AirAura

Digital Spectral Processor

Technical
Manual

Wheatstone Corporation
March 2012

VORSIS ULTRA-HIGH RESOLUTION PROCESSING

AirAura Digital Spectral Processor Technical Manual - *Software Version 2.4.0 and higher*

©2012 Wheatstone Corporation

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

Attention!

Federal Communications Commission (FCC) Compliance Notice:
Radio Frequency Notice

NOTE: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

This is a Class A product. In a domestic environment, this product may cause radio interference, in which case, the user may be required to take appropriate measures.

This equipment must be installed and wired properly in order to assure compliance with FCC regulations.

Caution! ***Any modifications not expressly approved in writing by Wheatstone could void the user's authority to operate this equipment.***

The History of Wheatstone Audio Processors

Introduced in 2005, the Vorsis product line evolved from Wheatstone's return to its original roots in audio processing. Having designed and integrated analog and then, later, complex digital audio processing, into their radio and television consoles and control surfaces, Wheatstone was perfectly poised to address the challenges of combining very high audio quality with competitive on air loudness.

Wheatstone audio processors are built to the same exacting standards as all Wheatstone products. Research and development, manufacturing, testing, and quality control for the line of audio processors are all accomplished in Wheatstone's large state-of-the-art facility located in New Bern, North Carolina. Keeping everything under one roof allows Wheatstone to have control over every facet of product production and ensures that our customers receive products of the highest possible quality *and* reliability. Staying true to "Made in the USA," Wheatstone does not utilize offshore manufacturing.

A dozen experts with deep experience in Digital Signal Processing, broadcast audio processing and other engineering disciplines comprise our design team. Led by audio processing expert and broadcast engineering veteran Jeff Keith, the team combines their talents to design and build audio processors that achieve new and higher standards of on air sound quality.

In its short history Wheatstone Processing has already invented many new and unique audio processing algorithms – algorithms that push audio processor performance to new and higher levels. Wheatstone was the *first* to develop a Vorsis intelligent, "program density aware" AGC (2007 - Sweet Spot Technology, or SST). We were also the first to employ the science of human psychoacoustics in the design of a multiband limiter (2005, our acclaimed 31-band final limiter). In addition, our well-regarded Vorsis Bass Management System (2007, our VBMS) solves the bass intermodulation problems that plague other brands of audio processors when pushed for competitive loudness.

Each Wheatstone Processing product comes equipped with its own Windows® software-based intuitive Graphical User Interface for control of the processor. Carefully tuned factory presets ensure that our audio processors can be placed into use quickly and easily and in any size market with a minimum of effort.

AirAura Technical Manual

Table of Contents

Chapter 1 - General Information

Chapter 2 - Front Panel GUI

Chapter 3 - PC Based Audio Processing Guru® GUI

Chapter 4 - AirAura Pro GUI

Appendices

Appendix A

Appendix B

Appendix C

Appendix D

Appendix E

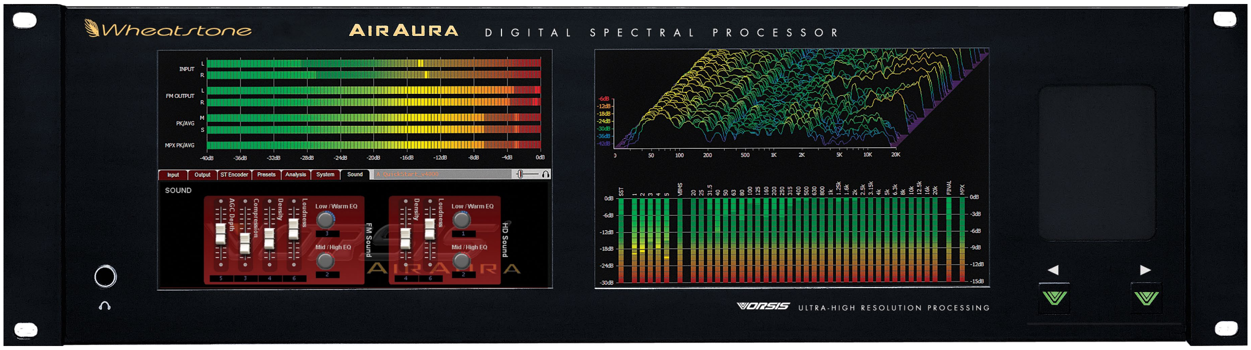
Appendix F

Appendix G

Appendix H

General Information

Chapter Contents



General Information

Introduction

Welcome to AirAura, Wheatstone's third generation top-of-the-line audio processor. AirAura has been carefully designed to deliver a clean and easy to listen to sound on the dial along with very competitive loudness when that is the goal. AirAura expands on technologies previously invented by Vorsis, incorporates several new ones, and delivers that 'sought after' sound to both FM and HD medias. Like other Wheatstone processors, AirAura has been carefully designed to not have a fixed sound of its own – the end user can use AirAura's vast pallet of tools to create any desired on air sound.

AirAura's processing algorithms have been carefully evaluated though thousands of hours of critical listening with a very wide range of program material. Experience on real radio stations in markets of every size has revealed that AirAura's processing algorithms deliver *amazingly clean, clear, and if desired, loud, on-air sound*. Music sounds like it was meant to sound. Voices are rich and full and without unnatural coloration or distortion. Transitions between widely varying program elements are deftly handled, and if desired, with no trace of "processing" during spectral balance and density corrections.

Remote control of AirAura is via Windows-based Graphical User Interface (GUI) software and can be accomplished via 100BaseT wired Ethernet and 802.11b/g wireless; simultaneously, too, if you wish! Up to four simultaneous remote control sessions are possible using any mix of wired and wireless Ethernet. Eight rear panel General Purpose Inputs (GPI) and four General Purpose Outputs (GPO) are provided to allow hardware-level interfacing of AirAura to station systems.

AirAura utilizes an internal base sampling rate of 192 kHz. Higher sample rates are used where it is beneficial to do so, such as within the AirAura clipper. A 24-bit, 144dB internal dynamic range signal path allows audio to be processed with *extreme* precision. Twin front panel LCD screens driven by a Linux-based industrial-grade CPU and a front panel touch-pad permit convenient control of system and basic processing setup parameters. Equipped with the popular Vorsis "Audio Processing Guru®" right on the front panel, the sound of any factory preset can be easily customized to over a million different combinations.

Those wishing for complete control over all processing parameters will appreciate our full-control Professional GUI which is designed for processing experts. This advanced GUI is available free of charge by registering for it at http:wheatstone-processing.com. Registered users of the AirAura Professional GUI have access to well over 400 individual processing adjustments.

Wheatstone products are designed and manufactured in our 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 design and manufacturing process. In order to maintain the high product quality that Wheatstone has been known for over nearly 40 years, no offshore manufacturing is utilized.

The following pages will help you get AirAura up and running as quickly as possible without having to know a lot of techie stuff.

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.

AirAura Feature Overview

The AirAura Digital Spectral 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 AGC with SST, the two signal paths are equipped with their own 31-band limiters and final peak control sections, each optimized for the intended medium.

AirAura's five band AGC incorporates our SST (Sweet Spot Technology™) and operates through a proprietary technique, Density Compensation™. The Density Compensation™ algorithm utilizes real-time analysis *and* data from the user controls to invisibly maintain the *user set* spectral balance *and* program density as program elements change. Exclusive Speech Smart Technology™ ensures natural and clean voice at *any* loudness level. The output of the five band AGC/SST is then split into two signal paths, FM and HD.

Unlike processors with fewer limiter bands, Vorsis taps into human psychoacoustics to render operation of the 31 limiter bands inaudible. While limiting only the discrete frequencies needed, subtle audio detail in adjacent limiter bands is revealed – detail not heard in other processors. The 31-band technology has similarities to the precision filter banks used in high-performance perceptual codecs. Positive attributes shared with perceptual codecs include low latency with good group delay and phase behavior. The 31-band limiters rely on a well-researched characteristic of human hearing known as "critical band masking" to hide operation of individual limiter bands from the ear.

The FM signal path utilizes our new AirAura final clipper technology. A selectable look-ahead limiter is also available which may be used in combination with the clipper if desired. A digital stereo generator follows the AirAura clipper and generates two multiplex composite stereo outputs and accommodates two SCA inputs. The rear panel BNC connector for SCA-2 may be switched internally to provide a 19kHz stereo pilot square wave "Sync" output for RDS applications that require a 19kHz synchronization signal.

The HD signal path is also equipped with a 31-band limiter and lookahead final limiter, allowing 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, the latter to accommodate a software bug in certain versions of HD radio encoder firmware.

A fully routable headphone monitoring path allows the audio in various parts of the processing chain to be auditioned. The selected Input Source and the unselected Analog and Digital inputs may be monitored separately, and unselected inputs may be monitored without putting them on the air. The FM and HD processing paths have multiple headphone monitor points including the final processing output.

Several dozen factory presets are provided with AirAura, making a very wide range of on air sounds and textures available to the user right out of the box. When used in combination with the front panel's Audio Processing Guru controls each factory preset offers over a *million* different on-air sounds to the user.

The System Menu of the GUI provides access to two preset scheduling utilities. As well as the usual short-term daily scheduler, there is also a "long-form" scheduler that can be programmed for automatic preset changes on dates and times 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 "special" presets.

GPI (General Purpose Inputs) provide easy access to preset changes via external events such as contact closures and automation control. AirAura is also equipped with Wheatstone's 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 using this feature).

Rack Mounting

AirAura 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.

AirAura 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 AirAura as long as the unit is being operated in a reasonable environment (below 104F or 40C).

AirAura 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 AirAura and devices immediately above and/or below it.

WARNING!

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

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

These voltages are hazardous and potentially deadly if accidentally contacted.

Special tools, software, and fixtures are required for service. There are no user-serviceable parts inside.

AirAura must be returned to Wheatstone Corporation under a Return Authorization in the unlikely event that repair is necessary.

AirAura Installation Tips

Grounding

Establish a low impedance common ground in the facility and try to route all equipment grounds to that point. Use ground conductors with the largest possible surface area and keep ground leads as short as possible. AirAura's ground reference is its chassis, which should be connected to the station ground. Such a connection is especially important when AirAura is operated in a high RF environment because it helps minimize differential voltages between the processor's chassis and other pieces of equipment such as the rack it is mounted within.

Surge Protection

Always place surge protection circuits as close as possible to the device being protected. AC power line surge protection should manage transients 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 AirAura), particularly when the equipment is located within the electrically hostile environment of a station's transmitter facility.

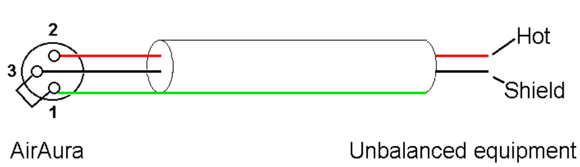
UPS/Power Conditioning

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:

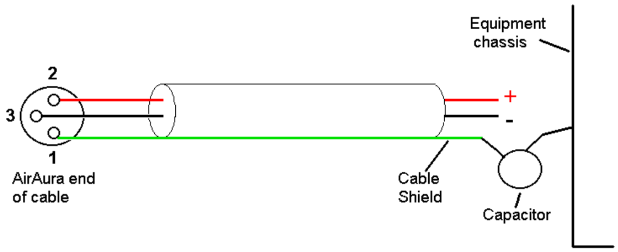
- How does the UPS behave when AC power is not exactly 60Hz, such as when the facility is on its backup generator?
- If the UPS has onboard surge protection, what kind of surge capability does it have and where are those surges directed to?
- Is the UPS equipped with remote monitoring capability?
- Does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

Analog Audio Input Connections

Balanced audio *input* sources are recommended and should be connected to AirAura 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 the "+" output to AirAura's "Hi" input (XLR pin2) and connect the shield wire to AirAura's "Lo" input (XLR pin-3). If the cable's shield is used (recommended) connect it at the AirAura end only (XLR pin 1) to prevent AC ground loops.



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

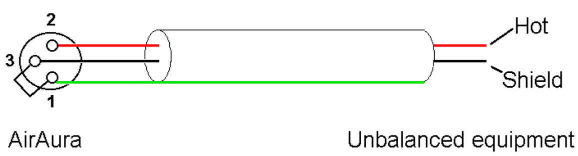
TDK	CS17-F2GA103MYGS
Murata	DE2F3KH103MA3B
Panasonic	ECK-ATS103MF
AVX	65N103MBLCP

Analog Audio Output Connections

Balanced audio loads are recommended and should be connected to AirAura'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).

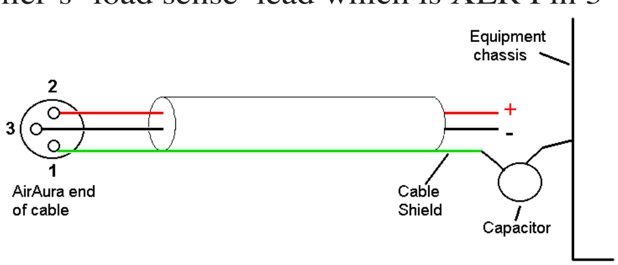
AirAura is equipped with an active balanced output stage that behaves like a transformer. Because of this the correct wiring method may be different than expected.

