controller can be set in 0.5dB increments from +6dB to -3dB. When the control is in any position it is measuring the current MPX Power level on the BS.412 meter on the front panel and GUI. Calculated corrections to the MPX power are only made when the BS.412 control is switched on, and when it is, modifications to the MPX power will take place immediately!

### IMPORTANT!

*The BS.412 MPX Power Controller's sole purpose is to reduce loudness and program density as required in certain European countries. **If you are not required to use the BS.412 Controller do not turn on the BS.412 option.** Turning on the BS.412 MPX Power Limiter can result in up to a 5dB loss in loudness.*

When the BS.412 is switched on, the MPX Power controller is then engaged and the algorithm immediately applies the measured MPX Power correction to the processed output. **After the MPX Power Controller is first engaged allow a full minute for the MPX Power to settle to its final value.**

As the controller measures and integrates the MPX energy over time the drive to the processing will be modified until the measured MPX power satisfies the reference level as set by the Stereo Encoder menu's BS.412 control. The control's "0dB" setting conforms to the current ITU-R BS.412-7 Multiplex Power standard.

There are some options in fine tuning the MPX power controller that are available in X1. These additional controls will never allow MPX power to EXCEED the legal maximum, instead they allow a tradeoff between subjectively "clean" or "aggressive" setting of the MPX power controller.

### BS.412 Peak Style (BS.412 MPX Power Control)

The BS.412 peak style control allows two options. Normal allows for very strict control of modulation of the signal in conjunction with MPX power management. In the "Peaky" mode, short duration peaks which do not affect total MPX power are allowed which can create a less restrictive sound. This is done dynamically by modifying the threshold of the distortion controller in the X1's left/right clipper and filter stage.

To change this setting, left click on the knob and use the mouse wheel to switch between NORMAL and PEAKY. Left click again to confirm your choice.



### BS.412 Speed (BS.412 MPX Power Control)

Adjusts the secondary recovery speed of the MPX Power control loop while the primary recovery speed remains program controlled. Higher settings result in faster release times of the secondary release.

To change the control, left click on the knob and use the mouse wheel to increase or decrease the value.



### BS.412 MPX Power (BS.412 MPX Power Control)

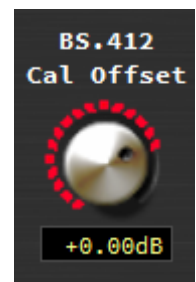
This control sets the maximum permitted multiplex power according to the BS.412 measurement algorithm. The control has a range from -3dB to +6dB in 0.1dB steps. This resolution allows small adjustments in actual MPX Power to allow for the maximum amount of power allowed by the local regulatory body.

To change the control, left click on the knob and use the mouse wheel to increase or decrease the value.



### BS.412 Calibration Offset (BS.412 MPX Power Control)

Adjusts the threshold of the power detector to modify its calibration. The control range is -18dB to 0dB. To change the control, left click on the knob and use the mouse wheel to increase or decrease the value.





### BS.412 Action (BS.412 MPX Power Control)

Adjusts the ratio between the primary and secondary timing loops. Higher settings result in more contribution from the slower, secondary timing loop, which also slows down increases in MPX power when it power tries to fall due to program variations.

The control range is -6dB to 0dB. To change the control, left click on the knob and use the mouse wheel to increase or decrease the value.



### SCA 1 and SCA 2 Inputs (Subcarrier Inputs)

Inputs applied to the SCA connectors are high pass filtered to reduce hum and then low-pass filtered at 94kHz before being digitized at 192kHz. The SCA 1 Gain and SCA 2 Gain controls set the SCA injection levels and can be set to OFF, or to levels between -80.0dB and +10.0dB and in 0.05dB steps.

To change the control, left click on the knob and use the mouse wheel to increase or decrease the value.



## Outputs Tab



The outputs tab allows X1 users to set the analog composite output drive, the analog FM output levels and the AES FM and HD output levels as well as the options for FM de-emphasis and associated options with the various outputs.

The controls in the Analog L/R section govern the two analog L/R XLR jacks on the rear of the X1. You can route the L/R HD processing to this output or the L/R FM processing to this output. The FM processing may be de-emphasized for feeding talent headphones or a local monitor, and it can also be selected before the X1's diversity delay for talent headphone processing when off air monitoring is not possible because of HD delay.

The FM DIGITAL output controls govern the type of output appearing on the AES XLR connector on the rear of the X1 as well as the level.

The HD DIGITAL output controls govern the output level and options that appear on the corresponding AES XLR jack on the rear of the X1.

### TX1 and TX2 Outputs (Analog MPX)

Sets the peak output amplitude of the MPX 1 and MPX 2 output on the rear panel BNC connectors respectively. These controls are used to set total modulation in the presence of audio and can be set to levels between -24dB and +6.00dB and in 0.05dB steps. Below -24dB, the control is set to OFF and no signal will appear on the output. The control setting for approximately 3.5V P-P (1V RMS) into a 10Kohm load is +2.40dB.

Before setting this control, you should have a calibrated modulation monitor with ample signal so that the correct (and legal) output level is set. To change the control, left click on the knob and use the mouse wheel to increase or decrease the value.

### TX2 Mode (Analog MPX)

TX 2 mode changes the TX2 composite output on the rear of the X1 to a 19kHz sine wave synchronizing source for RDS generators. To change this setting, left click on the knob and use the mouse wheel to switch between MPX OUT and PILOT. Left click again to confirm your choice.



## Source (Analog L/R)

Allows the user to route either the FM processing or the HD processing to the analog L/R XLR jacks on the rear of the processor. To change this setting, left click on the knob and use the mouse wheel to switch between MPX OUT and PILOT. Left click again to confirm your choice.



## Level (Analog L/R)

Adjusts the output gain of the selected source. The range of the control is -79.95dB to 0dB in 0.05dB steps. OFF is also available if the control is turned fully counter clockwise. To change the control, left click on the knob and use the mouse wheel to increase or decrease the value.



## Pre-Delay (Analog L/R)

If FM is selected as the analog L/R output, this option allows you to pick off the FM processed signal before the diversity delay that's used to match the analog FM signal to the latency of the HD signal. This pre-delay signal is helpful if you are monitoring the output of the X1 for processing adjustments, or if you need to feed your talent headphones with a low latency output.

To change this setting, left click on the knob and use the mouse wheel to switch between YES (audio is pulled before the delay) and NO (audio will have diversity delay IF diversity delay is applied). Left click again to confirm your choice.



## De-Emphasis (Analog L/R)

If FM is selected as the analog L/R output, this option de-emphasizes the analog L/R output jacks. The de-emphasis applied here follows the curve of the pre-emphasis selected on the FM Limiter page (see page 3-32).

To change this setting, left click on the knob and use the mouse wheel to switch between YES (audio is de-emphasized based on the selected pre-emphasis curve) and NO (audio will be pre-emphasized). Left click again to confirm your choice.



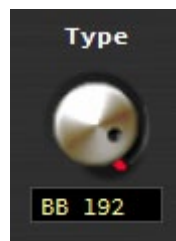
## Level (FM Digital)

Adjusts the output gain of the FM digital signal. The range of the control is -79.95dB to 0dB in 0.05dB steps. OFF is also available if the control is turned fully counter clockwise. To change the control, left click on the knob and use the mouse wheel to increase or decrease the value.



## Type (FM Digital)

The digital output has two options for the type of output. The first is baseband192 (BB192). When set to BB192, the digital output XLR jack on the rear of the X1 carries the Wheatstone baseband192 AES over MPX digital composite signal that is compatible with excitors designed to ingest a digital composite signal.



**WARNING:**

***You should check with your exciter manufacturer to make sure your equipment has the capability to ingest the baseband192 AES over MPX standard before selecting this option. Interfacing the X1 to an non-compatible exciter with this option selected can cause unwanted results.***

Unlike traditional AES, Baseband192 completes the final phase of the all-digital airchain that is clean AND highly competitive. Baseband192 allows you to use the composite clipper and stereo generator in the audio processor while the signal remains in the digital domain. Previously, if you wanted to interface your audio processor with your exciter using an AES signal, you needed to use the stereo generator in the exciter to complete the airchain. With the bandwidth available in Wheatstone audio processors and on the input of compatible exciters, it is now possible to interface a COMPOSITE signal over AES between the processor and the exciter. The stereo generator is now BACK in the audio processor, where it can work intelligently with the rest of the audio processing.

The X1 also supports the traditional AES interface between processor and exciter. By selecting AES3, the output appearing on the rear XLR connector is a standard AES L/R signal that can interface with STL systems or exciters that have AES input options.

**WARNING:**

***If your exciter has the option for AES over MPX (Baseband192) but you require or desire to use a traditional AES interface, make sure your exciter and the X1 have compatible settings. Mismatching the settings between the exciter and the processor can cause unwanted results. You should consult the user manual for your exciter or contact the manufacturer for information on configuring your exciter for the correct mode.***

To change this setting, left click on the knob and use the mouse wheel to switch between AES3 (traditional AES L/R output) and Baseband192 (composite digital output). Left click again to confirm your choice.

**Pre-Delay (FM Digital)**

This option allows you to pick off the FM processed signal before the diversity delay that's used to match the analog FM signal to the latency of the HD signal. This pre-delay signal is helpful if you are monitoring the output of the X1 for processing adjustments, or if you need to feed your talent headphones with a low latency output.

To change this setting, left click on the knob and use the mouse wheel to switch between YES (audio is pulled before the delay) and NO (audio will have diversity delay IF diversity delay is applied). Left click again to confirm your choice.

**De-Emphasis (FM Digital)**

This option de-emphasizes the analog L/R output jacks. The de-emphasis applied here follows the curve of the pre-emphasis selected on the FM Limiter page (see page 3-32).

To change this setting, left click on the knob and use the mouse wheel to switch between YES (audio is de-emphasized based on the selected pre-emphasis curve) and NO (audio will be pre-emphasized). Left click again to confirm your choice.



### Level (HD Digital)

Adjusts the output gain of the FM digital signal. The range of the control is -79.95dB to 0dB in 0.05dB steps. OFF is also available if the control is turned fully counter clockwise.

To change the control, left click on the knob and use the mouse wheel to increase or decrease the value.



### Phase Reverse (HD Digital)

When the Phase Reverse option is set to YES, the polarity of the HD Radio AES/EBU output signal is reversed by 180 degrees to compensate for phase errors between the FM and HD audio paths.

To change this setting, left click on the knob and use the mouse wheel to switch between YES (phase is reversed 180 degrees) and NO (phase is normal).



### L/R Swap (HD Digital)

The L/R Reverse checkbox allows the left and right channels of the HD output to be swapped (left becomes right, right becomes left) which can be useful during troubleshooting equipment that is external to the X1.

To change this setting, left click on the knob and use the mouse wheel to switch between YES (L/R is swapped) and NO (L/R is normal).





## Delay & Test Tab



The controls on this tab set the analog diversity delay as well as the settings for the test oscillator.

### Diversity Delay

The X1 is equipped with an internal digital delay to compensate for the corresponding delay in the analog signal path of stations using the HD Radio codec. The delay section is capable of providing up to ten seconds of compensating delay (480,000 samples) in steps of 20.83 microseconds.

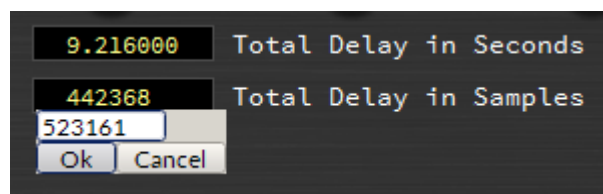
When the FM Diversity Delay controls are both set to their minimum settings, OFF appears in the delay window to signal that there is no delay the FM signal path.

Even though you can adjust the three virtual knobs in this section, there is a quicker and easier method to setting the delay, especially if you are using a third party device to calculate the time difference between your analog and digital audio signal.

MOST third party devices will tell you in samples how far off the two signals are. When your third party device has calculated the difference, you can double click in the numerical field and type in the new value.

If your third party device shows the difference in seconds rather than samples, the same can be accomplished by clicking on the Total Delay In Seconds number area and entering the new value in seconds.

**Third Party Auto Correction** – The X1 has the ability to interface with third party HD monitoring equipment to prevent time drift of the FM analog audio signal. All the



current products on the market from known manufacturers have been verified to work with Wheatstone processors, including the X1.

For more information on this technology, please feel free to reach out to Wheatstone at 252-638-7000 Monday thru Friday 8:30AM to 5:30PM ET (excluding major holidays).

## Test Oscillator

The built in Test Oscillator is capable of generating sine wave test signals from 50Hz to 80kHz at modulation levels from 0 to 100%. The oscillator is On when the Tone Enable option is set to YES.

To switch the Test Oscillator on, left click on the Tone Enable knob and use the mouse wheel to switch between YES (Test Oscillator is On) and NO (Test Oscillator is Off).

To change the Test Oscillator level, left click on the Tone Level knob and use the mouse wheel to increase or decrease the value. The range is 0-100% (modulation).

To change the Test Oscillator frequency, left click on the Tone Frequency knob and use the mouse wheel to increase or decrease the value. The range is 50Hz to 20kHz, Bessel, 40kHz, 60kHz and 80kHz.



## SYSTEM

The System screen allows system-wide settings to be configured, such as how the GUI communicates with the hardware and security settings within the processor.



### Front Panel Security Settings

Setting and clearing passwords for the front panel touchscreen is possible from the security section of the GUI. You can choose to separately lock the Processing-related functions and those associated with Output functions such as stereo generator settings and output levels. Each may have its own passcode or they may share the same passcode at the user's discretion. The reason for this functionality is to give stations who must cooperate with a designated transmission authority the ability to adjust their processing in any way they choose without giving the transmission authority access to the processing functions. Likewise it allows the transmission authority to manage "regulatory" things such as modulation levels, and without giving the transmission authority access to the processing controls.



There is no default password in the X1. If you are setting this up for the first time, you can skip the PASSWORD part and just enter the NEW PASSWORD and VERIFY the new password.

Other options in this menu include the ability for the PC to remember the password for the processor, to hide the password as you type it (with dots instead of the actual characters) and the ability to gang the password for both the processing and output options. When access is restricted, only the SYSTEM tab will be allowed to be viewed. If you try and access another tab, you will be prompted to input the proper password that covers the functions in that tab.

Pressing PROCESSING PASSWORD will bring up the screen to enter the new processing control password.

If the X1 does not have a password, you will be prompted to enter the password you would like to use to lock down the processing functions. Once entered, you will be asked to verify the password selection. If the passwords match, the X1 will indicate the password has changed.



Repeat these steps to set or change the password for the output settings of the X1.