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



The unbalanced load's shield should be connected to AirAura's output XLR Pin 3 ("Lo"). Then, and to enable AirAura's balanced output amplifier to operate correctly when driving the unbalanced load, it is recommended that AirAura's XLR output Pin 1 (ground/shield) also be connected to Pin 3 ("Lo"), noting that this MUST BE DONE at the AirAura 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. Doing so can induce external noise and crosstalk on the output amplifier's 'load sense' lead which is XLR Pin 3 whenever 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 0.01uF 250V AC *rated* capacitor to see if it helps. Please refer to the listing above for capacitors known to be suitable for this task.

Digital Audio Connections

For digital audio connections always use a good quality balanced 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 including 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 usually have higher capacitance than digital cable and high capacitance cables can impair the ability of the AES3 receiver to recover the digital signal without errors. Increased jitter, dropouts, or no audio at all can be an indication of an improper cable type.

Where to Install AirAura

The best location to install AirAura 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 AirAura's built-in lab-grade stereo encoder which allows much tighter control of modulation peaks.

A transmitter site location has the additional benefit of allowing the use of AirAura's highly oversampled composite processor to gain an additional loudness advantage. The Vorsis composite processor is much cleaner and more forgiving than those in other products, and in combination with AirAura's tight pilot and SCA protection filters can create additional loudness without the audible grunge that composite clippers typically create.

We recommend that, whenever there is a choice between using an exciter's composite MPX or AES3 input, the processor (any processor) should be interfaced to the transmitter using the exciter's composite stereo multiplex input. The exciter's AES3 digital input may be 'clean' and it may be 'digital,' but it also precludes the ability to gain additional loudness through the use of AirAura's intelligent oversampled composite clipper. Also, depending on several 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 AirAura 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 employs codec-based audio compression AirAura should be located at the transmitter site which places it *after* the codec. This is very important because 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 AirAura, creating a modulation (loudness) disadvantage.

Installing AirAura at the studio end of a "compressed" STL brings with it at least two caveats:

- AirAura's stereo generator and composite clipper will not be available. Many digital exciters offer stereo generator and composite clipper functions, but their clippers have historically been quite crude and spectrally "dirty" compared to AirAura'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 has been pre-emphasized. 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 far fewer opportunities to remove redundant audio information and then *mask* that removal from our hearing. When handling heavily processed (limited dynamic range) material, codec operation can be 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 AirAura at the studio end of the STL will preclude the use of AirAura's stereo generator and composite clipper.

TIP: If using AirAura 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 AirAura. This will prevent gross distortion and potentially large modulation overshoots.

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, robbing modulation and reducing on-air loudness.

Composite Analog STL

A high quality analog composite STL has some advantages over an analog left/right STL in that it will typically have broader bandwidth and better audio performance than a discrete analog STL. Most also have the capability to add subcarriers for SCA and RDS along with the composite audio. This means that many SCA and RDS generators may 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 tight frequency response and phase matching requirements. Furthermore, in many countries wideband 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.

Where Should Pre-Emphasis Go?

Pre-emphasis should ***always*** be applied by the audio processing and ***never*** by the exciter. 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. *No FM exciters have this technology.*

To summarize: the best location overall for the audio processor is always at the transmitter.

Arbitron People Meter (PPM)

Field experience has indicated that AirAura favorably passes the data watermarking scheme used in the Arbitron People Meter rating service technology, regardless of the aggressiveness of the processing being performed.

AC Power Considerations

Please note that in order to enhance its long-term reliability AirAura has no power switch because *all* power switches notoriously become intermittent over time without regular use.

AirAura 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. AirAura is, after all, a highly specialized "computer."

The use of a UPS as recommended will protect AirAura from short duration power interruptions and glitches which might otherwise signal it to reboot. When AirAura reboots there will be a loss of audio for approximately 15 seconds.

Rear Panel Connections

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

AirAura Rear Panel Connections

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

Bottom Row

Analog Left Channel In
Analog Right Channel In
FM Analog Left Channel Out*
FM Analog Right Channel Out*
HD Analog Left Channel Out
HD Analog Right Channel Out
AES Digital Input
FM AES Digital Out
HD AES Digital Out
GPI Input
Ethernet 1/Wheatnet-IP Interface
Ethernet 2/Wheatnet-IP Interface
IEC Standard male AC Power Input

Top Row

SCA-1 Input
SCA-2 Input
TX-1 Out
TX-2 Out
WiFi Diversity Antenna 1
WiFi Diversity Antenna 2
GPO Status Output

** When AirAura's output mode is set to Composite Multiplex, the FM Analog left and right output XLR connectors emit a balanced Composite Multiplex signal. A balanced signal is preferred when interfacing with exciters having a floating and differential MPX input (most modern exciters do) and can provide a lower overall noise floor than an unbalanced MPX circuit. Please see the appendix for how to prepare the special XLR to BNC cable required if using this feature.*

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

- Nine XLR connectors are provided for analog and digital audio input and output connections.
- Four BNC connectors are provided for SCA inputs and TX (transmitter) out, or Composite Multiplex (MPX) connections.
- Two RJ-45 connectors are provided for Ethernet connections. Either connector may be used to interface AirAura to a WheatNet-IP audio network or to connect it to a Windows® PC running the Vorsis AirAura 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 1-18 through 1-21 summarize wiring connections.

Audio Inputs

AirAuras which are equipped with software version 2.4.x and up accept *three* types of audio input sources:

- 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 <u>on both channels</u> 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.
- Balanced analog left/right stereo, de-emphasized according to pre-emphasis in use.
- AES3 digital, either pre or post diversity delay, and/or de-emphasized according to pre-emphasis in use.
- *Balanced* line level composite stereo multiplex on male XLR connectors.
- Unbalanced composite stereo on two BNC female connectors.
- WheatNet-IP audio network.

HD Path

Output audio for the HD path is available as:

- Balanced analog left/right analog.
- AES3 digital.
- WheatNet-IP audio network.

Analog Out – XLR-M

Pin 1 XLR FM LT SH – LINE FM LT OUT SH
Pin 2 XLR FM LT HI – LINE FM LT OUT HI
Pin 3 XLR FM LT LO – LINE FM LT OUT LO

Pin 1 XLR FM RT SH – LINE FM RT OUT SH
Pin 2 XLR FM RT HI – LINE FM RT OUT HI
Pin 3 XLR FM RT LO – LINE FM RT OUT LO

Pin 1 XLR HD LT SH – LINE HD LT OUT SH
Pin 2 XLR HD LT HI – LINE HD LT OUT HI
Pin 3 XLR HD LT LO – LINE HD LT OUT LO

Pin 1 XLR HD RT SH – LINE HD RT OUT SH
Pin 2 XLR HD RT HI – LINE HD RT OUT HI
Pin 3 XLR HD RT LO – LINE HD RT OUT LO

AES Out – XLR-M

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

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 output located on the left side of the AirAura 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.

Network Connections

AirAura is equipped with two methods of remote control connectivity:

- Wired Ethernet via two 100Base-T Ethernet ports on the rear panel. These ports are 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 GUIs.
- 802.11b/g compatible wireless interface. The wireless interface is based on a high performance, industrial class WLAN WiFi transceiver module equipped with diversity receivers and twin antennas. The salient features of the module are as follows:
    - 802.11b/g WLAN (Wi-Fi) standards-based technology.
    - Extended temperature and environmental specifications.
    - Built-in TCP/IP and UDP features allow flexible LAN connectivity options.
    - Built in WEP, WPA, and LEAP security protocols.
    - Up to four simultaneous “wireless” GUI connections.

With the factory default configuration of the wireless interface, AirAura is an ad-hoc wireless device with no security which provides the least complex out-of-the-box first use for the majority of end users. Security options within the wireless interface’s web browser-based configuration utility allow for password protection and connections via various security protocols if desired. Please see the appendix for detailed programming information for the wireless interface.

AirAura is not a desirable hacking target and its wireless interface cannot serve as a gateway to a private LAN. This is because the wireless module acts only as a “bridge” between the single fixed IP address on its WLAN side and a single fixed IP address on the Ethernet side which is AirAura’s hardware IP address. All Ethernet traffic is constrained between those two addresses, leaving no possibility of a connection on the

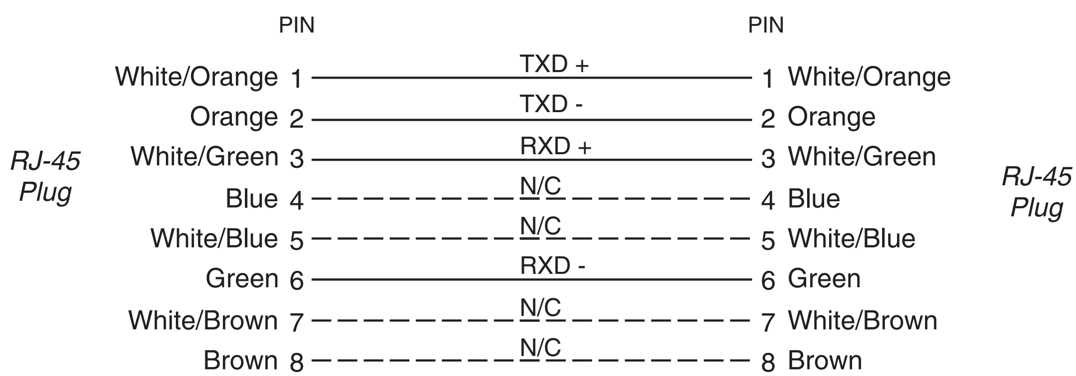
WLAN side tunneling to a different IP address on the wired side. This is an important concept to grasp and this “limited” functionality was purposely designed to protect a station’s Ethernet-based business network from any rogue access via AirAura’s wireless interface.

Important: AirAura’s 802.11b/g wireless interface is intended for remote control only. The wireless interface does not support the WheatNet-IP audio transport.

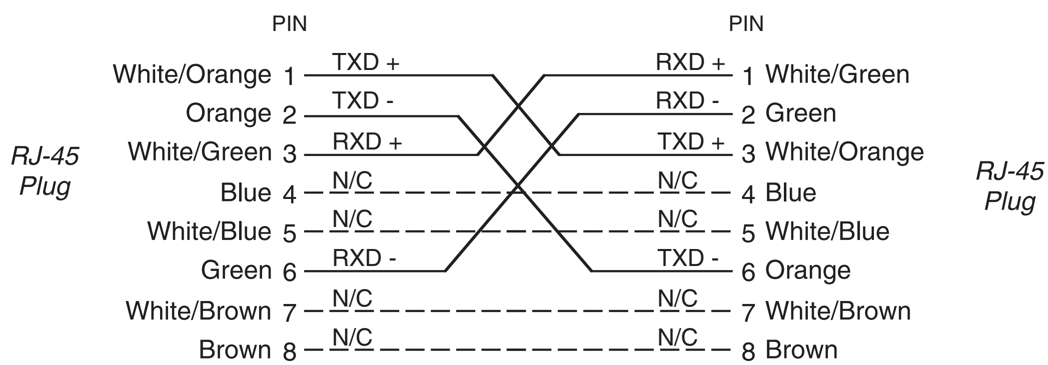
Ethernet – RJ-45

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 Ethernet Cable



Typical Crossover Cable



General Purpose Interface (GPI/GPO)

AirAura 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 AirAura's internal circuitry to prevent external ground loops and to prevent dangerous voltages from being introduced into AirAura. 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 AirAura 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 AirAura 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

Processing Presets

AirAura comes equipped with several dozen factory presets and it can hold a total of 80 presets in its onboard memory. Customer-created presets may be saved within AirAura'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.

In order to prevent annoying clicks and pops when changing presets, preset parameters are slewed between the current values and the new values whenever a new preset is taken and it may take several seconds for the new settings to completely settle in. It is important to remember this concept!

AirAura's preset behavior is purposely designed to make preset changes as unobtrusive as possible and therefore parameter changes are not instantaneous, but gracefully take place over a matter of seconds. This factor must be taken into consideration whenever switching back and forth 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 AirAura's hardware the preset name will again be displayed in green.

Factory presets can be retuned and saved to new preset 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 broadcast audio 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:

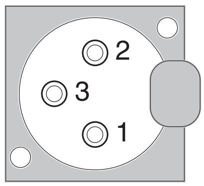
- Presets stored in slots 1 through 8 may be activated through the optoisolated GPI interface.
- Factory presets occupy slots 9 and up, with slot #9 always hosting the Factory BYPASS preset.

- User presets, other than the GPI selectable presets in slots 1 through 8, may be stored above the highest-numbered factory preset. The number of available user preset slots depends on the number of factory supplied presets which may vary by software version.
- The total number of presets on AirAura's hardware is limited to 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's locked presets.