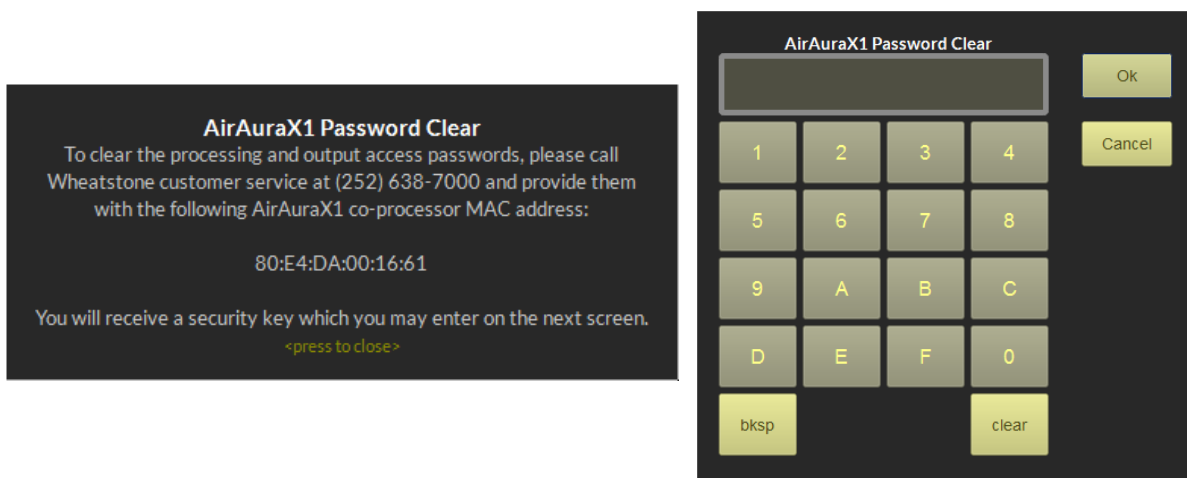


## Lost Passwords

If you have lost or forgotten your password, or encounter a locked X1 from a previous engineer/owner, resetting the password is possible. Tapping on the CLEAR PASSWORDS button brings up a screen that displays the MAC address of the processor. You will need to submit this MAC address to Wheatstone to get an unlock key which will clear the passwords on the processor.

Once you receive the passcode, you will be able to tap the CLEAR PASSWORDS option, close the MAC address message and move on to the keypad screen to enter the code provided by Wheatstone.

Once the passwords are cleared, you can choose to leave the box open or set new passwords.



**The password clear message and the password clear keypad**

Setting up the passwords via the GUI now allows you to lock the front screen to prevent access. You can either have someone manually lock the screen, or do nothing and within 15 minutes the front panel will automatically lock.

For more information on setting the password from the front screen and options to lock the front screen, see page 2-26.

## Time Features /Dayparting

The Time Features options allow you to change presets in the X1 at specific times as well as set the internal clock.

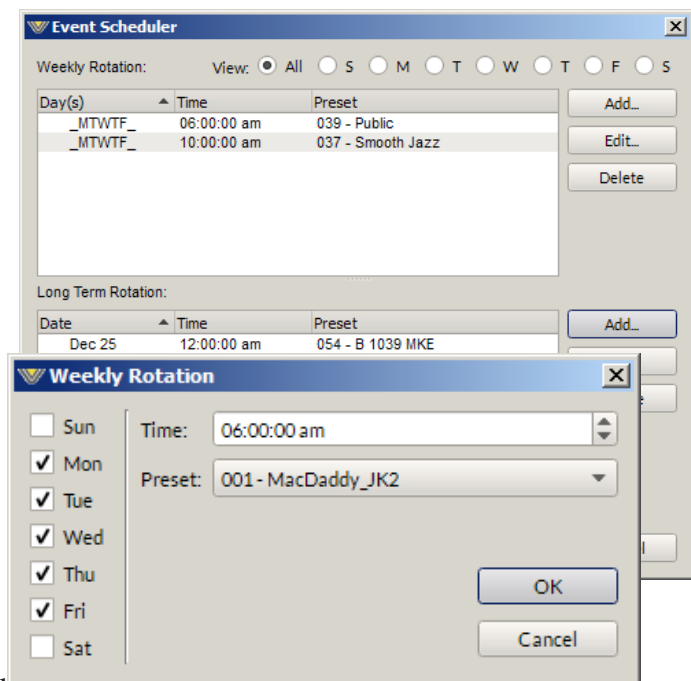
### Preset Scheduler

The X1 contains a comprehensive scheduling utility that allows presets to be automatically changed on desired dates and times. There is also a Long Term Rotation scheduler that allows preset changes to be scheduled for any time in the future.



### Creating a Weekly Rotation Schedule

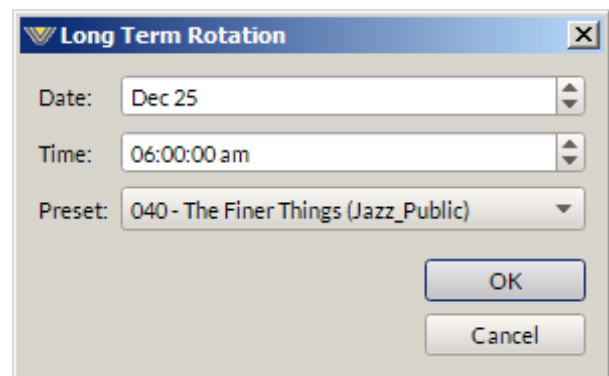
1. Click the Add button in the Weekly Rotation area of the Event Scheduler window. The Weekly Rotation window will open as shown on the right.
2. On the left column, select the days that the preset should be selected. Any combination of checkboxes can be chosen.
3. Next select the time that the preset should change.
4. Next choose the preset that should be selected at the time programmed in the previous step. When finished editing, click OK.



As many preset change events as desired may be programmed in the Event Scheduler. If a one-time preset change is required, such as might be necessary for a special holiday or other event, the Long Term Rotation routine should be used.

### Creating a Long Term Rotation Schedule

Adding a new event in the Long Term Rotation scheduler works in a similar manner. The Long Term Rotation entry window is shown here on the right.



### Deleting a Scheduled Event

Deleting a scheduled preset change is as simple as adding one. Open the Event Scheduler window by navigating to the System menu and clicking Scheduler. The Event Scheduler window will open.

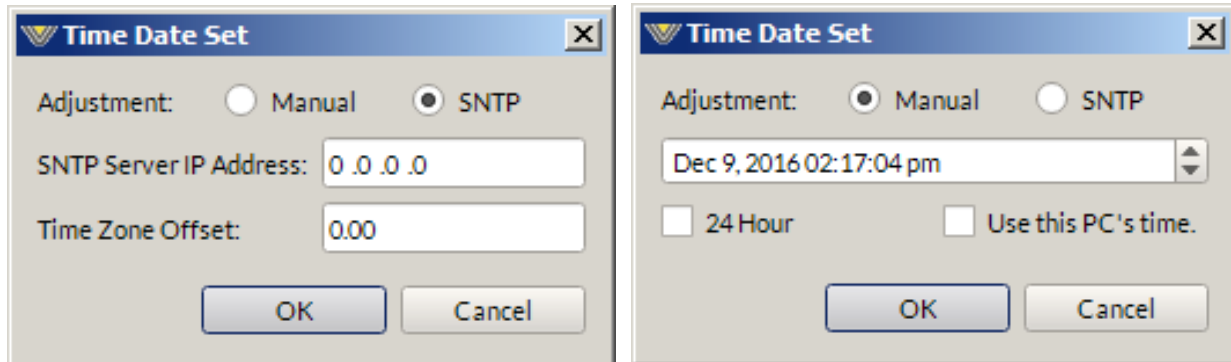
To delete an item in either the Weekly Rotation or Long Term Rotation schedules simply highlight it in the list and then click the appropriate Delete button on the right side of the Event Scheduler window. The highlighted scheduled event will be removed from the list.

**NOTE: There is no confirmation prior to the actual deletion of the event!**

## System Time

Allows the system time of the AirAura onboard computer to be set or changed. The AirAura system time can be synchronized to external time servers (SNTP, or Simple Network Time Protocol) if desired. Using a search engine, you can find the IP addresses of active SNTP servers in your region.

Alternately, time can be synchronized to the PC that is running the Windows-based GUI. In this instance time updates will occur upon successful connection to the GUI on the remote PC.



## Options and Status

The following status indicators are available on the X1. A red dot next to a status indicator means there is either no audio present, a feature is not active or a status has alarmed. A green indicator means that there is audio present, a feature is active or no alarm is pending.

**Analog** – The analog mode is active when the “(active)” text appears next to the mode. A red dot indicates no audio is present at the input of that mode. A green dot indicates audio is present.

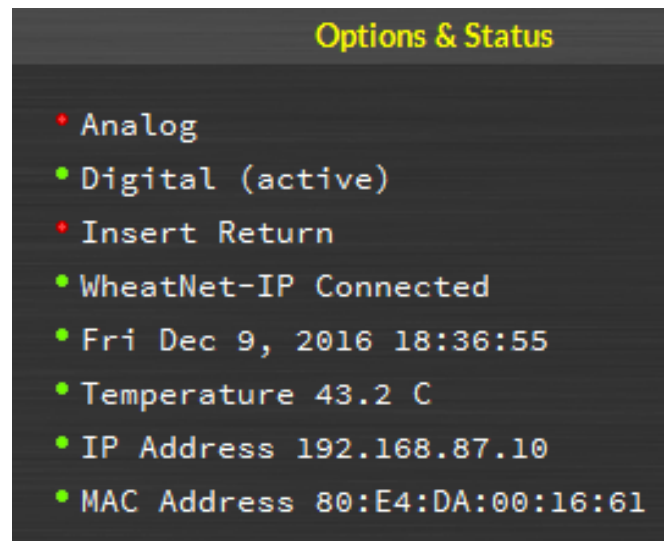
**Digital** – The digital mode is active when the “(active)” text appears next to the mode. A red dot indicates no audio is present at the input of that mode. A green dot indicates audio is present.

**Insert Return** – The insert return is active when the “(active)” text appears next to the mode. A red dot indicates no audio is present at the input of that mode. A green dot indicates audio is present.

**WheatNet-IP Connected** – A green indicator means the X1 is connected to the WheatNet-IP AoIP network.

**Date and Time** – Processor current date and time (24 hour clock) described above.

**Temperature** – Internal operating temperature of the X1 in Celsius. The green indicator will turn red and a closure on the GPIO panel will occur for notification purposes if the internal temperature is measured above 50 C.



**IP Address** – Shows the hardware IP address.

**MAC Address** – Shows the unique MAC (Media Access Control) address of the X1.

## Miscellaneous

By default, the X1 utilizes the UDP protocol for transmitting metering data to the remote GUI (The TCP protocol with its guaranteed accurate packet delivery is always used for control).



In some situations UDP packets are blocked by Internet Service Providers, and when this occurs the GUI can successfully connect to X1 but there is no visible metering data. Unless one can successfully convince the ISP to allow unsolicited UDP traffic on their network (doubtful) the solution is to utilize VPN Tunneling (which encapsulates all packets into TCP) or simply switch the X1 and its GUI to use the TCP protocol for metering. When the Metering Data over TCP/IP box is checked the TCP protocol is used for transmitting metering data instead of UDP.

## Headphone Source

The headphone source option allows you to listen to your audio at various stages of processing. Not only is it helpful in that it allows you to hear the audio as it goes through the various processes, it also allows you to troubleshoot potential issues (like comparing the source audio to the output product or comparing the FM output with the HD output)

There are seven monitor points –

**Input** (monitors the active input)

**Analog Input** (allows you to monitor the analog source if it isn't active)

**Digital Input** (allows you to monitor the digital source if it isn't active)

**HD Output** (allows you to monitor the HD output after all processing)

**FM Pre-Delay** (allows you to monitor the FM output before diversity delay)

**FM Output** (allows you to monitor the FM output after all processing and delay)

**Insert Return** (allows you to monitor the insert point return).



To switch the headphone source, left click on the knob and use the mouse wheel to switch between one of the seven options. Left click the mouse to select the new source. The source WILL NOT CHANGE until you confirm it by left clicking the mouse.

## Control Timeout

When any of the “and/or” options are changed in X1, the user needs to press the left mouse key or the jog wheel on the front panel in order to confirm the change. The control timeout sets the amount of time allowed to confirm the change before the control being adjusted times out and returns to the previous setting.

The control range is OFF (never goes back to the previous state) to 60 seconds. The shortest time allowed is 1 second. The default is 10 seconds.

To change this control, left click to lengthen or shorten the time of the timeout.



## Presets

On the right of the System screen is a vertically disposed row of buttons to manage, save, backup, and copy presets. The three buttons are:

**List** – Shows the list of presets on the hardware. Double clicking on any of the presets will take that preset.

**Save** – Opens the save dialogue box that allows you to name a new preset and pick the preset slot you would like to save the preset to.

**Library** – Opens the preset manager. The preset manager allows you to copy presets from the hardware to the local PC and vice versa. It also allows you to extract presets from the hardware to share with other users.



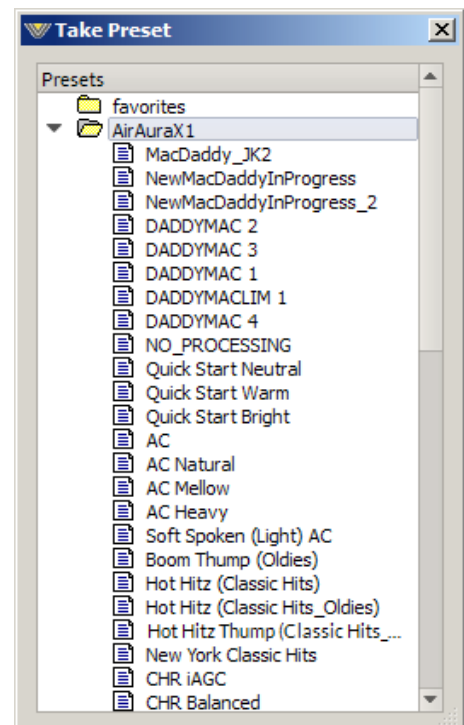
## List

When left-clicked, a dialog box appears, showing all presets stored on the hardware.

Double-clicking on a preset brings it immediately into use in the X1. The box stays open until deliberately closed, allowing differing presets to be double-clicked upon readily in succession. This is a very direct means of comparing presets.