Analog XLR Connections

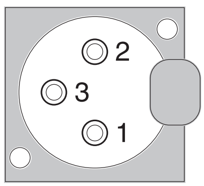
Analog In - XLR-F

XLR LT-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

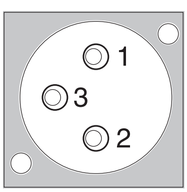
XLR RT-F



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

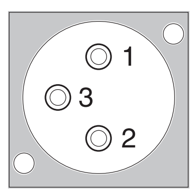
Analog Out - XLR-M

XLR FM LT-M



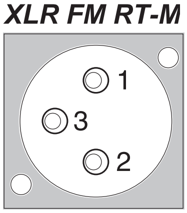
PIN 1 XLR FM LT SH - LINE FM LT OUT SH
PIN 2 XLR FM LT HI - LINE FM LT OUT HI
PIN 3 XLR FM LT LO - LINE FM LT OUT LO

XLR HD LT-M



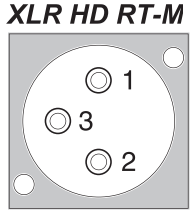
PIN 1 XLR HD LT SH - LINE HD LT OUT SH
PIN 2 XLR HD LT HI - LINE HD LT OUT HI
PIN 3 XLR HD LT LO - LINE HD LT OUT LO

XLR FM RT-M



PIN 1 XLR FM RT SH - LINE FM RT OUT SH
PIN 2 XLR FM RT HI - LINE FM RT OUT HI
PIN 3 XLR FM RT LO - LINE FM RT OUT LO

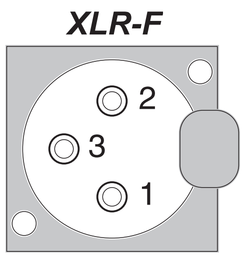
XLR HD RT-M



PIN 1 XLR HD RT SH - LINE HD RT OUT SH
PIN 2 XLR HD RT HI - LINE HD RT OUT HI
PIN 3 XLR HD RT LO - LINE HD RT OUT LO

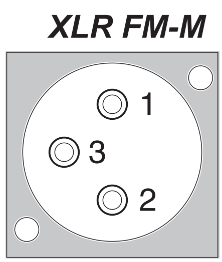
Digital XLR Connections

AES In - XLR-F

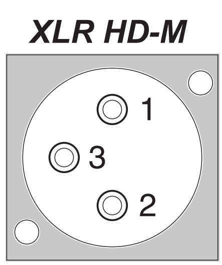


PIN 1 XLR SH - AES IN SH
PIN 2 XLR HI - AES IN HI
PIN 3 XLR LO - AES IN LO

AES Out - XLR-M

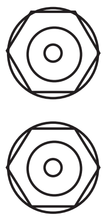


PIN 1 XLR FM SH - AES FM OUT SH
PIN 2 XLR FM HI - AES FM OUT HI
PIN 3 XLR FM LO - AES FM OUT LO



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

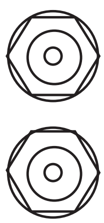
BNC Connections



1
PIN 1 BNC 1 HI - SCA 1 IN HI - CENTER PIN
PIN 2 BNC 1 SH - SCA 1 IN SH - SHELL

BNC - SCA

2
PIN 1 BNC 2 HI - SCA 2 IN HI - CENTER PIN
PIN 2 BNC 2 SH - SCA 2 IN SH - SHELL



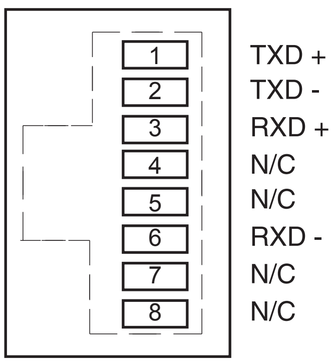
1
PIN 1 BNC 1 HI - TX 1 OUT HI - CENTER PIN
PIN 2 BNC 1 SH - TX 1 OUT SH - SHELL

BNC - TX

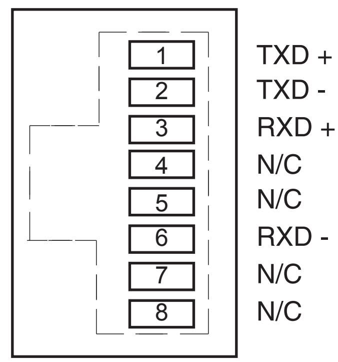
2
PIN 1 BNC 2 HI - TX 2 OUT HI - CENTER PIN
PIN 2 BNC 2 SH - TX 2 OUT SH - SHELL

Ethernet - RJ-45

RJ-45 ETH1

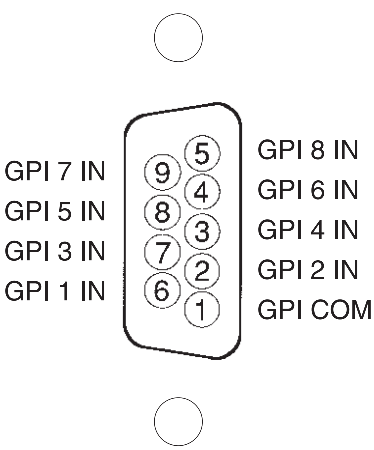


RJ-45 ETH2

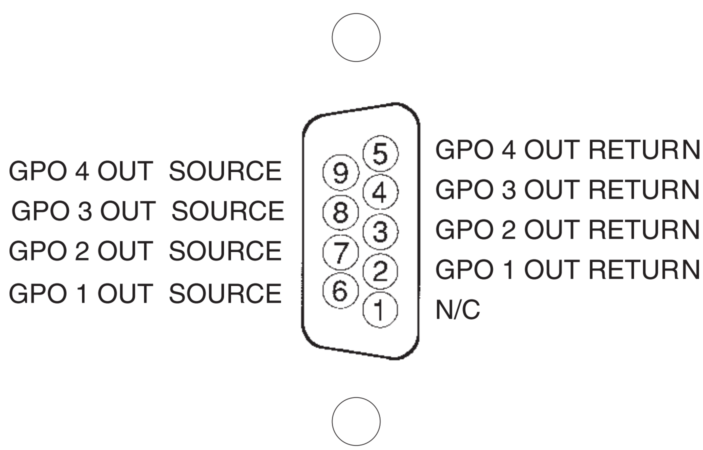


General Purpose Interface - DB-9

GPI - DB-9



GPO - DB-9



Front Panel GUI

Chapter Contents

Front Panel GUI

In addition to comprehensive metering and analysis displays, the front panel also hosts our highly-regarded "Audio Processing Guru®" user interface. This interface condenses more than 400 individual processing adjustments into six easy to use controls. These six controls can morph each factory preset into over 1.7 million (actually 1,771,561) different on air sounds and textures.

Additional information about "Audio Processing Guru®" is presented in its own chapter.

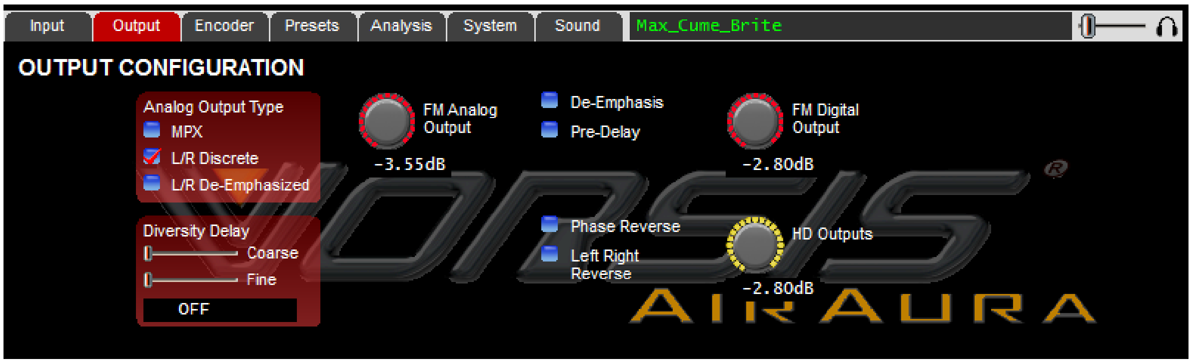
Basic Setup

Once AirAura is rack mounted and connected to a clean source of AC power it is ready for operation. Basic setup and adjustment may be accomplished using the front panel Guru GUI and its tabbed navigation. There are seven tabbed options across the top of the control area: *Input, Output, Encoder, Presets, Analysis, System, and Sound*. These tabbed menus contain all of the controls required to install AirAura, get it on the air, and make modifications to factory presets.

- Connect the Analog or Digital audio inputs as appropriate for your installation.
- Click on the *Input* menu tab of the front panel GUI and select the primary input source to match the type of audio source connected to the rear panel. If the audio source is already providing an audio feed you should see meter activity on AirAura's displays.

- While in the *Input* menu and observing the Input meters, adjust the input gain control (*Digital Gain* or *Analog Gain* as appropriate) until the input meters are just peaking at -12dBFS with normal program levels coming from the studio or program source. Setting input levels correctly is an important step towards getting the best audio performance from AirAura or any on-air processor. Note that the input meters are sample-accurate and include peak (dot) over average (bar) displays, and it is normal to see the average levels reading quite low (perhaps below -20dBFS on some program material) when feeding AirAura with unprocessed audio.
- For best overall audio chain performance, the Input of the AirAura should be calibrated so that it reaches 0dBFS when the signal chain *prior* to the AirAura reaches its 0dBFS, or digital full scale level. This aligns the overload point of the audio chain prior to AirAura to that *of* AirAura, setting levels for maximum headroom and best signal to noise.

- Set the Pre-Emphasis to 75μS or 50μS as appropriate for your region.
- Now make the audio output connections appropriate for your installation, noting that the Analog and Digital outputs may be used simultaneously.



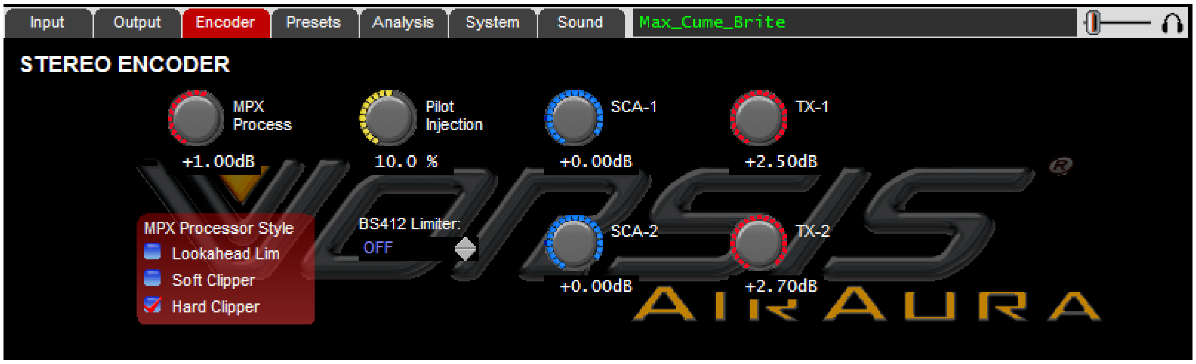
- Navigate to the *Output Configuration* screen of the GUI (shown above). If *L/R Discrete* or *L/R De-Emphasized* has been selected as the Analog Output Type, the *FM Analog Output* control will be visible. When the *MPX* option is selected, this control will be hidden. If L/R is the output choice, also select the Discrete (flat) or De-Emphasized option as required for your installation.
- Note that if *Pre-Emphasis* has been enabled in the *Input* menu all FM outputs will be pre-emphasized *unless De-Emphasis* for that output is explicitly selected.
- Front panel adjustment of the Diversity Delay using the trackpad has enough resolution to get the diversity delay in the ballpark, and finer adjustments (down to 100 microsecond steps) can be made using Guru GUI to set the delay.

  For best resolution of the *Diversity Delay* controls they are best adjusted using the host PC's keyboard and mouse and using the following table as a guide.

COARSE ADJUST SLIDER			FINE ADJUST SLIDER	
Slider only	500	mS	Slider only	0.9 mS
Slider plus Shift key	1700	mS	Slider plus Shift key	8.6 mS
Slider plus Ctrl key	50	mS	Slider plus Ctrl key	0.1 mS (100μSec)

Note that the smallest increment of delay time adjustment is 100 *microseconds*.

- Adjust the *FM Analog Output* control to set the correct Left and Right output level.
- If using the AES Digital Output, connect it to the equipment following the AirAura and adjust the *FM Digital Output* level control in the *Output* menu as appropriate for your installation.
- If using AirAura's built-in stereo generator, select *MPX* as the *Analog Output Type* in the *Output* menu. The *FM Analog Output* level control will disappear. Navigate to the *Encoder* menu (shown on the next page) and adjust the TX, SCA, and Pilot Injection controls as necessary to set the correct modulation levels.



• If the digital outputs are being used, their output levels can be adjusted within the *Output Configuration* screen. Whenever the HD output is being used, the FM diversity delay will probably also be used. Please adjust it as necessary for your system.

• Next click on the *Presets* tab to reveal the list of Factory supplied presets.

When the unit is powered up the first time the Factory Default "A Quick Start" preset will be active. This preset was carefully tuned to provide a balanced mixture of "competitiveness" and "moderate" processing. It provides a very good starting point for a variety of program formats and often ends up being the preset of choice. Even if it's not the final on-air preset, "A Quick Start" offers a rich and balanced sound with reasonable loudness – it won't embarrass anyone when put on the air.

Please explore the entire list of presets to see how each one sounds and which one gets closest to your on-air sound goal, but PLEASE remember **one very important point**:

Preset names don't necessarily mean that the preset can or should ONLY be utilized for a program format with a similar name. Listen to each preset to see how it sounds on your station, in your market, and with your program material.

Remember again – AirAura *slews* preset settings from the old to the new to prevent annoying clicks and pops on the air. Newly selected presets will always take a few seconds to "settle in" to the final sound.

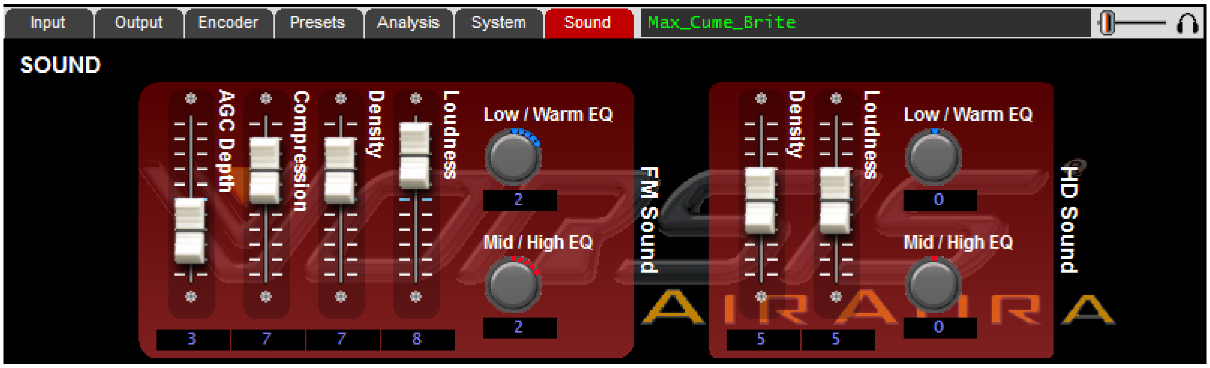
From time to time we add new presets to our Vorsis website for our customers to download. Presets can easily be installed into AirAura using the Preset Library management features built into the remote GUI.

You're on the air!

NOTE: Detailed information on the particulars of the GUI application on the front panel can be found in the following sections which also cover the operation of the PC-based Guru® GUI application.

Modifying Factory Presets Using Front Panel Guru® GUI

Selecting the *Sound* tab will reveal the ten controls of Guru GUI, six for FM and four for HD. It is these simple to use but quite powerful controls that can morph the sound of any Factory preset into over one million different sounds and textures.



There are two sections to the *Sound* screen. The left side contains the controls that modify the sound of the FM signal path while the far right side hosts the controls for adjusting the sound of the HD signal path. Note that the FM and HD signal paths share the common five band AGC/SST structure so as to maintain a consistent spectral texture as an HD receiver blends from analog to HD. The *AGC Depth* and *Compression* sliders are *shared* between the FM and HD signal paths.

Each sound slider is intelligently connected to myriad controls (over 425) that are accessible via the full-control "Expert" level Professional GUI. As each Sound slider is adjusted to the eleven positions within its 0 to 10 range, many complementary adjustments are occurring behind the scenes to modify the sound according to the naming and purpose of the controls. The following is an explanation of what each FM Sound slider control does, again keeping in mind that the *AGC Depth* and *Compression* slider are shared between the FM and HD processing chains.

FM Sound Adjust

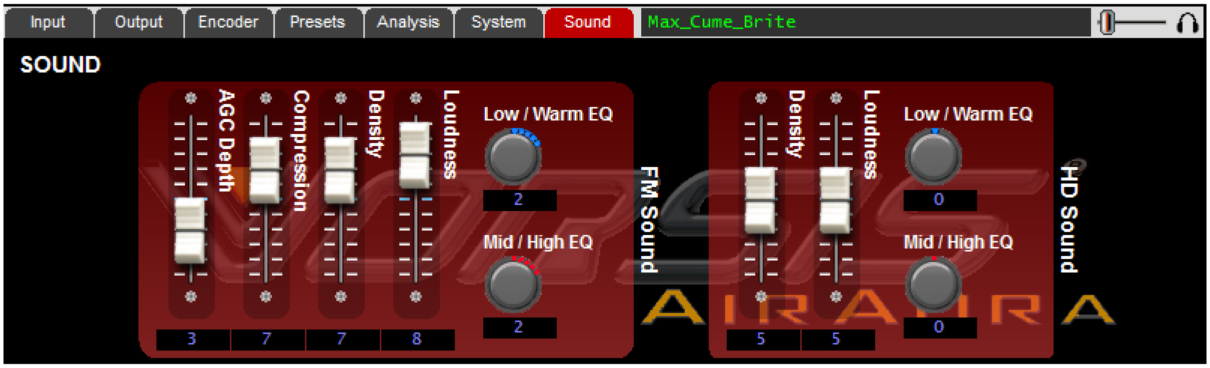
AGC Depth

The *AGC Depth* control adjusts how deep the AGC's *long term* gain reduction will be with typical program material and with appropriate input levels. Higher settings cause deeper long-term gain reduction and the additional "reserve" gain caused by raising the setting of this control will tend to hold up fading songs longer. Conversely a lower setting of the control causes gain reduction that is less deep, leaving less gain available to hold up program levels of material (such as songs) as it fades.

Note that the *AGC Depth* slider does not modify the time constants within the AGC/SST section. That is the job of the *Compression* slider, discussed next.

Compression

The *Compression* control adjusts the medium term program density and how obvious the processing will sound. Lower settings cause the processing to be virtually inaudible as it corrects levels and spectral balance as incoming program elements change. Adjusting



the slider to higher numbers makes the gain controlling action more audible. Higher Compression settings also cause the audio to "fill up" and become more "full."

Density

The *Density* control adjusts how much short-term limiting is permitted. Higher settings will make the audio more dense and "processed" sounding while lower settings will be less so. Stations looking for a "compressed" sound will find the *Compression* and *Density* controls quite useful for tailoring the "feel" of their audio. Note that increasing the Density setting will not cause the final clipper to be overdriven because GUI Guru's intelligence makes other adjustments behind the scenes to prevent it. In other words, distortion will not increase simply because Density was increased.

Low/Warm EQ (we're leaving Loudness for last!)

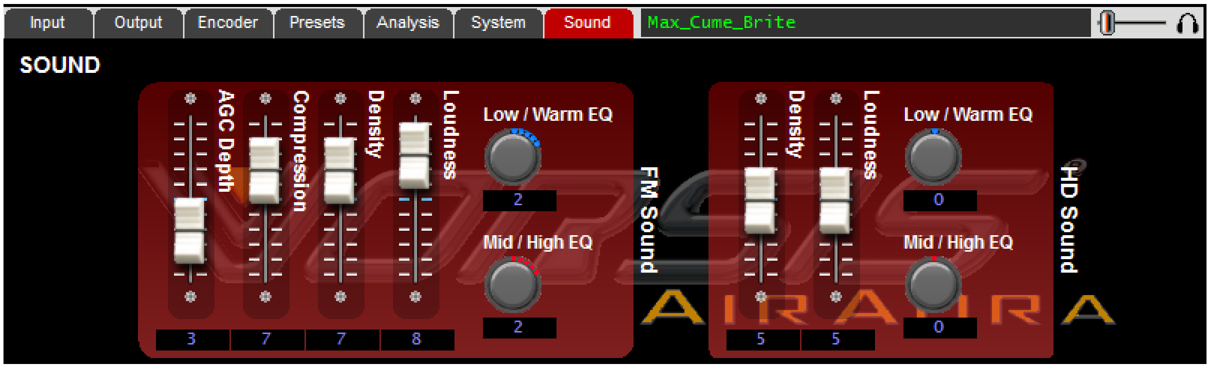
The *Low/Warm EQ* (Equalization) control is the one with the red ring around it and it adjusts how powerful low frequencies sound on the air. When the control is at "0" and has not been touched since a preset was last taken the factory assigned equalization for that preset is being applied. Adjusting the control either side of "0" increases or decreases the factory equalization settings. In general, setting the control to lower numbers reduces on-air bass while moving it to higher numbers increases it.

Mid/High EQ

The *Mid/High EQ* control is the one with the blue ring around it and it adjusts how the higher frequencies sound on the air. When the control is at "0" and has not been touched since a preset was last taken the factory assigned equalization for that preset is being applied. Adjusting the control either side of "0" increases or decreases the factory equalization settings. In general, setting the control to lower numbers reduces on-air highs while moving it to higher numbers increases them.

Loudness (finally!)

The *Loudness* slider is used to set the desired on-air loudness. The most complex control within the entire Guru GUI structure, it intelligently communicates with three powerful sections of AirAura's processing: the 31-band limiter, the FM Limiter/Clipper and the MPX processing. Its intelligent hand on these other processing sections automatically optimizes the distortion vs. loudness tradeoffs as the *Loudness* slider is adjusted.



HD Sound Adjust

Density

The HD *Density* control adjusts how much short-term limiting is permitted within the multiband and broadband limiting sections. Higher settings will make the audio more dense and "processed" sounding while lower settings will be less so. Stations looking for a "compressed" sound on their HD signal will find both the *Compression* and *Density* controls quite useful for tailoring the "feel" of their audio. Note that increasing the Density setting will not cause the lookahead limiter to be overdriven because Guru's GUI intelligence will make other adjustments to prevent it. Distortion and/or audible pumping will not increase simply because Density was increased.

Low/Warm EQ

The *Low/Warm EQ* control is the one with the red ring around it and it adjusts how powerful low frequencies sound on the HD signal. When the control is at "0" and has not been touched since a preset was last taken, the factory assigned equalization for that preset is being applied. Adjusting the control either side of "0" increases or decreases the factory equalization settings. In general, setting the control to lower numbers reduces on-air bass while moving it to higher numbers increases it.

Mid/High EQ

The *Mid/High EQ* control is the one with the blue ring around it and it adjusts how powerful the higher frequencies sound on the HD signal. When the control is at "0" and has not been touched since a preset was last taken the factory assigned equalization for that preset is being applied. Adjusting the control either side of "0" increases or decreases the factory equalization settings. In general, setting the control to lower numbers reduces on-air highs while moving it to higher numbers increases them.

Loudness

Once the previous controls have been adjusted for the desired sound texture, the *Loudness* control can be used to set the desired HD loudness. The Loudness slider intelligently communicates with the HD processing's 31-band limiter and its lookahead limiter.

As the *Loudness* control is adjusted upwards and away from "5" the sound will get denser and louder. The opposite occurs when the control is adjusted counterclockwise away from "5."

Note that the Loudness control's interaction with other processing sections has been tuned to avoid undesired artifacts such as audible pumping and/or distortion when the control is set to either extreme.

Saving Presets

The *Presets* screen is equipped with a QWERTY style virtual keyboard for entering preset names when they are saved. For entering capital letters there is a Shift key to either side of the space bar which toggles the state of the Caps key – press once to lock the keyboard in Caps, once again to lock it in lower case. The visual feedback for which mode the keyboard is in is indicated by whether lower or upper case letters are visible on the keys of the keyboard.

When the *Save* button is pressed a dialog opens above the keyboard allowing a preset name to be entered (see below).

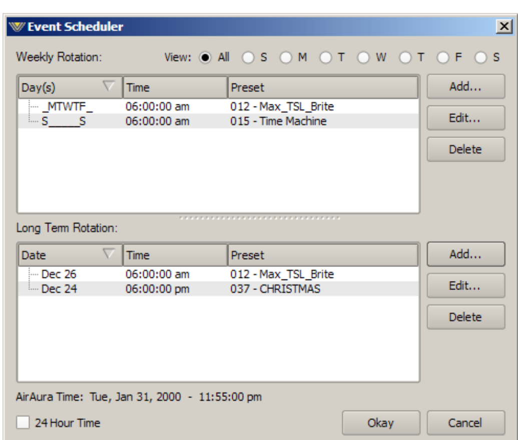
Once the preset name has been entered satisfactorily, press the *Save* button in the upper right corner to commit the new preset to AirAura's onboard memory, noting that it will be stored in the next available preset slot.

Preset Scheduler / Events

Both short term (daily/weekly) and long term (future) preset schedules are available within AirAura's preset scheduler.

The short term scheduler works much like other preset schedulers that you may be familiar with and is used to change presets according to a specific day of the week and time of day. Once a preset has been changed by the scheduler that processing preset will remain active unless and until another entry is programmed to change that preset to something else.

The long term scheduler is a Vorsis-exclusive feature and is useful when a processing preset is not part of the usual daily rotation. As an example, the scheduler could be programmed to switch to a specially prepared holiday preset at a specific time at the beginning of a holiday period.



The long term scheduler will keep a selected preset active *until* an entry in *either* the short or long form scheduler changes it to something else. It is not necessary (or possible!) to schedule an entry in the long form scheduler to cause an exit from the long form scheduler.

The screen on the left illustrates an example of short and long form preset schedules.

Virtually any combination of schedules and presets can be accommodated, noting that AirAura's GPI ports *also* allow preset switching via hardware connections to remote devices such as transmitter site remote controls, etc. The GPI's are completely separate from the scheduler and please take note: ***they override it***.

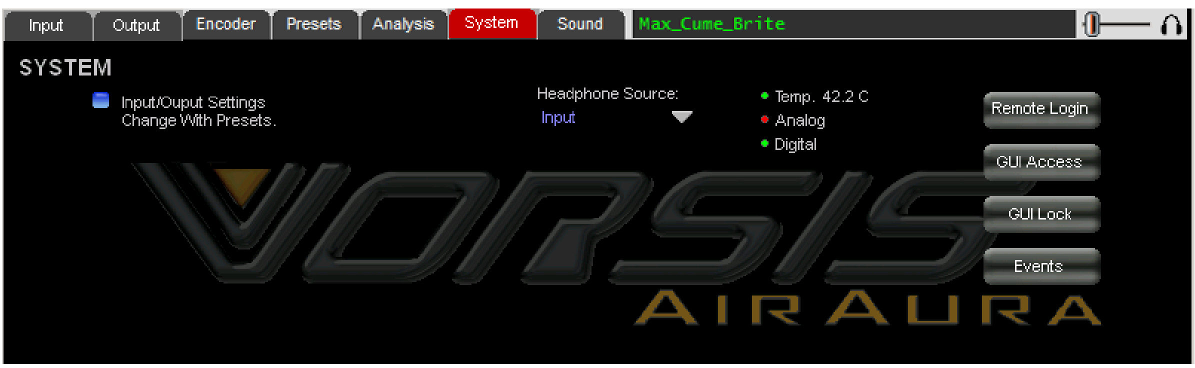
AirAura is also equipped with our exclusive ACI (Automation Control Interface) allowing automation-system based control of not just processing presets, but also modifying ANY processing or system parameter.