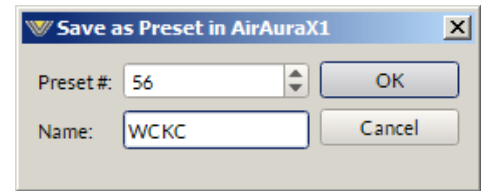
**NOTE: It is important to remember to save the current present settings before invoking others, or they will be lost.**

Note also that you can only Take presets that are already loaded in the X1 from this dialog box.



## Save

Clicking on the Save button opens a dialog, prompting the user for a name for the present being saved. The preset will be saved into the next available empty slot unless another empty slot is chosen by nudging the preset number (Preset #) up or down. Unless a preset is user protected or factory protected, it is possible to overwrite saved presets if one is not careful.

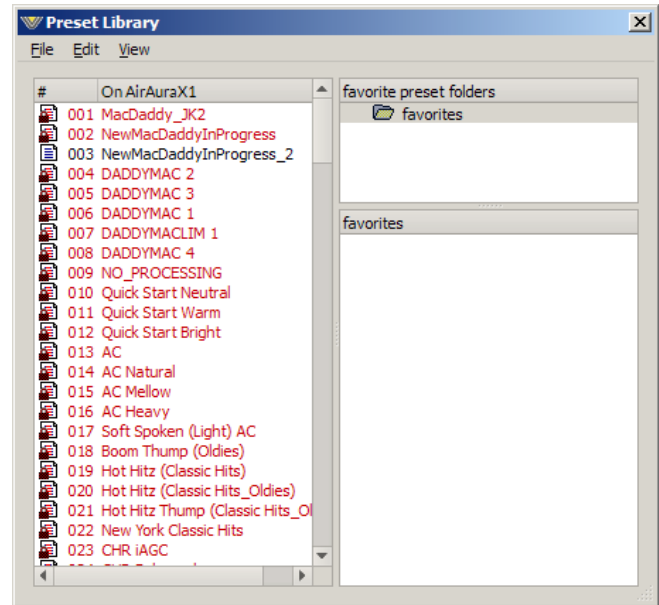


## Library

When the Library button is pressed, a window-style dialog box appears containing three panels. The large panel on the left shows the presets currently stored on the X1.

The upper right panel shows the preset folders on the PC hosting the GUI, and the lower panel shows the presets stored in the folder chosen in the above panel.

Normal Windows-style drag-and-drop functions allow presets to be freely moved between the hardware and the PC and vice versa. Note that when preset are moved this way, it is copies of presets, not the presets themselves which are moved. The original preset always stays where it was stored last.

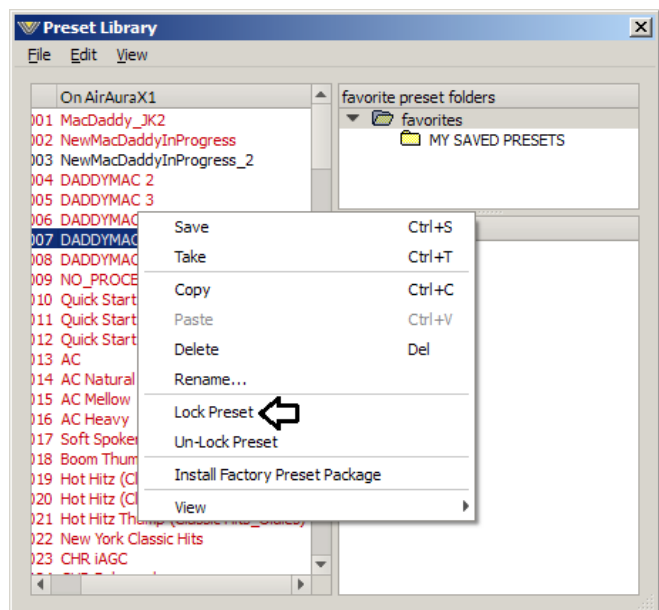


## Locking Presets

Presets stored on the X1 may be locked by the user to prevent inadvertent overwriting, renaming, or deletion. This is accomplished by opening the Preset Library by clicking on the Library button in the GUI. Once the list of presets is open, the ones actually stored within the processor hardware itself will be visible in the left pane.

There are two ways to manage the lock status of user presets:

- The first method is by highlighting a preset (single left click) and then right clicking it to open a dialog box. Among the options are Lock Preset and Unlock Preset.
- The second method is by highlighting a preset as above, and then clicking the Edit option at the top of the Preset Library dialog box to reveal the Lock Preset/Unlock Preset options. Note that these are user-level lock options and therefore cannot be used to override the lock status of a factory-locked preset!





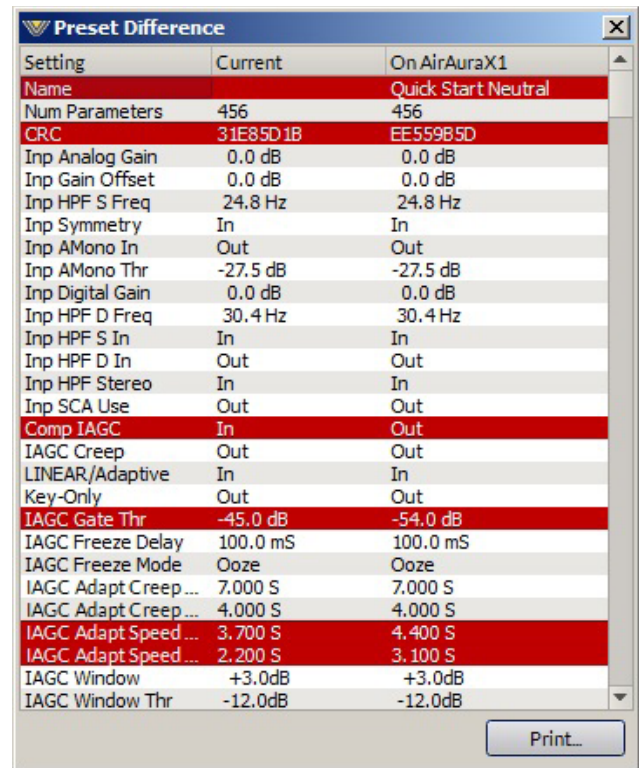
## How To View Preset Differences

It is possible to visually compare two different presets on the X1 with a special feature called Preset Difference.

When the Library tab is opened, highlight any preset you'd like to compare with the current preset running on the processor. Then right click and select View and Diff from the sub-menu. You will see the window on the right open.

Any parameter that is listed with a RED background indicates a parameter that is difference between the preset running on the hardware and the preset that has been highlighted. Parameters which are not highlighted have the same values between the two presets.

Preset settings can also be printed or exported as a .csv file which can be read by Excel programs.



Setting	Current	On AirAuraX1
Name		Quick Start Neutral
Num Parameters	456	456
CRC	31E85D1B	EE559B5D
Inp Analog Gain	0.0 dB	0.0 dB
Inp Gain Offset	0.0 dB	0.0 dB
Inp HPF S Freq	24.8 Hz	24.8 Hz
Inp Symmetry	In	In
Inp AMono In	Out	Out
Inp AMono Thr	-27.5 dB	-27.5 dB
Inp Digital Gain	0.0 dB	0.0 dB
Inp HPF D Freq	30.4 Hz	30.4 Hz
Inp HPF S In	In	In
Inp HPF D In	Out	Out
Inp HPF Stereo	In	In
Inp SCA Use	Out	Out
Comp IAGC	In	Out
IAGC Creep	Out	Out
LINEAR/Adaptive	In	In
Key-Only	Out	Out
IAGC Gate Thr	-45.0 dB	-54.0 dB
IAGC Freeze Delay	100.0 mS	100.0 mS
IAGC Freeze Mode	Ooze	Ooze
IAGC Adapt Creep ...	7.000 S	7.000 S
IAGC Adapt Creep ...	4.000 S	4.000 S
IAGC Adapt Speed ...	3.700 S	4.400 S
IAGC Adapt Speed ...	2.200 S	3.100 S
IAGC Window	+3.0dB	+3.0dB
IAGC Window Thr	-12.0dB	-12.0dB

## Hardware

The two hardware buttons are ANALYSIS and DEVICES.

**Analysis** – shows the extra metering options for monitoring the left/right domain signal as well as the MPX signal

**Devices** – lets you configure the GUI to talk to a specific X1 over your network or public internet.

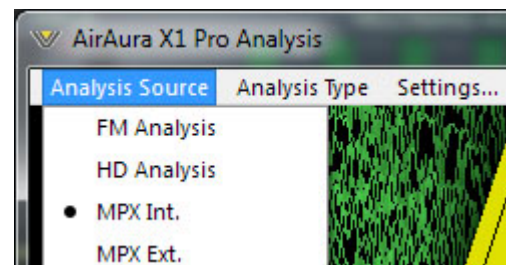


## Analysis

The X1 is equipped with extensive processing analysis features that give a more detailed look at what the processing is actually doing. There are eight analysis options. All eight are available in the FM or HD internal mode, while only four are available in the MPX mode.

### Analysis Source

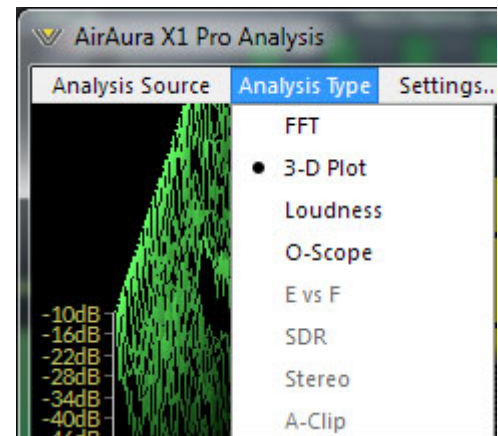
Analysis Source selects the mode of analysis. You can monitor the FM or HD processing, or choose the MPX Internal or External sources. The MPX internal mode allows you to use the tools available to monitor the baseband of the X1's signal, taken at the output of the stereo generator. The MPX external mode looks at the baseband injected into the SCA 1 input (if that input is selected for analysis)





## FM & HD Analysis Options Type

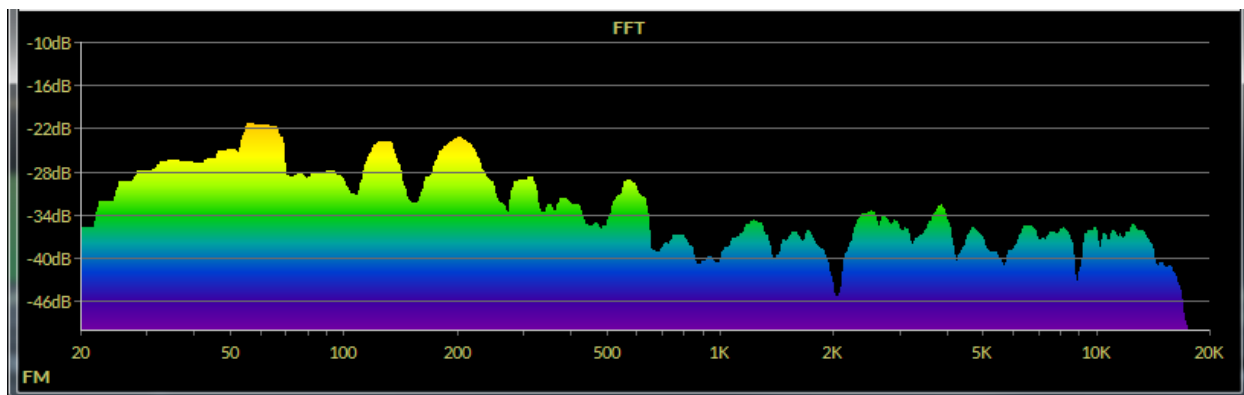
- **FFT** – Displays the spectral output of the FM or HD processing paths after processing. The FM display includes any pre-emphasis being used in order to present an accurate view of the signal actually being sent to the transmitter by the X1.
- **3-D Plot** – A three-dimensional plot of the audio, after processing, which displays signal amplitude versus frequency versus time. The display has a ten second historical window depth allowing some historical perspective about the FM or HD audio after processing.
- **Loudness** – Displays a meter which shows the long and short term loudness of the audio processing
- **O-Scope** – Displays the audio waveforms at the output of the FM or HD processing chains.
- **E vs F** – Displays the measured RMS Loudness.
- **SDR™** – Spectral Dynamic Range™. This display shows the amount of dynamic range within 31 ISO frequency bands after processing.
- **Stereo** – Displays a meter which shows the L+R/L-R ratio from 0 to 1.0.
- **A-Clip** – This is AirAura's final FM clipper activity display that shows where in the audio spectrum distortion products are being detected and reduced and by how much.



A detailed explanation of each analysis type follows.

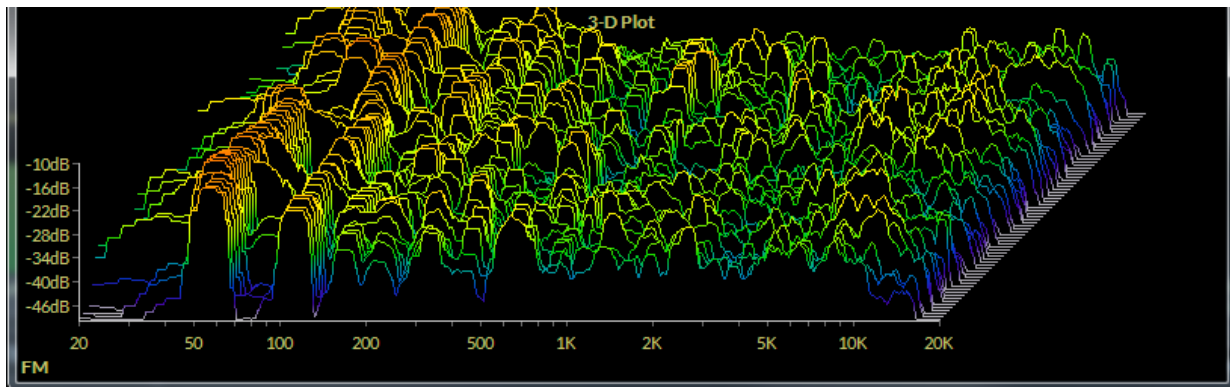
## FFT

This display shows a Fast Fourier Transform of the audio spectrum after processing has been applied by the final FM clipper or HD limiter. The FFT display's high resolution is useful for locating where in the audio spectrum certain signals are heard. With complex program material the FFT gives good visual feedback on the levels of audio residing in different parts of the audio spectrum.



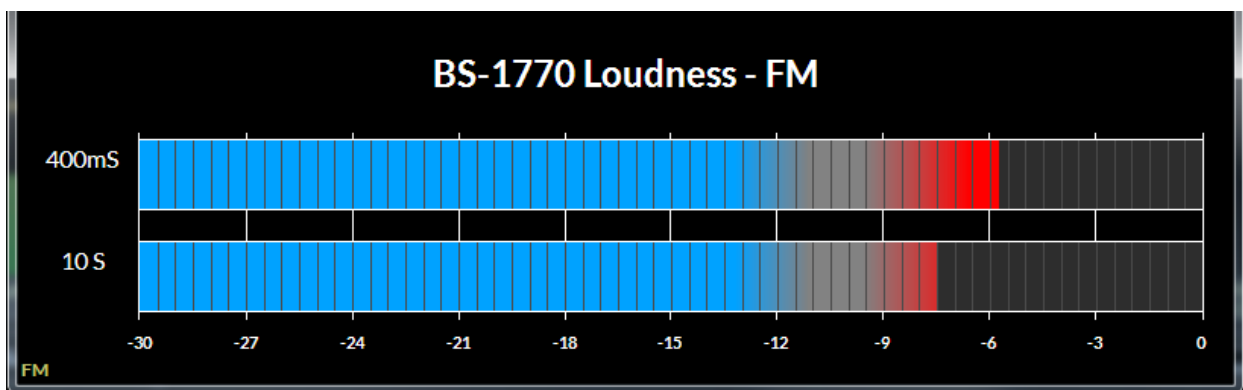
### 3-D Plot

A 3-D plot of the audio spectrum shows the last ten seconds of audio and what the audio levels were at each frequency during that time. The vertical scale represents level, the horizontal scale frequency, and the diagonal scale is time. The graph being drawn right up front represents “now.” As the display is updated, “now” becomes “then” and it moves towards the back of the display and is replaced by a new “now.” The 3-D plot very accurately displays how levels within different parts of the audio spectrum have been changing over time. The floor of the display is purple and represents levels below -48dBFS. As the levels become higher the colors morph towards red, the maximum displayed level. The higher the signal is in the plot, the larger the signal amplitude at the frequency of the peak.



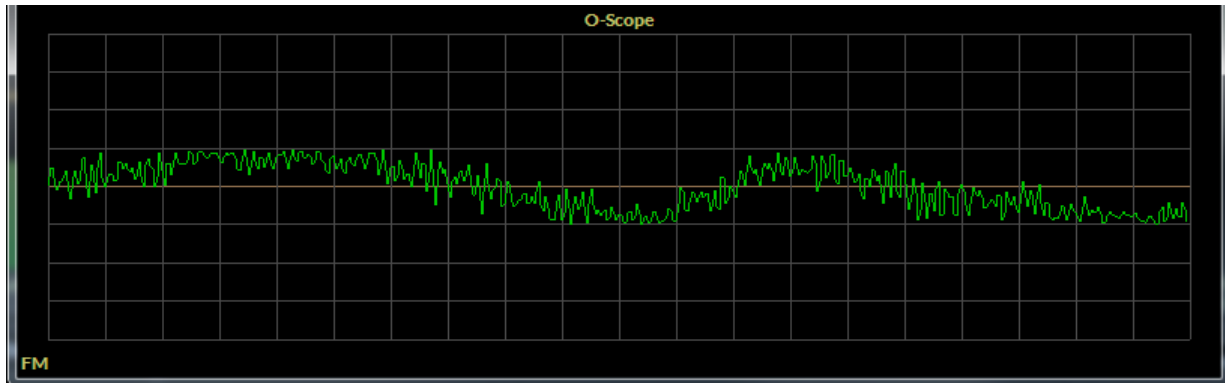
### BS-1770 Loudness

The loudness meter shows the short term and long term average loudness of the processed audio on the HD or FM side. The blue region is for more relaxed audio while the contrast between the grey and red areas indicate the level of loudness achieved with older digital or analog processors (grey) and the newer digital audio processors (red).



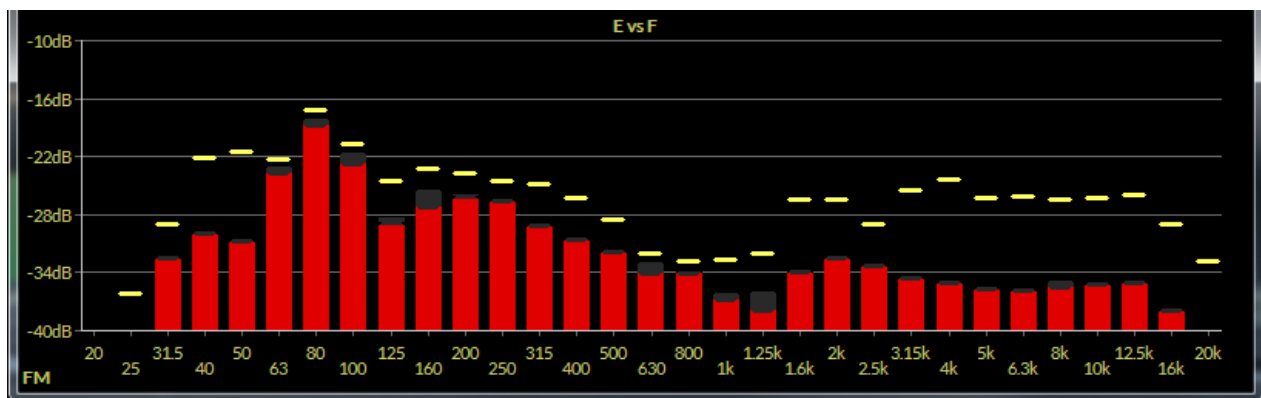
## O-Scope

This is a time-domain (oscilloscope-type) display of the audio at the output of the FM and HD final peak control sections. The FM O-Scope display shows what the pre-emphasized and clipped audio looks like. The HD path, with its more “gentle” lookahead limiter, will appear less “aggressive” than the FM audio.

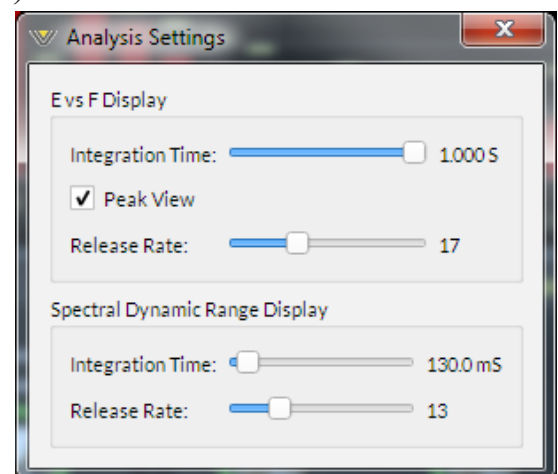


## E vs. F

This display represents the RMS Loudness generated in each band of the FM or HD Limiter as well as the peak value.



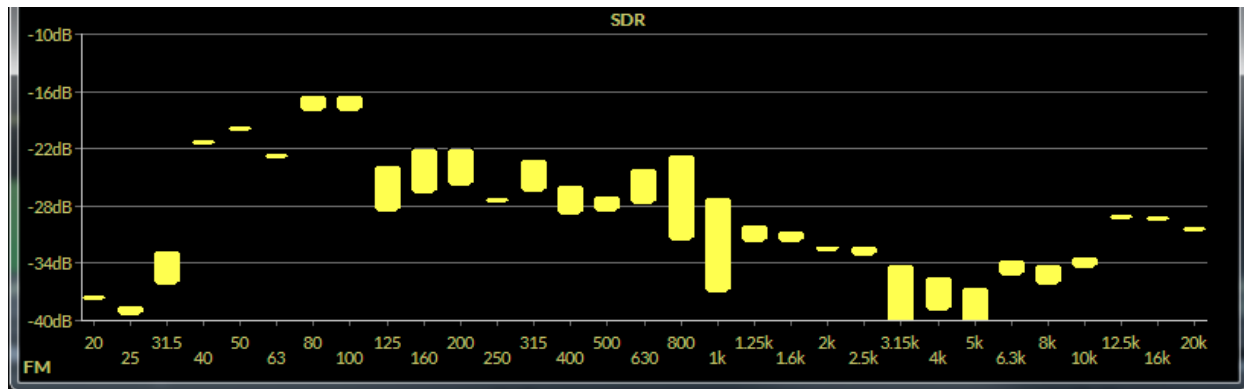
The E vs. F display has a submenu (shown right) that allows customization of the RMS Integration Time, the display Release Rate, and whether or not peak levels are to be displayed. The factory default Integration Time is 300mS – the same as a standard VU meter – which is also a fair estimator for how loud something seems to the human ear.



## SDR™

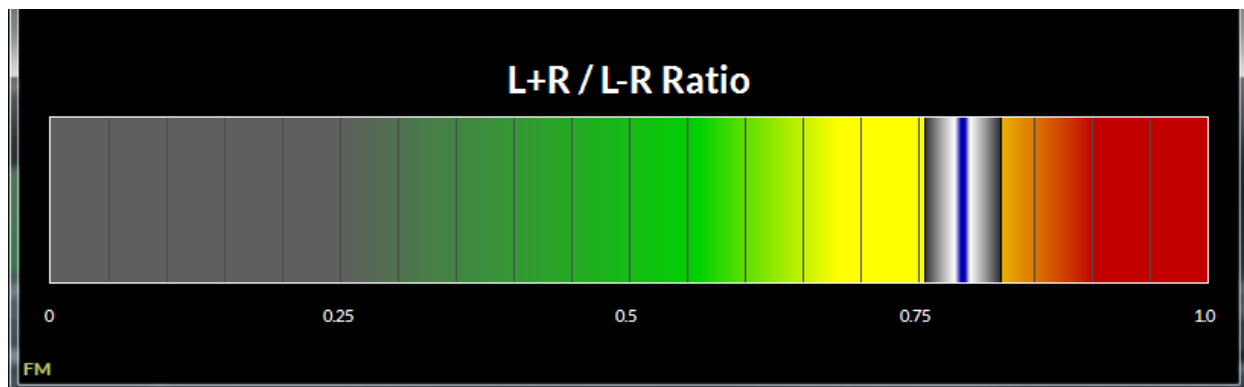
During development of the AirAura back in 2010, we invented a new and useful processing analysis tool, the Spectral Dynamic Range™ Meter. This special 31-Band analyzer was designed to measure dynamic range. The SDR meter is easy to interpret – the top of the bar represents the maximum peak level measured over the last sample period. The bottom of the bar is the average level over the last sample period.

Therefore the taller the blue bar is, the more dynamic range there is within an analysis band's frequency range. Conversely, the shorter the bar's length, the denser the audio is and the louder it is within that frequency range.



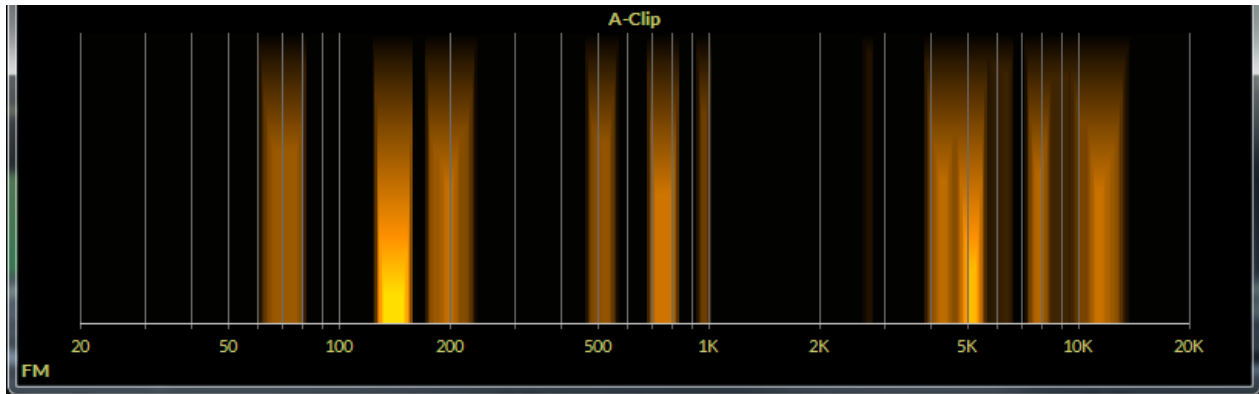
## Stereo

This meter shows the ratio of L+R/L-R (sum to difference). The further to the right, the more stereo separation is being transmitted (enhanced at the source of a combination of source separation and added enhancement in the X1). The further to the left, the less stereo separation being transmitted. For mono or voice, the meter should sit at 0.



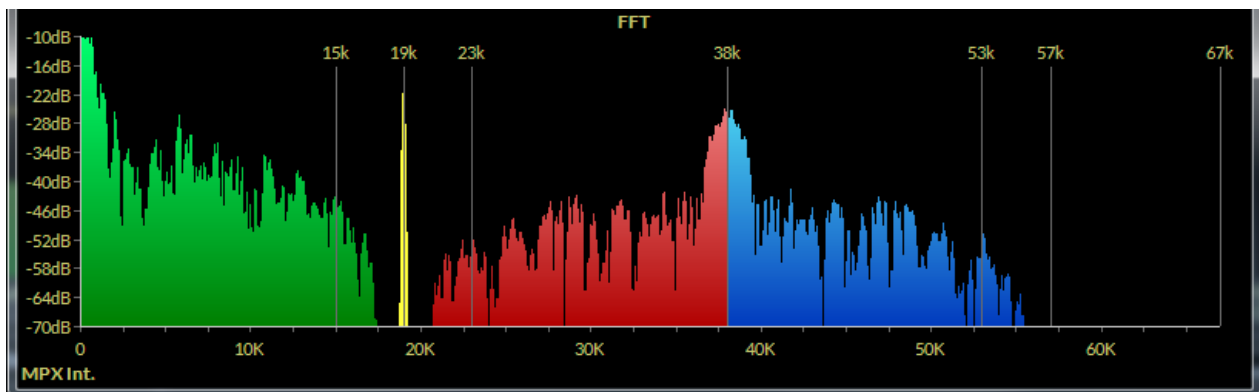
## A-Clip

During development of our the AirAura-FM clipper in 2010, we invented a way to watch it work. Known as the A-Clip display, it's now designed to work with the X1. The A-Clip shows in detail where in the audio spectrum the X1's distortion management is working. The “ghosts” that appear show where in the audio spectrum the distortion management is.