Please contact Wheatstone for details about our ACI protocol and how to use it with AirAura or any Vorsis processor.

Front Panel GUI Lock and Security

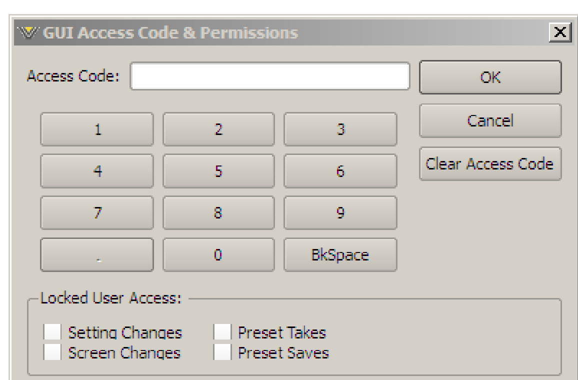
AirAura's front panel is equipped with a security feature to keep curious fingers at bay. As shipped from the factory the front panel security is not enabled. The user may assign multiple passcodes to the front panel to grant different levels of access to separate individuals if desired.

Because the system allows access to be granted at different levels of "permission" these permissions are granted by access *code*, not by user *type*. In fact, user types are created *through* the generation of specific access codes intended to prevent access depending on the passcode assigned.



On the System screen of the front panel are two buttons associated with front panel security; *GUI Access* and *GUI Lock*. In order to lock the front panel a GUI Access code must be assigned first. Attempting to lock the front panel without first assigning an access code will result in an error message to that effect.

The GUI Access button opens a dialog that allows the user to select what permissions a particular passcode may have. Every item under the Locked User Access dialog whose box is checked will be *disallowed* when the GUI has been locked and no passcode has been entered. Below are two examples of how the security feature might be used to customize local access to the front panel.



Example #1:

Suppose *Access Code* "1234" is entered and the only box checked is *Preset Takes*. This means that the person who is assigned Access Code "1234" can do anything but take Presets. Everything else is unlocked.

Example #2:

Suppose *Access Code* “5678” is entered and the boxes for both *Preset Takes* and *Setting Changes* are checked. This Access Code then allows the person with that code to do anything but take presets and change settings.

We suggest that the designated AirAura Admin create a personal passcode that has all four option boxes checked.

When security has been enabled by configuring the *GUI Access Code & Permissions* Dialog, selecting *GUI Lock* will immediately lock the front panel, prohibiting access until it has been unlocked with the proper access code.

If the front panel has been locked and a passcode has been assigned only to the *Preset Takes* option, a person could walk up and access the front panel and do anything except take presets. If they attempt to take a preset a security code dialog box will appear asking for the correct passcode for that locked feature. Unless the proper code for that is entered, access to that feature will be blocked.

PC Based
Audio Processing Guru® GUI

Chapter Contents

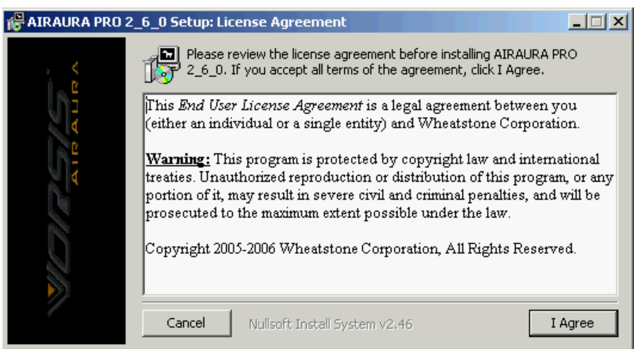
PC Based
Audio Processing Guru® GUI

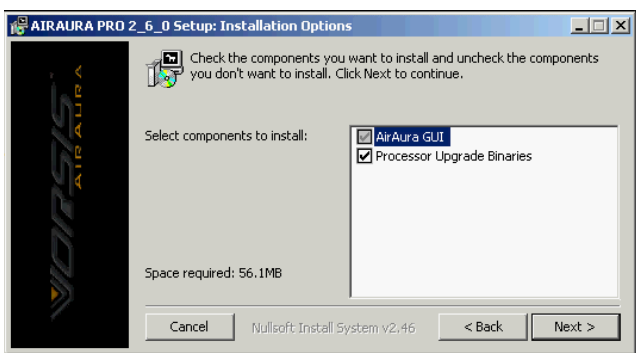
Installation

The Audio Processing Guru® software may be installed on any Windows-based PC running Windows 2000 Service Pack 2 or later. Installation is virtually identical across the various Windows operating systems and in general is as follows:

- Place the CD containing the AirAura software in the PC's CD drive and close the tray.
- Navigate to the CD drive using either Windows Explorer or through My Computer.
- Locate the file on the CD called "AirAuraGuru_x_x_x.exe where x_x_x denotes the software version number. Then…

Double click on this file and the installer will start, followed by the welcome screen shown below.





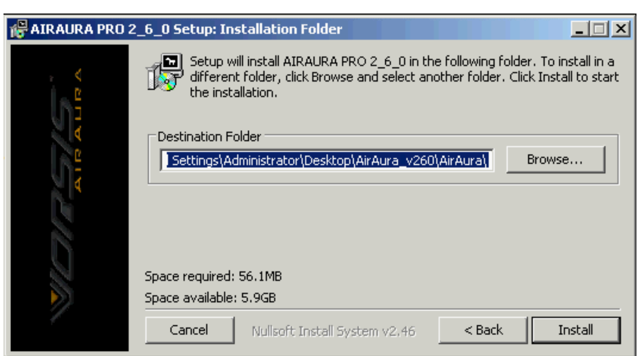
Click on "I Agree" and the installer will present a new prompt asking for what files to install. Leave all check marks selected and press the *Next* button to proceed.

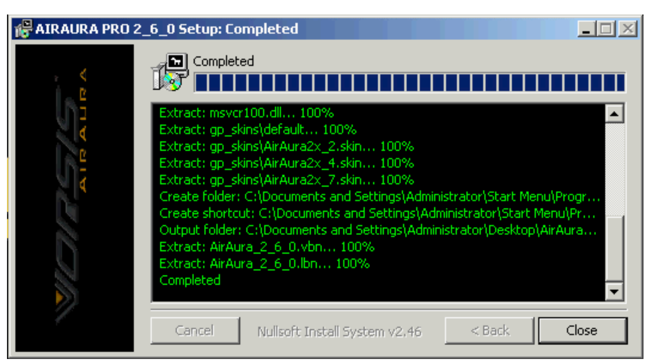
Next the installer will ask where to install the GUI.

The default is: *C:\Program Files\Vorsis\AirAura*, though our example at left has it being installed into a folder on the desktop.

With the exception of Windows® Vista 7 the GUI installer does not modify the registry and does not require external drivers. Therefore it can be run from any folder, including directly from a USB thumb drive!

When the installer has finished placing the files in the chosen locations a "Completed" status will be shown. Click the *Close* button to close the installer.





The GUI may now be started by selecting it from within the PC's Start Menu Programs list. Once the GUI starts it will appear as below, *minus* the metering and analysis activity shown below because we haven't yet connected the GUI to AirAura or even told it how to connect!

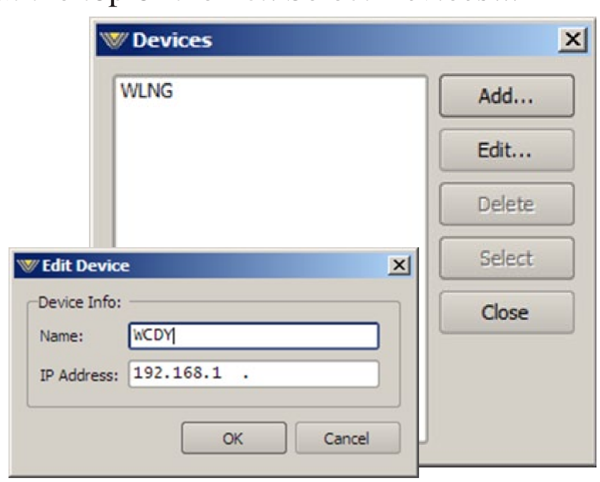
In order to communicate with AirAura the following conditions must be met.

1. The PC has an Ethernet interface and a cable is connected between it and AirAura (a crossover cable is NOT required).
2. The PC's network interface is enabled and is configured to be on the same IP address subnet as AirAura. The default IP addresses for AirAura are 192.168.1.200 for hardware and 192.168.1.201 for the front panel. These addresses cannot be used for the PC hosting the remote GUI.
3. AirAura is already powered up and the front panel displays are finished initializing and the screens are active.

Once all of the above is in place, the GUI must be configured to talk to the AirAura.

1. Place the mouse cursor in the upper part of the screen (where the controls are) and right click. A popup menu appears. Mouse over the word *Hardware* on this menu and another popup appears with the item *Devices*... at the top of the list. Select *Devices*... and the *Devices* dialog appears.

   In the event no AirAura units are listed, or one is adding a further device, or adding one for the first time, *Add* brings up the *Edit Device* dialog box, which asks for the new prospect's name and IP address.



2. In the *Edit Device* dialog we can give each AirAura a unique name. We suggest naming each processor, perhaps using the station's call sign.
3. Enter 192.168.1.200 in the IP Address field and click *OK*. The *Edit Device* dialog will close. With the new device in the list highlighted, click on the *Select* button of the *Device* dialog.

Once the *Edit Device* dialog closes, when the GUI's *Online* button is clicked (to the right of the *Sound* tab) the status indicator should change to say Online and the AirAura's name should appear inside the Device window underneath the Online status.

If the AirAura will be staying on its default IP address no other steps need to be taken. If the IP address configuration needs to change to match the local network's environment, please see the next section for how to accomplish those changes.

Changing AirAura's IP Address Configuration

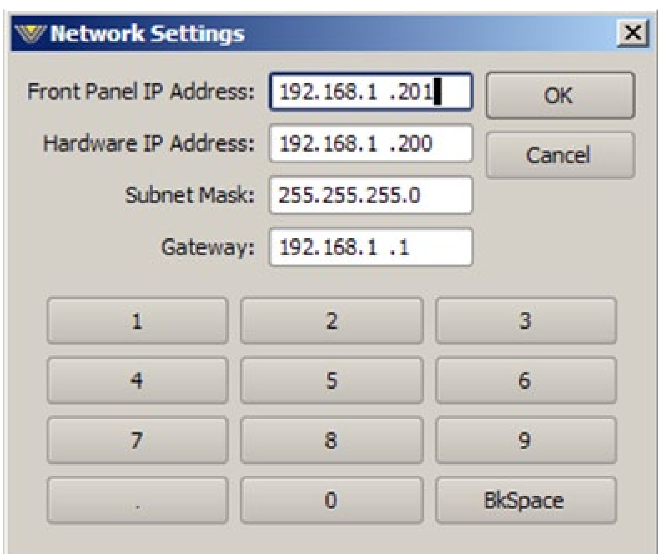
AirAura's pre-configured factory default Ethernet and Wireless Configuration is as follows:

Front Panel IP Address:	192.168.1.201
Hardware IP Address:	192.168.1.200
Subnet Mask:	255.255.255.0
Gateway:	192.168.1.1*
Wireless Address:	192.168.2.100**
WLAN Channel:	1
Country:	United States

* Using a Gateway address on the wired side of the WLAN module can cause a local connected network to become confused if it and the gateway addresses of the module are the same. It is sometimes best to program the WLAN module's Gateway address to an unused address on the locally connected network. If an invalid entry is used for the WLAN module Gateway that address can no longer be used for the module's web-based configuration and the wireless address will need to be used instead.

** in accordance with NAT routing rules the *wireless* IP address and AirAura's wired hardware IP addresses must be on a different subnet!

AirAura's network configuration can be viewed or changed by using the front panel's right hand pushbutton to bring up the *Network Settings* configuration where the network settings are located. The IP address setup page appears as shown below:



On this screen there are four entries (default settings shown):

Front Panel IP Address – This is the reserved address that the front panel (not the remote GUI on the PC!) uses to communicate with AirAura's DSP farm.

This address is set at the factory to be one IP Address *higher* than the IP address assigned to AirAura's *Hardware IP Address*, but it doesn't have to be set that way as long as the front panel is configured to use a valid address on the local network and that address is not in use by another device.

NOTE: We believe it's a good idea to avoid using an IP address that ends on ".1" as these are sometimes reserved for network gateways.

Hardware IP Address – This is the address of AirAura's DSP hardware and is the IP address that a remote GUI will use to connect to and control AirAura.

Subnet Mask – The subnet mask entry must be appropriate for the network being used. Please see your local IT person if this item causes confusion, or consult one of the free online resources such as http://www.subnetmask.info/ or http://www.subnet-calculator.com/.