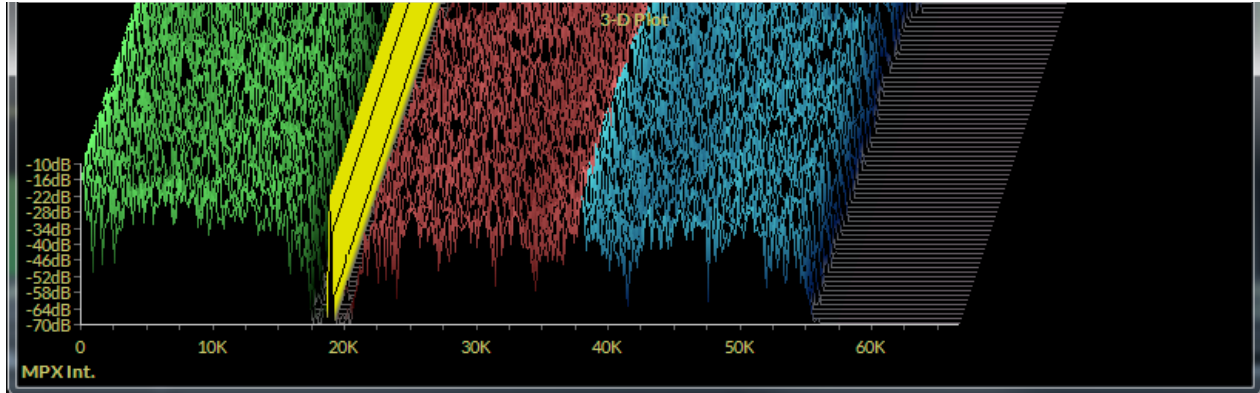


## MPX (internal & external) Analysis Options

- **FFT (MPX)** – Displays the spectral output of the baseband of the AirAura including L+R, Pilot, L-R Upper and Lower sidebands and subcarriers. This analysis shows the baseband spectrum of the AirAura audio processor's composite output. It is color coded as follows:
  - Green = L+R
  - Yellow = Pilot
  - Red = L-R Lower Sideband
  - Blue = L-R Upper Sideband
  - Purple = Subcarriers (RDS etc)



- **3-D Plot (MPX)** – A three-dimensional plot of the spectral audio. This analysis takes the baseband and shows it in our popular 3D (Time vs Frequency vs Amplitude) display. The color coding follows the same legend as our FFT analysis.



*The Loudness and O-Scope options are the same in the MPX analysis as they are in the FM analysis.*

## Devices

The DEVICES button opens up the application to configure a the GUI to talk to a specific X1 based on its hardware address. For information about this, please see page 3-6.



## X1 Hardware Update

The X1 is extremely easy to update when new firmware becomes available from Wheatstone. As mentioned above there are two different file types involved in updating the X1. They are:

**The VBN file** – VBN stands for “Virtual Binary Nugget.” This file contains main board application code, FPGA images, and DSP code.

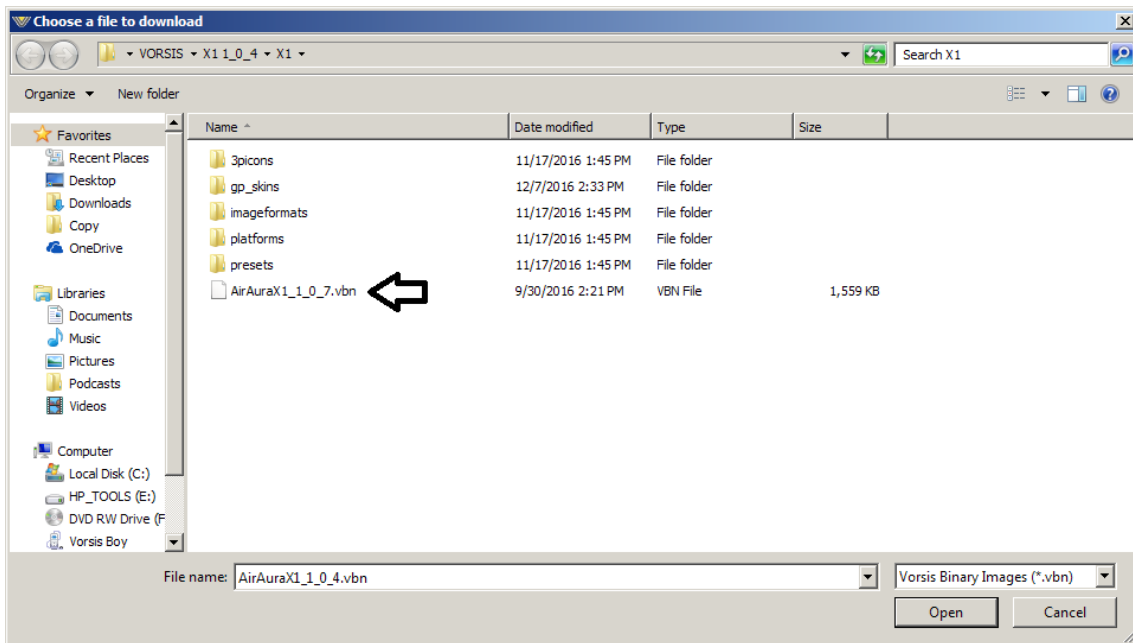
**The LBN file** – LBN stands for “Linux Binary Nugget.” This file contains the front panel GUI application code, Guru GUI operating parameters and skin files for the front panel graphics display.

VBN and LBN file updates may be performed separately, or they may be performed together if updates for both sections of the X1 have been made available.

Before performing updates to the X1 make sure an auxiliary processor is available for when X1 needs to be rebooted after the upgrade as there will be 15-20 seconds of silence until the X1 reboots. Alternately perform the upgrade at a time when you can temporarily remove audio from the air without it being an issue.

To update the X1, right click on any portion of the upper control area in any screen of the remote GUI to open pop up menu tree.

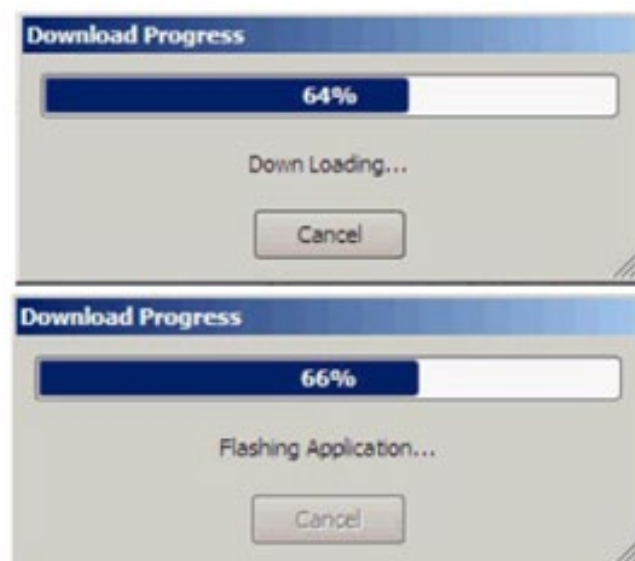
For **VBN** updates of the motherboard, select the **Update Co-Processor...** option listed above the **Update Front Panel...** The screen below will appear.



Navigate to the location of the VBN file that was downloaded from the Wheatstone Processing website or if a new GUI version was just installed, the VBN that was unpacked during its installation. Verify that the version number is correct in that it matches the version number listed in the documentation about the firmware upgrade. If so, select the **Open** option in the *Choose a file to download* window. The prompt shown at right will appear with wording of the text varying according to versions to be uploaded.

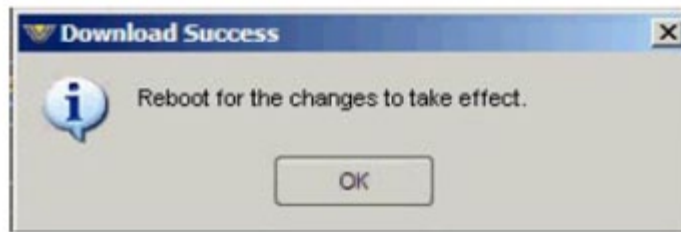
Recheck that the versions match the documentation! Then click **Yes**.

Progress boxes will pop up as the download progress occurs:



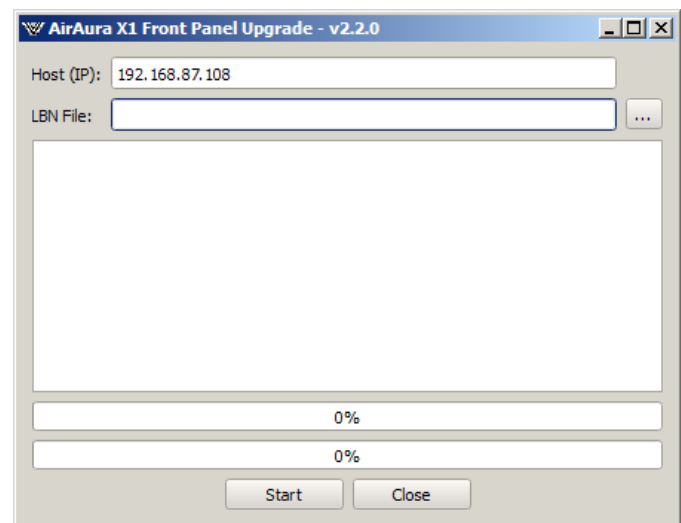


When the system has finished downloading, you will get a prompt to reboot the X1.



For **LBN** updates of the Front Panel, select **Update Front Panel...** and the screen on the right will appear.

Notice that the *IP Address* field is blank. This must be filled in to agree with the **Front Panel IP** address of your AirAura as it is currently configured. Next, click on the ... button at the right end of the LBN File: field to navigate to the location of the LBN file. Verify that the version number is correct and matches the version number listed in the documentation received about the firmware upgrade. The file has a *.lbn* extension. Please make certain this is the case before proceeding. When ready to begin the update process, click the *Start* button at the bottom.



**It is important to NOT interrupt the upgrade process by any means while it is in progress!**

**VBN/LBN UPDATING SEQUENCE** – The VBN and LBN updates, while separate update processes, may be completed in any order and/or at the same time before rebooting the X1 once to bring all the updates into effect. For instance, you could perform the VBN update and then, when prompted to reboot, not reboot, and perform the LBN update instead, and THEN reboot after that process has complete.

You may also delay rebooting the X1 until any convenient time. After the VBN and LBN uploads, the updates are securely stored in flash memory and won't take effect until the next reboot.

## X1 Sub Menus

Right clicking anywhere on the GUI brings up the X1 sub menu options. Many of these options are redundant and this menu serves as a shortcut to access them.

### View

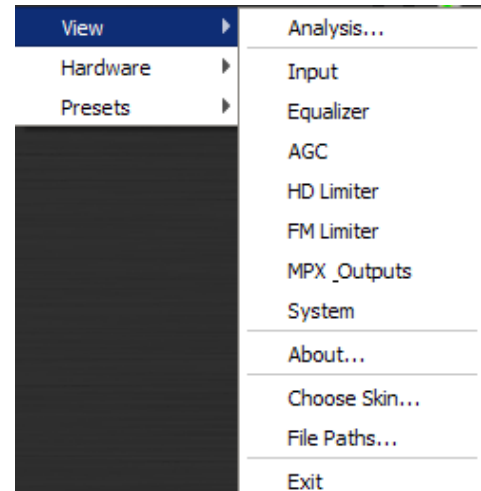
The **View** menu allows you to launch the **Analysis** screen as well as change the current processing page. The options for **Input**, **Equalizer**, **AGC**, **HD Limiter**, **FM Limiter**, **MPX\_Outputs** and **System** duplicate the menu option buttons between the metering and controls.

**About...** – Displays the version of the GUI.

**Choose Skin...** – Allows you to change the skin color/texture. As the X1 product evolves, Wheatstone may release different versions of skins. This is where a user would go to select a new skin. As of the printing of this manual (December 2016), there is only one default skin available.

**File Paths...** – Display the location of various files that the X1 GUI needs to reference for correct operation.

**Exit** – Closes the GUI.



### Hardware

The **Hardware** menu duplicates some of the options found elsewhere in the X1 and are shown here as a shortcut.

**Devices...** – Opens up the application to configure a the GUI to talk to a specific X1 based on its hardware address. For information about this, please see page 3-6.

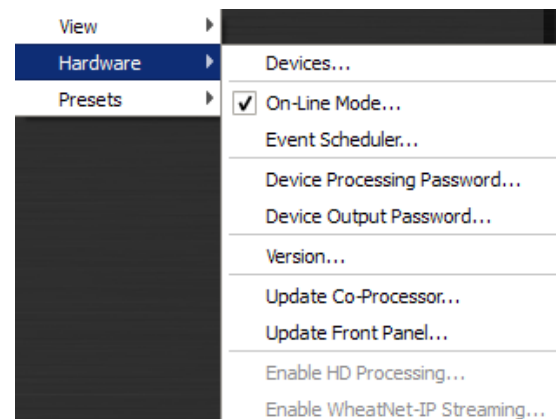
If configured properly, clicking ON-LINE MODE will allow the GUI to connect to the target hardware.

**Event Scheduler...** duplicates the Preset Scheduler button in the SYSTEM menu, while the **Device Processing Password...** and **Device Output Password ...** duplicate the corresponding buttons in the SECURITY section of the SYSTEM menu.

**Version...** – Queries the hardware and displays the current version running on the hardware.

**Update Co-Processor...** and **Update Front Panel...** allows you to update the audio processor when updates are made available from Wheatstone (see page 3-59).

**Enable HD Processing...** and **Enable WheatNet-IP Streaming...** should normally be greyed out if all options on the X1 have been purchased. If you do not have operational HD processing or WheatNet-IP audio capabilities, please contact Wheatstone at techsupport@wheatstone.com to inquire about adding these features.



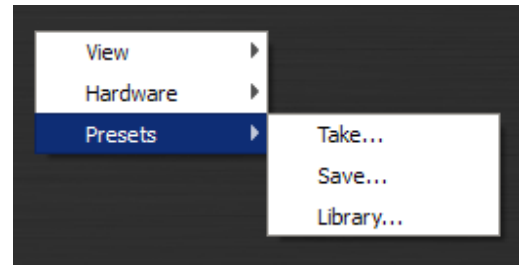
## Presets

The **Presets** menu duplicates the corresponding buttons in the Preset section of the SYSTEM menu.