Gateway – This is the gateway address that AirAura will use to reach connections <u>*outside*</u> of its own internal network. If the controlling PC and AirAura are connected to the same network subnet a Gateway address *usually* isn't necessary and it is *usually* safe to set the *Gateway* address to either 0.0.0.0 or 255.255.255.255. Note that on the PC side of things, Gateway behavior when the address is set to all zeros or all 255's can be "Windows-version dependent."

After determining the IP address that you want AirAura to use, enter that address in the *Hardware IP Address* field using the front panel navigation buttons. It is always a good idea to clear the entire IP address field first before entering new data to ensure there will be no leading or training blanks in the address that may be hard to see. Always use the decimal point button to navigate to the next position of the network address field.

Next, using the same procedure as was used to enter the Hardware IP Address, enter a new front panel GUI IP address by adding "1" to the IP address that you entered for the hardware in the above step (see example below). We typically use one address above the hardware address as our factory default for the front panel but this is not a requirement. The only "rule" is that the front panel, the hardware address, and all other IP addresses on a connected network are on their own unique addresses and on the same network subnet.

Example:

If you entered 192.168.0.200 as the hardware address, then the logically correct front panel GUI address will be 192.168.0.201. Note that the front panel GUI address is not utilized externally by AirAura; however no other device on the connected network should be using this address because it is visible to the connected network. In other words this address must be reserved in your IP address plan even though it is not externally being used.

Next, enter the Subnet Mask and Gateway using the same procedure used for the IP addresses. Then click the *OK* button. When prompted to reboot AirAura for the changes to take effect, wait about 15 seconds to ensure that the settings have been saved to onboard flash memory, click *OK* and then interrupt power to the unit for about 30 seconds. Reapply power, and once the AirAura has rebooted and comes back to life, the new IP addresses should be valid.

How to Verify The New IP Addresses are Working

The most revealing way to see if the address change was successful is to simply check the front panel displays. If audio is applied to the unit and the Input source is set for the type that is actually in use, the input and output level metering should be showing activity and the analysis display should be active.

If no audio is connected to the unit yet, connectivity to the front panel can still be verified by putting the Analysis display in the "3-D Plot" mode to see if a purple "floor" is being painted on the display at about a 45 degree angle from the lower left to upper right of the upper half of the right hand screen. If so, this verifies that the front panel and hardware are communicating and that valid data is being exchanged.

Reconfiguring The Remote GUI

If the IP addresses of AirAura have been changed from those configured in the remote GUI a few pages ago it will be necessary to reconfigure the remote GUI software running on the user's Windows PC to communicate on AirAura's new IP address. Here is how to do that…

Right click in the controls area of the remote GUI to activate the drop down menu and select the *Devices* option under *Hardware*. When the *Edit Device* dialog opens highlight the AirAura device that was created earlier and click on *Edit*. Then modify the IP address so that it perfectly matches the new hardware IP address just given to the AirAura hardware and then click *OK*.

Assuming that a good network path exists between the host PC (don't forget to change its address too if the IP address subnet has changed!) the GUI should be able to go online to AirAura by clicking on the *Online* button at the top of the GUI. When the status indicates Online the GUI is communicating with AirAura.

Connecting Directly Without a LAN

You can work without a LAN by connecting AirAura and the PC Ethernet ports together with a standard Ethernet cable – either a straight through or crossover cable may be used – AirAura's Ethernet interface ports are Auto MDIX.

NOTE: The controlling PC and AirAura'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 also be on the same network numbering scheme – although the network subnet can be any one that is valid, both local and remote devices must be on the same subnet in order to communicate.

Example:

If AirAura is using the 192.168.1.x address range and a laptop is using the 172.68.122.x address range the laptop will not be able to communicate with AirAura. The laptop will need an IP address configuration that puts it on the same 192.168.1.x network subnet as AirAura.

Wireless Communications to AirAura

Many like to make adjustments to AirAura while listening in a convenient remote location such as a familiar car in the station's parking lot or outside the transmitter building. This is easily accomplished by using AirAura's built-in high-power industrial grade 802.11b/g wireless capability.

Once it is powered up AirAura should appear as AIRAURA among the wireless networks that the PC can detect and its status will show as "On Demand." Clicking on the AIRAURA wireless device and asking to connect will bring up a Windows warning box stating that AirAura is an "Unsecured Network"*** and asking if you wish to connect anyway. Say *Yes* to this prompt.

*** To ease "first time" connectivity in the field we've set AirAura's wireless interface security to disabled and made it an Ad Hoc network. Once connectivity has been established the wireless interface's security can be reconfigured later as desired by referring to the procedures in Appendix B.

Connecting wirelessly to the AirAura is easy:

1. Open the PC's wireless connections dialog – AirAura should be listed as an "On Demand" device.
2. Highlight AirAura and click on the Connect button at the bottom of the dialog.
3. A "Connect to unsecured network?" prompt may appear. Click *Yes*.
4. The "Acquiring Network Address" message should appear next to the AirAura.
5. Once the status shows "Connected" (this may take a while depending on how Windows is configured to connect to wireless devices) open the PC GUI and click on the *Devices* button.
6. Right-click on the upper part of the GUI and select the Hardware/Devices option.
7. Click the *Add* button and the *Add Device* dialog will open.
8. Enter a new device name for AirAura (such as AirAura Wireless).
9. Enter the IP address **192.168.2.100**, the WLAN factory default, and click *OK*.
10. Highlight the new device that was just created and click the *Select* button.
11. Click on the GUI's *Online* button.
12. The AirAura GUI should connect to the AirAura and control and graphical display activity should commence.

AirAura/Internet Security

AirAura's wireless interface has a "web server" for configuration only. The wireless interface cannot reach any other IP addresses on the wired side of the network because the NAT router within the wireless module *only* routes between its *single* fixed IP address and the *single* fixed address on the wired side which is AirAura's hardware address.

About DHCP and AirAura

AirAura does not utilize nor does it support Dynamic Host Configuration Protocol (DHCP, or automatically assigned network addressing). AirAura needs to be assigned a "STATIC" and unique IP address.

AirAura's Ethernet Ports and Network Protocols

AirAura uses both TCP and UDP protocols to communicate with the remote GUI. The TCP protocol is used between the GUI to AirAura for control functions. TCP's high reliability (due to handshaking in the protocol) ensures that control changes sent to AirAura by the GUI will be received *without* any errors.

Metering data isn't critical, and further, only the *differences* between the last and next meter values are sent. Therefore a "no-handshaking" protocol, UDP, is all we need for sending metering and other analysis messages data back to the GUI. Dropped or corrupted packets have little visual effect on the metering and analysis displays, and the lack of (or no need for) handshaking *drastically* reduces the amount of network bandwidth required in order to run the GUI's remote meters and analysis displays, especially over the Internet or on weak or noisy wireless connections.

AirAura uses the **TCP** protocol for sending of commands, adjusting controls, taking presets, etc. We utilize TCP because it guarantees perfect data integrity under all connectivity conditions. AirAura utilizes a fixed port, **55899** for TCP control. This port is hard coded and cannot be changed in the field.

AirAura uses **UDP** ports in the range of **60000** to **60010** for sending metering and analysis data back to the remote GUI from the hardware. Whenever the GUI connects, it tries to open UDP port 60000 first, and if that port isn't busy, it uses it. If port 60000 is busy, port 60001 is tried next. This process repeats until either an idle UDP port in the 60000 to 60010 range is found, or the search stops. This port range cannot be changed in the field.

NOTE: The above referenced TCP and UDP ports must be open in order to support connectivity through routers and firewalls. This also includes the Windows firewall on the host PC that is running the GUI software – it must have these ports open too.

If the GUI Shows That It Is "Connected" But the Meters and Analysis Displays Aren't Active

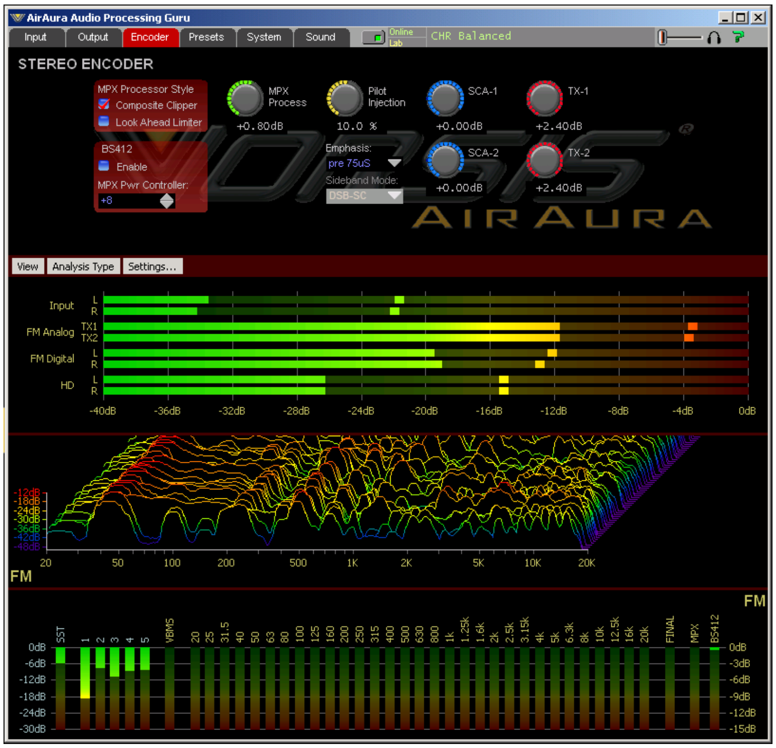
Occasionally a network or internet service provider will block what it considers to be unsolicited UDP traffic. When that happens the GUI will connect to AirAura and AirAura will be able to be controlled, but no metering or analysis displays will be active. Fortunately there is an easy workaround:

Within the *System* screen of the GUI is a checkbox labeled *Metering data over TCP/IP*. When this box is checked the GUI tells the hardware to use the TCP protocol instead of UDP for sending meter data. While using the TCP protocol can be visually slower on some networks (because now every packet must be checked for accuracy) the "no metering" issue should be solved.

Choosing the TCP protocol for metering naturally increases the amount of network traffic because every metering and analysis packet requires a handshake to ensure its integrity. The poorer the network, the busier the GUI will be while it handshakes on every data packet because a poor network implies the need for a lot more handshaking and retries in order to ensure valid data. Naturally, the worse the network is, the more handshaking is required to get good data.

Using Guru GUI

The Guru GUI may be moved around on the screen of the host computer by left-clicking and holding its title bar and dragging it to the new position. The GUI may also be minimized on the desktop (taken off the screen) and/or closed by way of the familiar Windows controls at the extreme top right. Unlike the Professional GUI (covered later) the Guru GUI has a built in Analysis window and may be re-sized on the desktop as desired by dragging its borders.



Descriptions of the features and functions of the GUI will follow the following logical path:

- Control Area at the top of the GUI (pages 3-11 - 3-26);
- Input/Output Level displays (page 3-27);
- Gain Reduction displays (page 3-28);
- Dynamic Analysis displays (pages 3-31 - 3-33).

The Guru GUI application that runs on the host Windows PC and the GUI that runs on the front panel of AirAura are virtually identical except for the sizing and positioning of the graphical analysis and metering displays. Therefore the following explanations for the Guru GUI software also pertain to the GUI on AirAura's front panel and vice versa.

NOTE: ***The following discussion describes AirAuras running software Version 2.4 or higher. When AirAura is running software prior to Version 2.4 the WheatNet-IP Input Source is not present in the Input screen; only AES and Analog are available.***

Input Menu

This is the first tab on the top far left of the GUI, noting that the screen that is currently being displayed is indicated by a red background on the selected tab.

The *Input Configuration* screen shown above contains controls for the Input Audio *Source*, *Automatic Input Failover*, *Analog Gain* and *Digital Gain,* and *L/R Balance* controls. It also shows the input source controls that are available when AirAura is running software Version 2.4 or higher (equipped with WheatNet-IP). All Input screen controls can be adjusted using AirAura front panel track pad.

Input Source

The Input Source can be selected as either *Analog*, which is the analog Left/Right input, or *Digital*, which can be **either** *AES3* **or** *WheatNet-IP*. The currently selected input choice is shown by a red check mark inside the blue choice box.

Audio Input Signal Presence Indicators

Below the source selection check boxes are a pair of signal presence indicators for each input. The *Analog* indicator lights if there is a signal present at the analog input above -48dBFS. The *Digital* indicator lights if there is a valid AES signal present at the digital input above -48dBFS.

Automatic Input Failover

In the event the currently selected input source fails *and* the *Automatic Input Failover* check box is checked, an alternate source of audio connected to the other input can automatically be put on the air.

If the primary source was Digital the Analog input will be selected immediately if invalid bits occur in the digital bitstream or after 30 seconds with an audio signal level below -48dBFS.

If the primary source was Analog, then a timeout of 30 seconds must elapse before the unit switches to the AES/EBU Digital input in an attempt to restore audio. The audio failure sense threshold is fixed at -48dBFS, and this in combination with the 30 second timeout, is suitable for virtually all program types.

Automatic Failover is between Analog and Digital sources only – failover between digital input sources (AES3 and WheatNet-IP) is not supported.

Analog Gain

The Analog audio input gain may be adjusted by clicking on the *Analog Gain* knob and dragging the mouse or touchpad to the right or left. The Analog Gain can be adjusted over a +/- 24dB range.