**Take...** – Brings up the **Take Preset** dialog box.

**Save...** – Brings up the **Save as Presets** dialog box.

**Library...** – Brings up the **Preset Library** dialog box.



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# Appendix A

## Contents

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## General Specifications

### Analog Line Input

Type:	Electronic Differential
Input Impedance:	> 10Kohm (bridging)
Optimum Source Impedance:	< 1Kohm
A/D Converter:	AKM5394, 192kHz, 24-bits

### Digital Line Input

Data Standard:	AES3 (AES/EBU)
Data Amplitude:	Per AES3-2003 assuming minimum allowable output signal amplitude of 2V and minimum allowable input signal amplitude of 200mV
AES Receiver:	CS8416, 192kHz, 24-bits
Type:	WheatNet-IP
Data Type:	Gigabit Ethernet Network per IEEE 802.3-2008

### Input Gain Adjustment Ranges

Gain Range:	+/-24dB
Gain Adjustment Resolution:	0.5dB
Gain Calibration:	A gain control setting of 0.0 aligns an external 0dBFS signal with the X1's 0dBFS internal reference

### Input Failsafe

Type:	Automatic
Analog Fail Cause:	Audio level below -24dBU (fixed)
Response Time:	30 seconds (fixed)
Digital Fail Cause 1:	Audio level below -48dBFS (fixed)
Response Time:	30 seconds (fixed)
Digital Fail Cause 2:	Corrupted or invalid AES data
Response Time:	Immediate (fixed)
Failsafe Direction:	Digital to Analog/Analog to Digital (no Digital to Digital)

### Audio Level Balance

Type:	Common to both Analog and Digital Inputs
Analog/Digital L/R Balance Range:	+/-12dB
Control Resolution:	0.5dB

## Voice Symmetry via Phase Rotator

Operating Modes:	In/Bypass
Filter Type:	4 <sup>th</sup> Order All-pass

## High-Pass Filter

HPF Filter Class:	24dB/octave Butterworth
Frequency range:	20Hz – 300Hz

## Parametric Equalizer

Sections:	Four
Frequency Range:	20Hz – 20kHz
Bandwidth Range:	0.2 – 3.0 octaves
Boost/Cut Range:	+/- 14.0dB
Options:	Bypass Bands 1&4 Shelving

## 5 Band AGC/Compressor with iAGC

Drive Gain Range:	-12.0dB to +6.0dB in 0.5dB steps
AGC/Compressor Thresholds:	- 50dBFS to -70dBFS
AGC Attack:	50mS to 3000ms
AGC Release:	1000mS to 7000mS
AGC Band Coupling:	0dB to -6.0dB per band to key band
Compressor Attack:	10mS to 1000mS
Compressor Release:	50mS to 3000mS (3Sec)
Compression Ratio:	2:1 to 20:1
L+R Mixer:	Boost/Cut +/-6dB
L-R Mixer:	Boost/Cut +/-6dB
Gating Mode:	Slow Release/Hold
Gate Threshold Range:	-20.0dB to -79.5dB, plus OFF
iAGC Attack/Release Speed:	2 to 7.0 seconds
iAGC Threshold:	-60.0dBFS to -40.0dBFS
iAGC Window Size:	0dB to 8.0dB
iAGC Operating Style:	Linear or Adaptive

## Crossover Frequencies

Band 1/2:	50.0Hz to 153Hz
Band 2/3:	223Hz to 687Hz
Band 3/4:	1.2kHz to 2.0kHz
Band 4/5:	4.45kHz to 7.55kHz



**10-Band HD Peak Limiter**

10-Band Limiter Drive:	80 – 100%
Final Limiter Drive:	-6.0dB to +6.0dB
Bass Enhance:	-6.0dB to +12.0dB
10-Band Limiter Knee Shape:	Soft/Hard
Final Limiter L/R Linking:	On/Off
Bass Enhance Style:	Off/Soft/Hard
10-Band Limiter Attack/Release Time:	Automatic
Final Limiter Delayed Release Time:	100ms to 1000ms
10-Band Limiter Master Threshold:	Automatic
Final Limiter Threshold:	-6.0dB to +6dB
Bass Tools Filter Frequency:	40Hz to 400Hz
Bass Tools Filter Style:	Broad/Medium/Tight
Bass Tools Threshold:	0dB to -12dB
Bass Tools Texture:	0% to 50%

**10-Band FM Peak Limiter**

10-Band Limiter Drive:	80 – 100%
Clipper Drive:	-6.0dB to +6.0dB
Bass Enhance:	-6.0dB to +12.0dB
10-Band Limiter Knee Shape:	Soft/ Hard
Final Limiter L/R Linking:	On/Off
Pre-Emphasis:	Flat/50 $\mu$ S/75 $\mu$ S
Bass Style:	Off/Soft/Hard
10-Band Limiter Attack/Release Time:	Automatic
10-Band Limiter Master Threshold:	Automatic
Bass Tools Filter Frequency:	40Hz to 400Hz
Bass Tools Filter Style:	Broad/Medium/Tight
Bass Clip Threshold:	0dB to -12dB
Bass Tools Texture:	0% to 50%

**Stereo Encoder**

Reference grade Stereo Encoder with embedded Composite Processing, Test Oscillator, SCA digitizer, and balanced and unbalanced composite outputs.

Composite Processor Modes:	Oversampled Clipper or Lookahead Limiter
MPX Processing Drive Range:	+/-6dB
Automatic Multipath Limiter:	10% to 100% in 5% steps, plus Off
19kHz Stereo Pilot Injection:	0-20%, 0.1% steps

19kHz Stereo Pilot Phase:	+/- 22.5 degrees referenced to 38kHz
SCA 1 & 2 Input:	Analog, 10Kohm input impedance
Maximum SCA Input Level:	+24dBu
Gain Control Range:	-79.95dB to +10dB; OFF
Stereo Encoder Operating Modes:	
Analog Left/Right Only:	Analog L/R or Analog L/R De-Emphasized
Stereo Multiplex:	Balanced and Unbalanced MPX Output
TX 1 & 2 Output Standard Level:	3.5V P-P (1V RMS)
TX 1 & 2 Maximum Output Level:	8V P-P into 1Kohm (BNC Outputs) 14V P-P into 1Kohm (Balanced Outputs)
BS.412 Power Control:	On/Off (Bypass)
BS.412 Power Control Options:	Level/Style/Speed/Calibration/Action

## System

Headroom Level:	>20dB
Nominal Operating Level:	-20dBFS digital
FM Path Processing Latency:	17mS maximum – all features engaged
HD Path Processing Latency:	13mS maximum – all features engaged
Total Harmonic Distortion:	<0.1%, 20Hz – 20kHz *
Intermodulation Distortion:	<0.1% SMTPE *
Signal to Noise Ratio:	>80dB *
Signal Chain Internal Dynamic Range:	>144dB
Stereo Separation:	>50dB (FM) 100dB (HD) *
Crosstalk:	>50dB, 20Hz – 15kHz (FM) 20Hz – 20kHz (HD) *
Power Requirements:	100-250 VAC (auto sensing) 50/60Hz, 100 VA Max.
Power Connector:	EMI suppressed IEC male
Shipping Weight:	37 pounds
Operating Temperature:	0 to 50 degrees C (32 to 122 degrees F) Overtemp alarm reporting via GUI

\* Bypass Preset, unity gains, and 75 $\mu$ S pre and de-emphasis

All specifications subject to improvement or change without notice.

## Preset Management

The X1's advanced preset management system allows the creation, storing, and recall of 80 presets within the X1 itself and an unlimited number may be stored on the Windows PC hosting the GUI. In addition to processing parameters, all System parameters are stored within presets allowing the preservation of the complete processing "environment." Optionally, Input and Output settings may be stored and recalled with presets.

## Preset Encryption

Presets are encrypted using the X1's internal 32 bit serial number as the encryption key.

## Onboard Real Time Clock

A highly accurate internal quartz-based timebase drives a real-time clock which allows the automatic recall of presets at predetermined times using the short or long-form scheduler features. This clock may be set to "free-run" or it can be synchronized to an external SNTP server.

## Day-Parting and Long-Term Scheduling

A "weekly," or short form preset scheduling utility establishes a weekly cycle of preset changes, allowing for automatic programmed day-by-day exclusions or additions. Additionally, a separate long-form scheduling table permits the establishment of "one-off" preset changes for any future time and date.

## General Purpose Input (GPI)

Eight optically isolated inputs are assigned to the first eight numbered preset slots. Presets may be recalled by providing the appropriate voltage to the associated GPI port to create a "Logic High" and may be either momentary or latching as desired. Voltages are applied between the appropriate GPI pin and Common and polarity does not matter. Please see the section on GPI use for the required external current limiting resistor for voltages above 3.3VDC.

## General Purpose Output (GPO)

Four optically isolated outputs bring system statuses and alarms to the outside world.

## Ethernet Interface

The X1 contains a five-port, auto-sensing 10/100BaseT Ethernet switch with port available on the outside rear panel for field use. Because the switch is Auto-MIDX it is not necessary to utilize a crossover cable when connecting a PC directly to the X1.

## Software Remote Control

The supplied Windows®-base X1 GUI software affords simplified access to X1's system and processing parameters via a 10/100BaseT Ethernet interface. A front-panel color LCD screen and touchpad allows complete local control of all system and processing parameters if desired.

# Appendix B

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## X1's GPIO Interface

The X1 is equipped with two GPIO (General Purpose Input/Output) control interfaces; an eight input optoisolated interface for selecting presets in the first eight memory slots and a four output optoisolated interface that can be used to remotely signal status about the processor.

### GPI – General Purpose Input

The GPI interface provides eight separate General Purpose Inputs which share a common DC return. The interface connector is a female DB-9 connector with the following pin assignments. Note that all pins are isolated from AirAura X1 internal circuitry!

GPI Input Number	DB-9 Connector Pin Number
One	Pin 6
Two	Pin 2
Three	Pin 7
Four	Pin 3
Five	Pin 8
Six	Pin 4
Seven	Pin 9
Eight	Pin 5
Common Return	Pin 1

The GPI input circuits are optoisolated by devices having exceptional input/output isolation. These inputs are designed to work very well in high RF environments and should not cause ground loop issues when connected to external equipment.

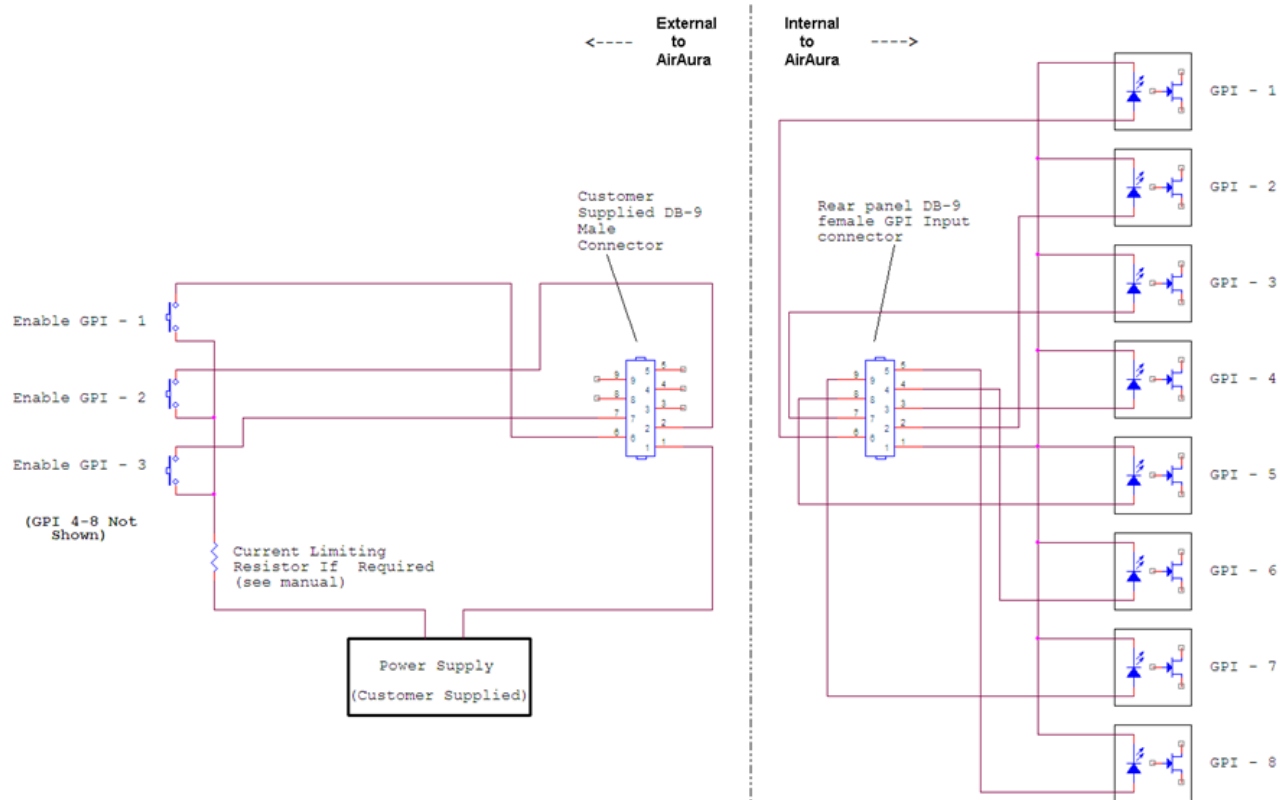
Each of the GPI inputs is current limited by a 475 ohm internal resistance. The LED device inside the optoisolators is rated at a maximum forward current of 50mA and a normal operating current of 5mA.

**Before connecting external circuitry to the GPI inputs,  
first ascertain whether it will be necessary to utilize  
additional outboard series resistance in your application!**

Use the following table to determine if an external resistance is required and if so, what value it should be. Note that a one-half watt resistor is sufficient. The resistance values in the table below have been calculated to result in approximately 5.0 milliamps forward current through the optoisolator's internal LED.

Applied Voltage to GPI	External Resistance
3.3 Volts	None
5.0 Volts	330 Ohms
6.0 Volts	470 Ohms
7.5 Volts	820 Ohms
10 Volts	1.3 kOhms
12 Volts	1.8 kOhms
15 Volts	2.4 kOhms
24 Volts	3.9 kOhms
30 Volts	5.1 kOhms
*48 Volts*	9.1 kOhms

\* For personnel safety reasons a voltage in excess of 48VDC is NOT recommended to be applied to the GPI interface.



**Example Schematic of General Purpose Inputs**

## GPO – General Purpose Output

AirAura X3 provides four General Purpose Output (GPO) circuits that may be used in a variety of ways. Like the GPI circuits, the GPO interface utilizes optoisolators in order to provide high electrical isolation from the outside world.

Unlike the GPI circuits each of the GPO's has completely isolated return circuits – that is, unlike the GPI port they do not share a common return. Because of this each GPO may be used as the end user chooses without regard to cross coupling between unrelated external circuits.

The optoisolators on the GPO interface protect internal AirAura X3 circuitry from voltage up to 350 Volts AC or DC. Because of the optoisolator's "AC" rating, the polarity of external circuit polarity is unimportant and therefore the GPO pins are simply defined as "Source" and "Return."

The MAXIMUM permissible load current is 100 milliamps (0.10Amperes).

The four GPO ports on AirAura X3 are hard coded in software to provide the following status outputs:

*GP Output #1* – Becomes enabled on an Analog Audio Failure.

*GP Output #1* – Becomes enabled on Digital Audio Failure (either AES3 or WheatNet-IP).

*GP Output #2* – Becomes enabled if the CPU Temperature reaches 50 deg. C (122 deg. F).

*GP Output #3* – Becomes enabled upon a General System Failure.

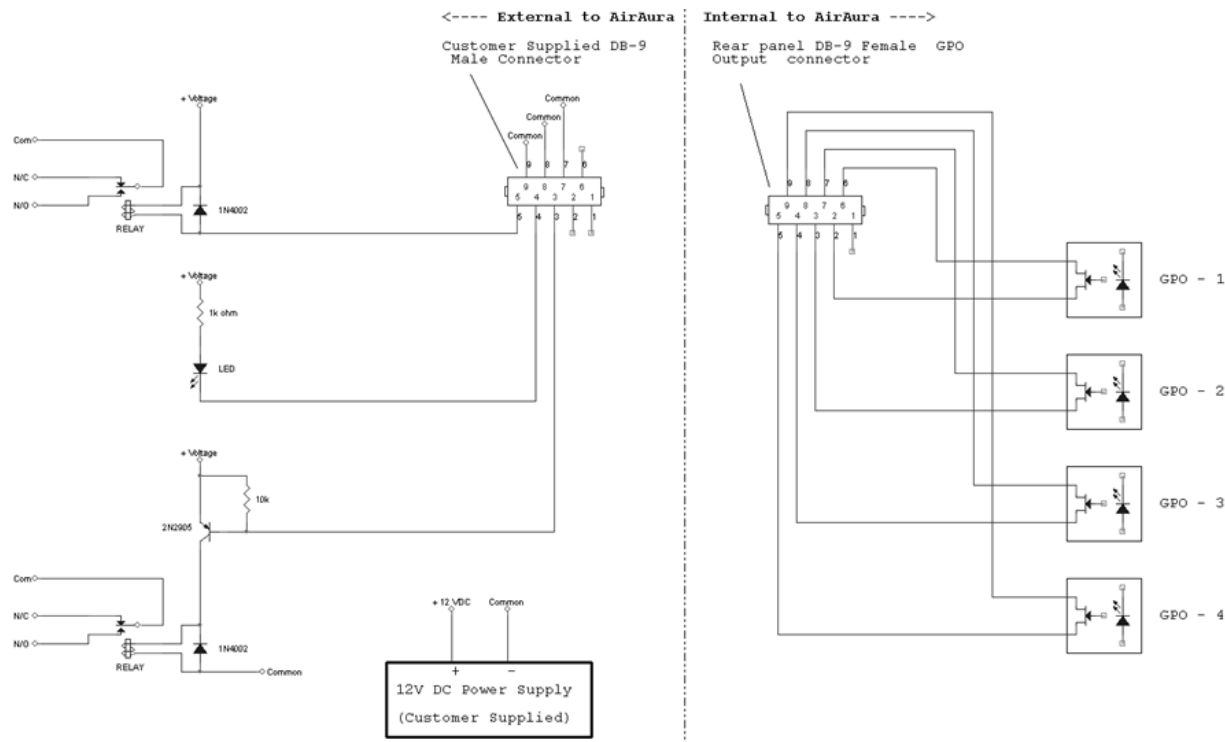
The pin assignments for the rear panel DB-9 GPO connector is as follows:

GPO Number	Usage	Source Pin	Return Pin
One	Analog Audio Fail	2	6
Two	Digital Audio Fail	3	7
Three	Overtemp (>50C)	4	8
Four	General Failure	5	9

Note that the GPO DB-9 connector pin 1 has no connection.

Please see the next page for a suggested way to interface the X1's GPO circuitry.





**Example Schematic of General Purpose Outputs**

In the above circuits the external devices are powered by a customer-supplied 12V DC power supply. When relays are interfaced it is recommended to use the reverse biased diode shown across each relay coil. The diode harmlessly dissipates the voltage “kickback” generated when the relay coil is de-energized. Unsuppressed, such kickback voltages can harm the X1, external circuitry, or both.

In the top circuit the relay is activated if GPO#4 goes true to warn of a General System Failure.

In the middle circuit the LED illuminates when GPO #3 goes true to warn of a System Temperature above 50C (122F).

In the bottom circuit the relay is activated by GPO #2 to warn of a Digital Audio failure.

No connections were made to the terminals for GPO #1, the Analog audio source failure warning.

# Appendix C

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## Interpreting Common Audio Processing Terms

### Background

When tasked with adjusting audio processing one is often faced with myriad meanings for the same sound descriptors. When the program director comments that it needs more “thump,” what does he really mean? If he says that the audio is too “crunchy,” what is he hearing? Or if he says that the competition is nice and “bright” and they have some nice “rumble,” what exactly is that? Or what if he wants more bass and you add some bottom end but now he says “we sound too muddy”? How do we untangle this and get to the bottom of what the PD really means?

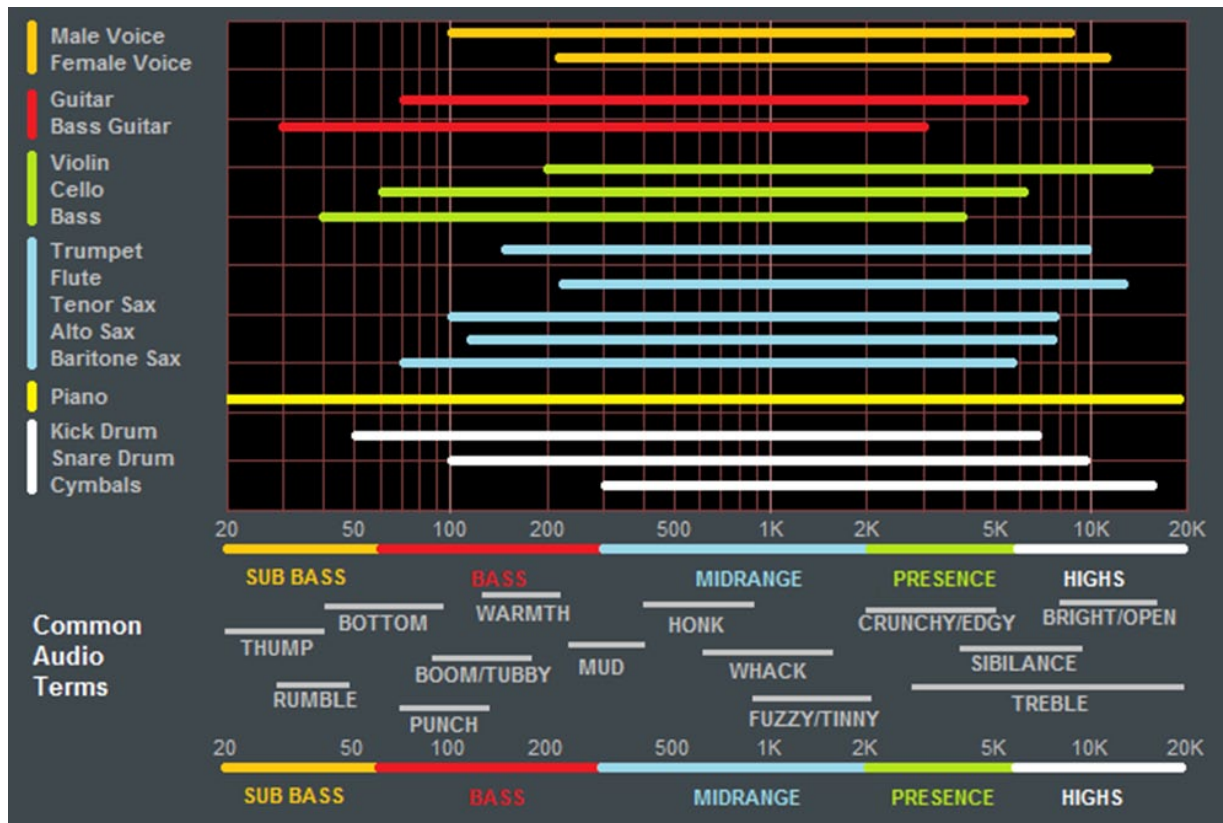
Lots of “audio” terms regarding audio processing have been invented and bandied about over the years. Ever since radio stations started employing circuits instead of humans to control the modulation of transmitters we’ve tried to come up with words to describe the effects of processing and what we hear. Many of these terms were born out of simply trying to put into words the positive and negative artifacts of early compressors and limiters (thump, muddy, honk, tubby). Still others were born later when EQ and different forms of “enhancers” were added to audio chains and audio processors to exaggerate certain parts of the frequency spectrum (rumble, punch for bass; bright, airy, open to describe presence and the high end).

The *language* of audio processing more often than not needs the benefit of an interpreter. Some oft-used terms can easily have multiple meanings and be confused with others. Some of the terms we use simply don’t describe very well what is actually happening to the audio. For instance, one listener might feel the audio is “muddy” while another one still might describe it as “warm.” Which one is right? For starters, “muddy” is not usually used in a positive sense, while warm, on the other hand, often means the sound has a pleasing low end. See the confusion here?

The terms themselves, the ones that we use to describe sound, can be confusing enough. The real question then becomes, what do these terms really mean, and which knob do we turn if we want to change the sound? In fact, here is a very common challenge: if the audio is too “muddy,” do you turn up the high end, or do you reduce the low end? Which control is the right one if you wanted to bring up some low end “rumble” or “sub bass”? And here’s another one; if the audio is “tinny,” is the problem that there’s too much energy up around 8,000Hz or is the problem actually further down in the spectrum?

We’ve created a graphic to try to bring some understanding to the more common terms we broadcasters use to describe sound and where these sounds actually reside in the audio frequency spectrum. While creating an exhaustive list isn’t impossible, it is well beyond the scope of what we’re trying to accomplish here. What we’ve done instead is compile the terms we most commonly hear customers use, and then lay them out in graphic form to help visualize what sounds live in what part of the audio spectrum we hear.

Included in the chart below are the frequency ranges of the most common instruments heard in compilations played on the radio. The idea here is to marry musical instruments and their sounds with the terms we all use. One good example might be the term “Honk,” which coincidentally happens to be right in the middle of the frequency range created by saxophones. The goal here is simple; offer a better understanding of how to adjust an audio processor so that the “thump” you want will be the “thump” you get and not “tubby.”



Common audio processing terms and the parts of the spectrum they govern are at the bottom of the chart. The top part of the chart shows the frequency range of common musical instruments heard on the radio (male and female voice included).

## Audio Terms and Wheatstone Processing

So how do you tweak a processor to change its sound “texture”? Let’s start with the low end...

### THUMP/RUMBLE/BOTTOM

These terms describe bass generally found at and below about 80Hz. Good stereo systems, closed or noise reduction headphones and car audio systems with subwoofers can usually reproduce the audio way down here. Many times such systems even have their own complex bass enhancement features. Wheatstone processors are equipped with a very specialized bass system called “Bass Tools™.” In general terms Bass Tools gets real bass on the air without it causing the problems commonly associated with “bass clippers.”

Using Bass Tools™ to add more Thump & Rumble is quite easy. By setting the Bass Tools Style to Soft the algorithm automatically places additional emphasis on lower bass frequencies. If the Style is set to Hard instead, bass “slap” will be more pronounced and the emphasis on thickening the bass that the Soft style creates will be reduced.

“Subwoofer” algorithm places targeted enhancement on the frequencies between 20Hz and 80Hz, and does not add bass harmonics. When the *Sub Bass* control is Off, enhancement of very low frequencies is disabled and bass energy in that region remains just as it was in the incoming program material.

Other ways of enhancing these lower frequencies would be to set the Band 1 to Band 2 crossover at 50 or 60 Hz and raise the threshold of Band 1 in the AGC/Compressor. This will allow very low bass to pass more freely through the AGC and compressor section, with the lower bands of the limiter and Bass Tools acting as a safety net to prevent bass-induced distortion. On the majority of formats we recommended the *Sub Bass* control being left in the “On” position.

### PUNCH/BOOM/TUBBY/WARM

The sound of bass “punch” and “boom” usually fall in the range of 70Hz to 150Hz. “Tubby” on the other hand usually falls between 150Hz and 300Hz. Band #2 of the parametric Equalizer may be used to *gently* and *broadly* boost audio in the range of 75Hz to 120Hz to safely reinforce the “punch” and “boom” of lower bass without creating undesirable effects such as “tubbiness.”

The enhancement of “punch” and “boom” can be especially tricky because many listeners’ sound systems don’t have the ability to reproduce lower and sub bass frequencies. Therefore it’s especially important to use several known reference systems when adjusting the enhancement of very low frequencies because it is easy to create distortion or “muddiness.” How many stations have awesome sounding bass on the PD’s car stereo but end up sounding terrible on clock radios and boom boxes? Always listen on multiple radios in multiple listening environments to get the best feel for the effects created by purposeful bass enhancement.

### MUD

When someone says that audio is “muddy” they usually mean too much energy is in the 250Hz to 400Hz range. Sometimes it’s caused by a misadjusted Equalizer or a processor maladjustment that is allowing an AGC or compressor band to add too much gain. In most cases it’s a good idea to keep from boosting energy in this region. In fact, additional clarity and detail can be created by a broad equalization *cut* in the range between 180Hz and 400Hz. This surprising little secret can be more effective at “adding” detail in low bass and midrange than adding equalizer *boost* at the same frequencies. It’s true! Try it!

### HONK

Think Saxophone. Honk is above “mud” but below “fuzzy” and “tinny.” Honk occurs between about 400Hz and 1,200Hz and begins to overlap “fuzzy” and “tinny” at around 1000Hz. Usually audio in this range stands out on its own in the mix and any enhancement should be slight because our ears are already very sensitive to these frequencies. This frequency range is also quite delicate because much of the perceived stereo sound field width also occurs in this range. When audio has a “honk,” “fuzzy,” or “tinny” texture (and it’s usually a texture that one DOESN’T like), faster attack times in the Compressor for that frequency band can help smooth things out. Another trick is to relocate the Band 3 to Band 4 crossover to around 2,000Hz which allows Band 3 to better manage any “harshness” that might tend to pop out on certain material.

### WHACK

This is the sound where percussion just “explodes.” First, note that this is much less of an equalization issue and much more about creating a *temporarily* dense sound by momentary fast compression. To achieve more “whack” it can help to speed up the AGC or Compressor release times in Band 3. Another trick is to slow down the attack time of the Band 3 AGC which will allow the Band 3 compressor to be more active. This will enhance “whack.”