Digital Gain

The AES/EBU Digital audio input gain can be adjusted by clicking on the *Digital Gain* knob and dragging the mouse or touchpad to the right or left. The AES Gain can be adjusted over a +/- 24dB range.

L/R Balance

Static level errors in Left/Right channel balance can be corrected by using the *L/R Balance* knob and may be adjusted over a +/- 12dB range. The *L/R Balance* control is common to both the analog and AES/EBU digital inputs.

Output Menu

This is the second tab from the left on the top of the GUI. (Note: the screen that is currently being displayed is indicated by a red background on the selection tab).

The *Output Configuration* screen contains controls for selecting the *Analog Output Type*, setting the FM *Diversity Delay* time, and for setting the *FM Digital Output* and *HD Outputs* levels.

For best resolution of the Diversity Delay controls they are best adjusted using the host PC's keyboard and mouse and using the following table as a guide.

COARSE ADJUST SLIDER		FINE ADJUST SLIDER	
Slider only	500 mS	Slider only	0.9 mS
Slider plus Shift key	1700 mS	Slider plus Shift key	8.6 mS
Slider plus Ctrl key	50 mS	Slider plus Ctrl key	0.1 mS (100μSec)

Note that the smallest increment of delay time adjustment is 100 *microseconds*.

The FM Digital Output offers options for applying De-Emphasis that is complimentary to Pre-emphasis that may be in use and also allows the digital output to be taken before or after the FM Diversity Delay.

The HD Output has options for reversing the phase of the left and right signals as well as for swapping the left and right channels.

NOTE: The Analog Output Type box needs special mention!

Unlike other audio processors AirAura's balanced analog outputs can also carry balanced composite stereo. When the MPX check box is selected, the rear panel TX-1 and TX-2 BNC connectors are carrying the standard unbalanced composite stereo signal – at the same time the XLR outputs are carrying balanced, line level composite stereo. This feature has distinct advantages when the ultimate signal to noise ratio is desired and the exciter being fed has a differential input stage (most, if not all modern exciters do).

When the L/R Discrete or L/R De-Emphasis check boxes are checked an FM Analog Output level control will appear in this screen and the TX level controls in the Encoder screen will vanish.

Stereo Encoder Menu

AirAura contains a mathematically perfect stereo encoder whose performance is limited only by the high speed output Digital to Analog converter (DAC) and its reconstruction filter. Stereo separation is typically >50dB between 50Hz and 15kHz. Clicking on the *Encoder* tab opens the *Stereo Encoder* screen hosting controls for this section.

MPX Processor Style

AirAura provides two methods to process the composite stereo waveform. These two methods are provided to allow the user to select what is most appropriate for his format and market.

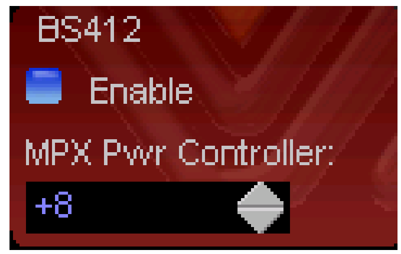
Composite Clipper – Emulates analog composite clippers but uses a very high sample rate and special algorithms to drastically reduce the artifacts caused by analog composite clippers.

Look Ahead Limiter – This is a true lookahead limiter which utilizes a 0.5 millisecond look ahead and extremely high sample rates to precisely control composite waveform peaks on a cycle by cycle basis. It works similarly to the oversampled lookahead peak limiters found in the FM and HD Limiter screens, but because of the much higher bandwidth required by the composite signal, it operates at a much higher sample rate.

MPX Process Control

The amount of MPX processing being applied to the composite signal is controlled by the MPX Process Drive control. This control is calibrated in 0.05dB steps and allows the amount of MPX processing to be precisely set. Unlike other processors the control may be set to values below 0dB for test purposes. Note also that unlike conventional composite processors the amount of multiplex signal processing can be set quite aggressively without encountering the usual artifacts.

BS412 MPX Power Controller



In certain European countries the control of maximum Multiplex Power is required by broadcast regulatory agencies. The regulation governing this is commonly referred to as ITU-R BS.412-7, and the primary aim of controlling maximum MPX power is to reduce adjacent channel interference by lowering the average modulation level.

AirAura is equipped with a sophisticated MPX Power Controller which enables it to operate within these regulatory guidelines. On the left side of the *Stereo Encoder* screen are the two controls associated with its BS412 MPX Power Controller.

Note that the *MPX Power Controller* is always measuring multiplex power, even if the *Enable* checkbox is not checked. Therefore if the MPX Power Controller setting is set to any value except off, when the *Enable* checkbox is checked MPX Power will immediately fall to the value set by the *MPX Power Controller* control.

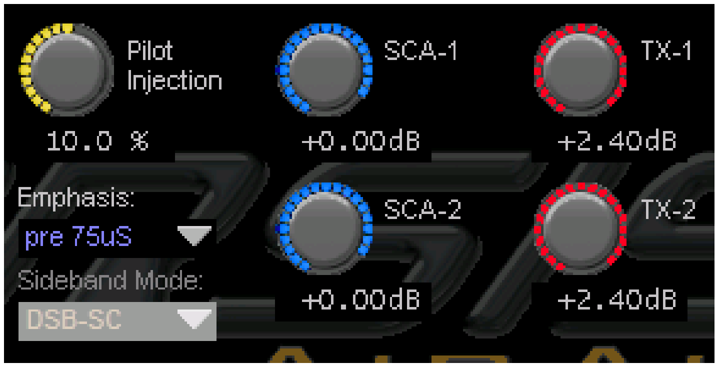
Enable – When the *Enable* checkbox is checked the BS412 algorithm is controlling the multiplex power. Likewise, when the checkbox is not checked, multiplex power is not being controlled, regardless of the setting of the *MPX Power Controller* control. As mentioned previously, when the *Enable* checkbox is checked the *MPX Power Controller* will immediately set the processing output's MPX Power to the level set by the MPX Power Controller's value.

MPX Power Controller – This control sets the maximum permitted multiplex power according to the BS.412 measurement algorithm. The control has 15 possible settings, plus an OFF position: +8db, +7dB, +6db, +5dB, +4dB, +3dB, +2dB, +1.5dB, +1.0dB, +0.5dB, 0, -0.5dB, -1.0dB, -1.5dB, and -2.0dB.

There is a 0.5dB resolution available around the control's zero setting to allow small adjustments in actual MPX Power to compensate for unusual processing settings and how they affect MPX Power.

IMPORTANT!

The MPX Power Controller's sole purpose is to reduce loudness and program density. If you are not required to use the BS412 Controller it should be left in the OFF position because turning it on can cause up to a 5dB loudness loss when the control is set to its "0dB" position. If your station is not required to comply with a Multiplex Power standard where it is licensed to operate the BS412 control should remain in its OFF position!



19kHz Stereo Pilot Injection

The *Pilot Injection* control adjusts the 19kHz stereo pilot level 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.

Emphasis

This control sets the desired pre-emphasis curve to apply, 50 or 75 microseconds, depending on the part of the world where AirAura is being used. The pre-emphasis may also be switched off (FLAT) for testing or other purposes.

Sideband Mode

AirAura is equipped with the ability to operate the multiplex generator in the standard double, or experimental single sideband mode. As of this writing the DSB/SSB control is deactivated pending the implementation of our new peak control method optimized for SSB use.

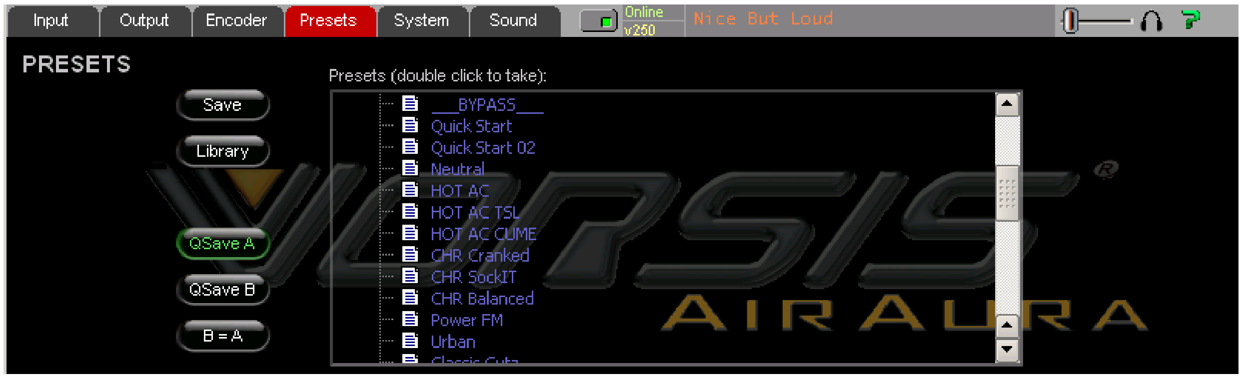
SCA-1 and SCA-2 Inputs

Input signals such as SCA and RDS applied to the SCA connectors are digitized at 192kHz. Prior to being digitized the signals are high pass filtered to reduce hum and then low-pass filtered at 94kHz. This control can be set to OFF or to levels from -79.95dB to +10.0dB in 0.05dB steps.

TX-1 and TX-2 Outputs

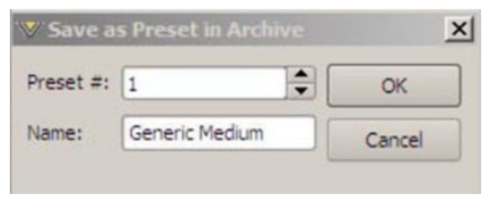
These controls adjust the output amplitude of the signals appearing at the rear panel BNC *and* XLR connectors when the Output Mode is set to MPX in the *Output Configuration* screen. These controls can be set from -79.95dB to +6.00dB in 0.05dB steps. Note that the audio levels at the balanced XLR connectors are always 6dB higher than the levels at the unbalanced BNC connectors.

Presets Menu



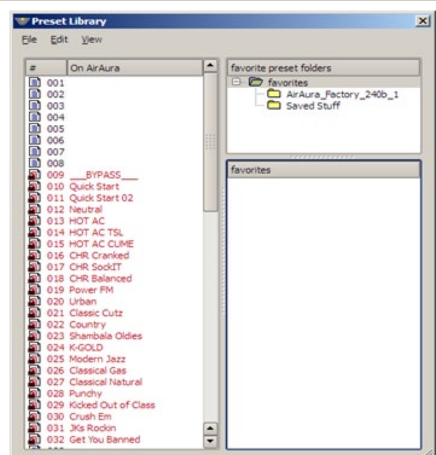
The *Presets* tab opens a screen where presets may be selected for on air, saved to the processor, or managed through AirAura's comprehensive Preset Library Management utility.

Selecting a Preset



Selecting a preset for on-air use is as simple as double-clicking on a preset in the list. The currently running preset will morph into the new preset over a period of several seconds in order to prevent abrupt clicks and pops when the new preset's parameters are being loaded.

Saving a Preset



When the *Save* button is pressed a dialog opens allowing a preset name to be entered.

Once the entered preset name is satisfactory, press the *Save* button to commit it to memory, noting that unless chosen otherwise it will be stored in AirAura's next available open preset slot.

Preset Library Functions

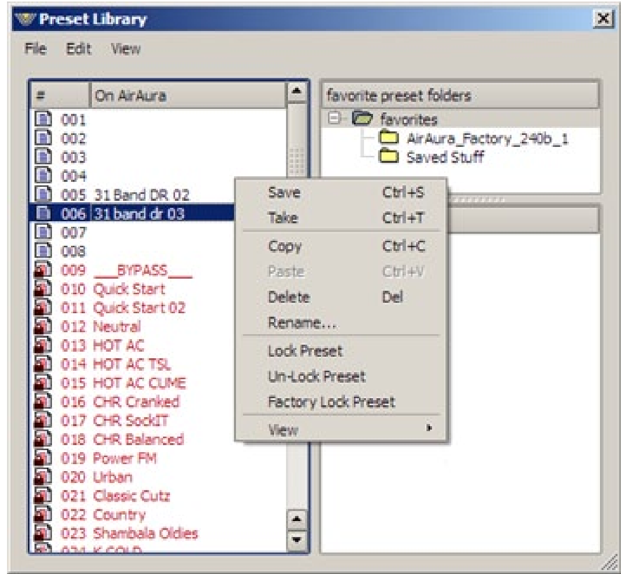
Clicking on the *Library* button opens up the preset library management feature. When this window is open both the presets on the AirAura hardware itself (on the left side of the window) and on the host PC (right side of the window) are visible.

To move presets from the hardware to the PC or vice versa, simply drag the preset in the direction you wish it to go. Whenever presets are being moved from AirAura to the PC, a copy of the preset being dragged over will remain in the Library window.

The same goes for the behavior when dragging a preset from the PC to the hardware – a *copy* of the preset is sent to the hardware – the original preset remains on the PC for safe-keeping.

Locking Presets

Presets stored on AirAura 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.