## FUZZY/TINNY

Yes, “fuzzy” and “tinny” really are this far down in the audio spectrum! A “tinny” sound might be described by some as a harsh midrange or too much presence instead of something happening higher up in the audio spectrum. Both “fuzzy” and “tinny” live between about 1,200Hz and 2,000Hz. One of the best ways to manage either is to broadly and slightly reduce equalization in this range, or even increase the attack time in Band 3 of the AGC and Compressor.

Sometimes there is a tradeoff between getting more “whack” and keeping “fuzzy/tinny” at bay. Speeding up the compressor in the 1,000Hz to 1,500Hz range to get more compression “whack” can sometimes add undesirable side effects, one of which is the creation of what sounds like “fuzzy,” “dense,” or even “tinny.” In audio processing, especially broadcast audio processing, everything is a tradeoff. While Wheatstone processors provide the most wiggle room for getting the sound you hear in your head on the air, sometimes compromises will need to be made.

## CRUNCHY/EDGY

This sound is mostly caused by the need to manage the FM pre-emphasis when aggressive processing is being applied and it usually shows up as artifacts in the 1,600Hz and 2,500Hz range. Sometimes it is best mitigated by rearranging the processing furniture – placing the 31-band limiters *after* the FM pre-emphasis, for instance. In fact, Wheatstone is the only processing company that we are aware of that allows the end user to move the pre-emphasis around. If a particular program format or market dictates that a smoother mid and high frequency range is needed, placing the pre-emphasis *BEFORE* the multiband limiter can help in this regard. While some high frequency transient detail can be lost, the end result will be a very smooth and predictable midrange and high end which may be more suitable for formats that primarily target female listeners.

Speaking of *high end* and *texture*, Wheatstone provides three different main clipper styles in most of our processor models. Each clipper style is specifically designed to create a specific artistic *effect* while also limiting signal peaks. The three clipper styles and their attributes are as follows:

**Hard:** The Hard clipper style is the most forgiving to maladjustment without creating the “crunchy/edgy” sound of other processors. Quite a bit of clipper drive (Lim/Clip Drive) can be dialed in without generating obvious and/or disturbing distortion artifacts.

**Firm:** The Firm clipper style is designed to generate low order harmonics which enhance the audibility of audio details and fullness without generating overtly obvious and/or disturbing distortion artifacts. Yes, the Firm style is more “distorted” than the Hard style – it’s supposed to be. It also generates more obvious loudness benefits than the Hard style without sounding rough or overdriven.

**Round:** The Round clipper style is the most obvious sounding of all three styles. It is *designed* to sound grungy when overdriven but without generating objectionable brightness due to clipping. It is the loudest of all three clipper styles but also the most sensitive to maladjustment.

Which clipper style is best? Wrong question! Which clipper style is the *right* one to use? It completely depends on your format, your competition, and your market. As in all things processing, the ears are always the best judge of what’s right and wrong.



## SIBILANCE

Sibilance is a vocal artifact and because it sounds so unnatural it tends to stick out like a sore thumb. Excessive sibilance can be the result of too much high-end boost or too much final clipper drive. Sibilance tends to be most prevalent in the 4,000Hz to 8,000Hz portion of the frequency spectrum. Microphone processors (like the Wheatstone M1 and M2) are great tools for keeping vocal sibilance under control. Their specialized de-esser sections are specially tailored for removing or minimizing excessive “esses.” When excessive sibilance is an issue and there is no microphone processor to control it, lowering AirAura X1’s Band 4 to Band 5 crossover setting and/or using a slightly faster attack time on the Band 5 AGC/Compressor can help.

Sibilance can sometimes be found in an unlikely place; the L-R. “But voice is mono...” you say, “...so there is no L-R...” Well, yes and no. When the left and right channels don’t have perfect balance or there is phase shift between the channels, energy ends up in the L-R. Why? By definition the L-R signal is the *difference* between the left and right channels, regardless of whether it’s level, or phase, or both. When phase is the culprit the error is generally larger at high frequencies, making the L-R energy also greater at higher frequencies. As if by magic, sibilance appears in the L-R and the only way to fix it is to tend to what’s causing it or reduce the level of L-R at frequencies where sibilance might reside. The AGC’s multiband mixer can help with this. Slightly reducing the setting of the Band 4 and Band 5 L-R mixer can help tame sibilance without having too negative an effect on stereo separation.

## TREBLE/BRIGHT/OPEN

Treble (like its friends Bass and Midrange) is a generalized term for the high end in most broadcast and recording systems. Bright and Open are oft-used descriptive terms of treble styles.

Unfortunately, bright can be a positive or negative term. For instance, laser-bright is usually a negative term used to describe too much enhancement in the upper end. The term probably borrows its origins from the early days of CDs when *brighter* supposedly meant *cleaner*.

Open is a term usually reserved for describing audio texture in the upper midrange when it doesn’t sound overly processed, packed in, or is lacking in detail. To achieve a more open sound in any processor there is usually some sacrifice in loudness. Fortunately AirAura X1 is much more forgiving in this regard and most users have an easier time being loud with AirAura X1 while still being quite clean and open. Operating the AGC and Compressor with slower attack and release times and making some minor tradeoffs in loudness will push perceived quality off the charts while going a long way towards achieving an open and easy to listen to sound. When *this* is the target sound, starting with one of our Classical or Jazz presets and then “turning things up” is a good way to approach achieving this sound. These presets are surprisingly competitive without sounding “processed.”

## FINALLY

As always, Wheatstone’s Tech Support team is available to answer any questions, help with setup, or assist in tweaking your station to the sound you hear in your head but might not know how to achieve. Shoot us an email at [techsupport@wheatstone.com](mailto:techsupport@wheatstone.com) or give us a call at 252-638-7000. We know our processors inside and out and can make them sit up and dance in any market or format.



# Appendix D

## Contents

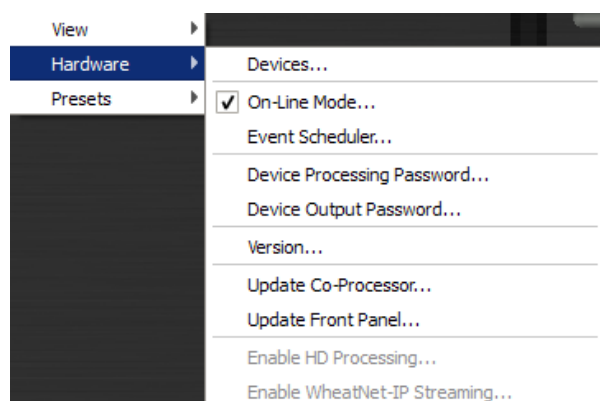
<b>WheatNet-IP Configuration.....</b>	<b>A-20</b>
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## WheatNet-IP Configuration

Along with Digital AES3 and Analog inputs X1 is also equipped with WheatNet-IP (WNIP), the Audio Over Internet Protocol (AoIP) used in our Wheatstone BLADE and IP audio network control surface environment. WheatNet-IP permits the X3's input audio and its FM and HD outputs to be carried over the same standard CAT5 Ethernet cable as that used for remotely controlling the X1.

This procedure assumes that your facility is equipped with at least one Wheatstone BLADE and the associated Navigator software. The BLADE used may even be the multiband, eight processor "Aura8ip" processing BLADE whose physical inputs and outputs (half analog, half digital) may even be used as a substitute for the X1's physical I/O.

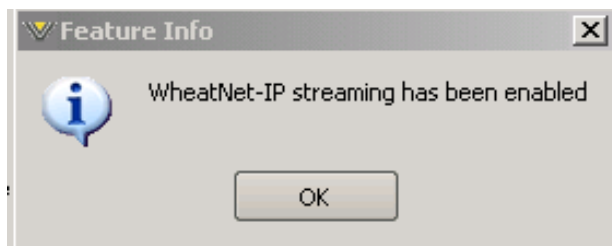
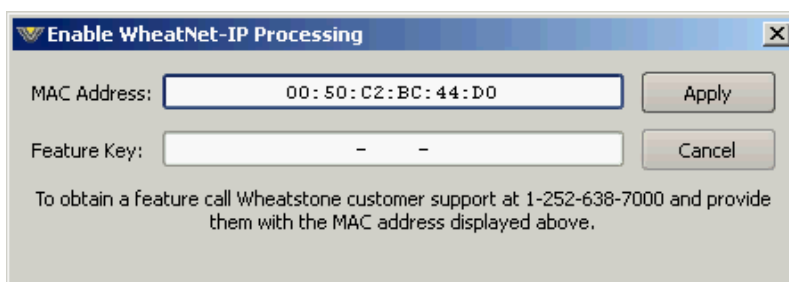
After a major firmware upgrade Wheatnet-IP may need to be re-enabled with a software key generated by Wheatstone Technical Support. If calling to get WNIP turned back on, please have the MAC address of the unit handy because that is required in order to generate the correct enable code for your hardware. Each AirAura has its own unique WNIP enable code, so providing us with the correct MAC address is crucial.



With the GUI configured to connect to the X1 device and online to it, right clicking on the controls area will reveal the dropdown menu at left. We will be navigating the Hardware menu option.

At the bottom of the dropdown list is the option to *Enable WheatNet-IP Streaming*.... Click on that item and another dialog box will open. This is where the special WheatNet-IP Enable code will be entered and the MAC address of your X1 can be seen.

Once the code is entered and the *Apply* button is clicked a message will indicate that streaming has been enabled. Click *OK* to close the dialog.

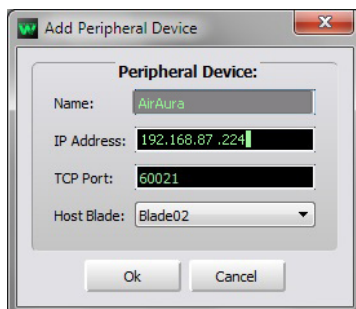
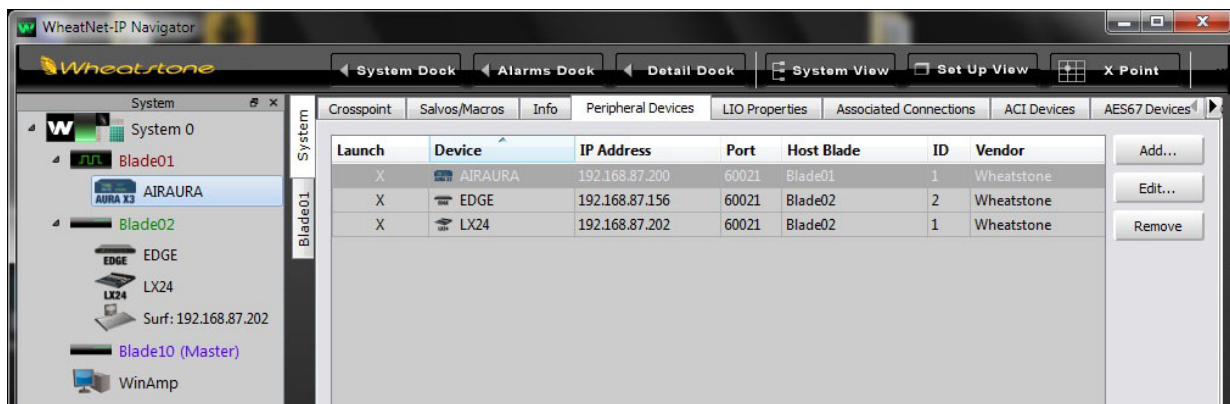


Now it's time to open up WheatNet-IP Navigator to gain access to the X1's WheatNet-IP-associated input and output signals.

Once the WheatNet-IP Navigator software is open and you are logged into the system we must create a new device for X1. Because X1 is not a “BLADE” it appears and operates as a “peripheral device” within the WheatNet-IP system. Our Sideboard, an outboard mixer/controller for use with any BLADE’s pair of 8-channel utility mixers is one example of a peripheral device. Another example of a peripheral device is Tieline’s Genie codec. These devices, while technically not BLADEs, can interact with any WheatNet-IP system and have signals routed to and from them. What they lack, however, is the built-in intelligence of an actual BLADE.

## Adding the X1 to The WheatNet-IP System

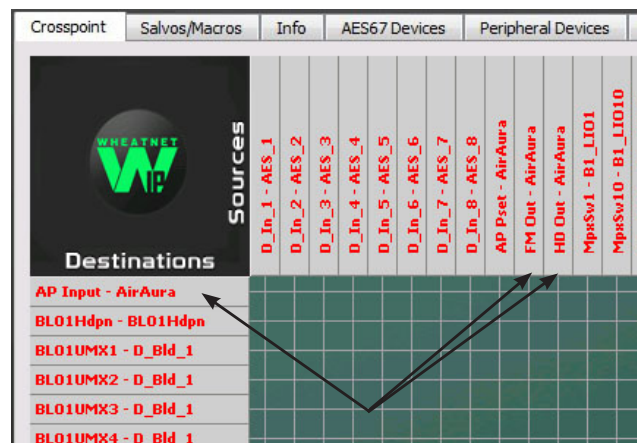
The first thing to do is click on Navigator’s *System Peripheral Devices* tab (in previous Navigator GUI versions this was the *System 3rd Party Devices* tab) which will open a new window listing any peripheral devices currently present. In the image below the X1 device already exists (it’s in our lab) but we will be adding another one nonetheless.



Clicking on the *Add..* button opens a dialog where a new X1 may be added to the system. Here we will add the new X1’s name, its hardware IP address, and tell the system what BLADE will be its host. At this time we do not have to modify the TCP Port setting so we can leave it set to 60021. Once input to the dialog is finished, click on the *Ok* button and it will close.

Next we’ll open the System Crosspoint window and see that our new X1’s Input and FM and HD output signals are now present and available for routing.

Note that as shown in the example on the right we’ve used Navigator’s signal name editing capability to assign the X1’s input and output signals the names we desire. Until changed the default names begin with “AP” and will be “AP Input,” “AP FM Out,” and “AP HD Out.” Of course these signals may be manually crosspointed (and locked if desired!)



to make connections, or it may be done through the many other methods available within the WheatNet-IP system for routing signals and changing the routes of both audio *and* logic signals.

More information on configuring and using WheatNet-IP Navigator and the other features and capabilities of the WheatNet-IP system may be found in the *WheatNet-IP BLADE3 Audio Over IP Network* manual.

Assistance in configuring the X1 for WheatNet-IP may be obtained by giving us a call at 252-638-7000 or by emailing [techsupport@wheatstone.com](mailto:techsupport@wheatstone.com).