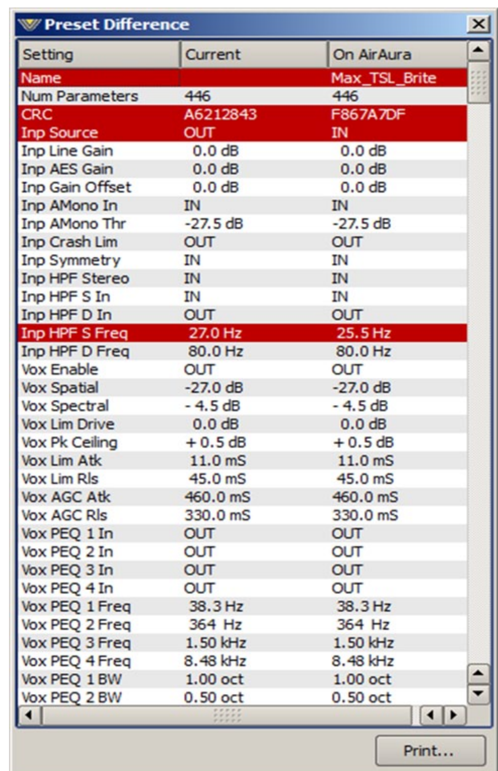


There are two ways to manage the lock status of user presets:

- 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 Changes

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

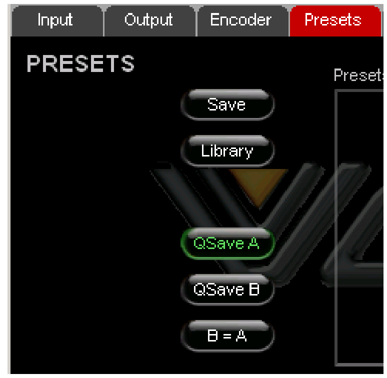


When the *Library* tab is opened, highlight any preset you'd like to compare with the current preset running on the processor. Then right click and select *View* and *Diff* from the sub-menu. You will see the window on the right open.

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.

Quick Save (QSave)

The AirAura has a unique feature that allows instant comparison of work in progress to a known reference, such as another preset, or some midpoint while adjustments are being made. We call this feature **QSave** (for **Quick Save**).



At the left side of the *Presets* menu you'll find three buttons like those shown on the right.

QSave A and *QSave B* represent two temporary memory buffers that hold all current AirAura processing settings as long as power is applied to the unit.

While *QSave A* is highlighted green any adjustments that you make to controls on the AirAura are being concurrently saved to temporary buffer *A*.

QSave B operates on buffer *B* and just like *QSave A* its being selected is shown by a green highlighting.

One way to use this feature is to compare the sound of a factory preset to changes you've made to it without having to first save it as a user preset. To do this:

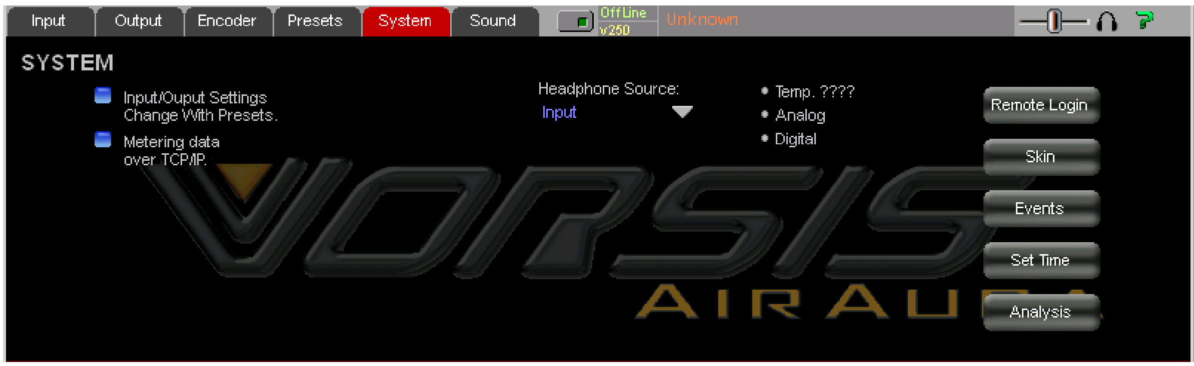
- Recall the factory or user preset that you wish to adjust.
- Ensure that *QSave A* is highlighted. If it is not, press its button to highlight it.
- Press the *B=A* button. This will copy the contents of QSave buffer *A* to QSave buffer *B*. Now the contents of both *A* and *B* buffers are identical.
- Change some settings on the AirAura and they will automatically be stored in the *A* buffer.
- Compare your changed settings to the recalled factory preset by pressing the *QSave B* button.
- Compare those settings back to the factory preset by pressing *QSave A*.
- When you are happy with your changes you can commit them to a user preset using the preset *Save* dialog explained previously.

Likewise, *QSave A* and *QSave B* can be used to compare the sounds of two *different* sets of *user* settings. To do this:

- Load the preset that you want to change, make changes to it, and press *QSave A* to save those settings to buffer *A*.
- Make some additional changes and then press *QSave B* to save those additional changes to buffer *B*.
- Now buffers *A* and *B* have your two different settings you can compare them by toggling back and forth between *QSave A* and *QSave B*.
- When you are pleased with one set of settings and need more buffers for further tweaking, you can use the *A=B/B=A* button to make the two buffers the same and have one of them to use to start comparing from again.

System Menu

The *System* screen allows system-wide settings to be configured. The following features and options are available within the *System* menu.



Input/Output Settings Change With Presets

Vorsis processors segregate "sound" settings like AGC Drive from "system" settings like Analog Output Level. However, unlike other processors you may be familiar with, when a Vorsis preset is saved <u>all currently active parameters are stored with that preset, including system settings</u>. This is an advantage when one wants to change one or more system settings *without* changing the sound, or change one or more system settings *and* change the sound.

This means that processing presets can be used to adjust system parameters whenever desired. An example of how this feature could be used is where a station is transmitting a mono sports broadcast and doesn't want to change the sound but does want the stereo pilot off while the broadcast is in mono.

When the check box labeled *Input/Output Settings Change With Preset* is checked, all of the system settings stored within a preset will be applied when that preset is taken. In the above example two presets saved as identical other than the stereo pilot setting could be used to turn the stereo pilot on and off without changing anything else. If these presets are stored in the first eight preset slots then GPI signals applied to the AirAura rear panel can select these presets with remote contact closures.

Metering Data via TCP/IP

As mentioned previously, sometimes UDP packets are blocked by an Internet provider or in house router, rendering the GUI without metering (because it uses the UDP protocol for that). In this situation metering can be restored simply by selecting the *Metering data over TCP/IP* check box.

The downside of operating the GUI in this mode is that, unlike UDP, which requires (and expects) no handshaking to guarantee data integrity, TCP/IP *requires* handshaking on every packet to ensure that each packet received is identical to the one transmitted. On a noisy or poor link, the need to handshake on every metering packet can slow the network down because of the additional handshaking and data transmissions required.

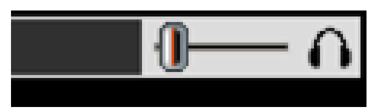
Headphone Source

The headphone source may be directed to several points inside the processing chain:

- **Input**: Output of the audio input selector, whether analog or AES input is selected.
- **Analog Input**: Audio appearing on the analog input even if not selected to be on air.
- **AES Input**: Audio appearing on the AES3 digital input even if not selected to be on air.
- **HD Output**: Output of the entire HD processing chain.
- **FM PreDelay**: Output of the FM processing chain prior to FM diversity delay. Note that if pre-emphasis is on, this signal will be "bright" because it is pre-emphasized.
- **FM Output**: Listens to the output of the entire FM processing chain, *including* the FM diversity delay. Note that if pre-emphasis is on, this signal will be bright because it is pre-emphasized.

NOTE: AirAura's headphone monitoring feature allows monitoring of an audio input *without* having to put it on the air. This is useful for confirming that backup audio is actually present and acceptable before putting it on the air!

Headphone Volume Control



On the top right corner of Guru GUI is the headphone volume control slider. To adjust the headphone volume slide the fader right to increase volume and left to decrease it.

Note: The headphone outputs of the AirAura are driven by National Semiconductor LM675 power operational amplifiers capable of >3A into any reasonable load. Actual headphone impedance is not a concern; however headphone circuit current limiting is provided at 160mA in order to protect external devices and also to keep you from setting your hair on fire from too high a headphone listening level!

Status Indicators

There are three status indicators: *Temp*, *Analog*, and *AES*.

Temp

This indicator is green when the GUI is online to AirAura and its internal temperature is below 50°C/122°F. The numbers displayed represent the current temperature of AirAura's CPU.

Analog

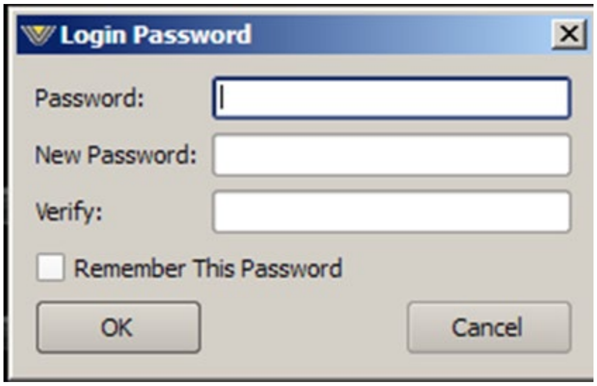
This indicator is green whenever there is analog audio present on *both* left *and* right channels and it is above -24dBu.

Digital

This indicator is green whenever there is a valid AES3 digital audio signal present and it is above -48dBFS.

Remote Login

When Remote Login is clicked, a *Login Password* prompt box will open as shown below:



A remote login password restricts GUI access to an AirAura processor over the Ethernet connection. When a remote GUI session is initiated, AirAura accepts the connection and asks the remote GUI for a password.

If the remote GUI can't provide the password, a dialog box pops up asking for the correct password. Until the login password can be verified AirAura will not send its current settings *to* the remote GUI nor accept any changes *from* the connecting GUI.

This dialog box allows you to change the login password. You need to know the current password beforehand (if one has already been assigned), and also enter the new password twice for verification. The password can be any combination of characters or numbers.

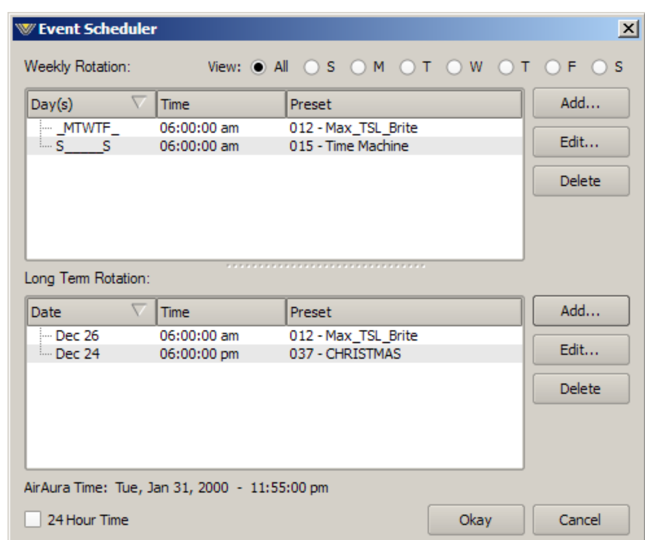
If the *Remember This Password* check box in the dialog is checked, the GUI stores the password on the local PC. If you do not check the check box you will be asked for the password *each time you connect remotely* (since the first round of verification will fail due to no locally stored password).

Occasionally you may see the word "pending" temporarily appear in the connection status box of the remote GUI while the AirAura and Remote GUI are handshaking on the password string being sent.

Events

The AirAura 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. To create, edit, or view entries in the *Event Scheduler* click the *Events* button on the System Screen.

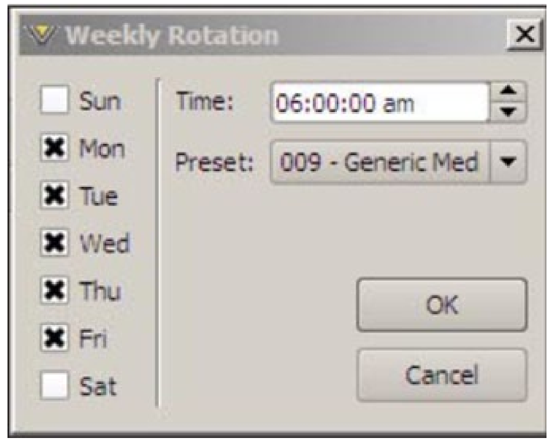
The *Event Scheduler* screen shown on the right will open and the events list will be empty when opened for the first time:



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 may be chosen.

3. Next select the time that the preset should change.

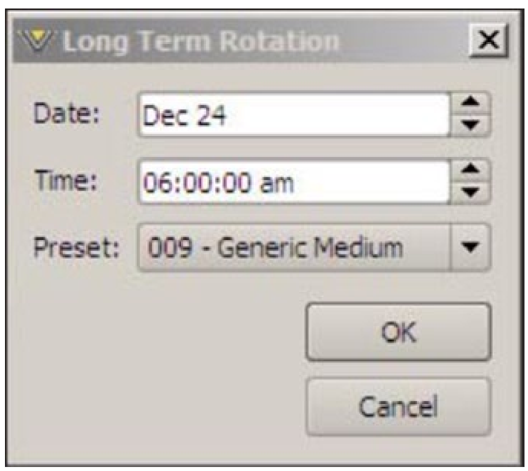
4. Next choose the preset that should be selected at the time programmed in the previous step. When finished editing, click *OK*.

As many preset change events as desired may be programmed in the *Event Scheduler*.

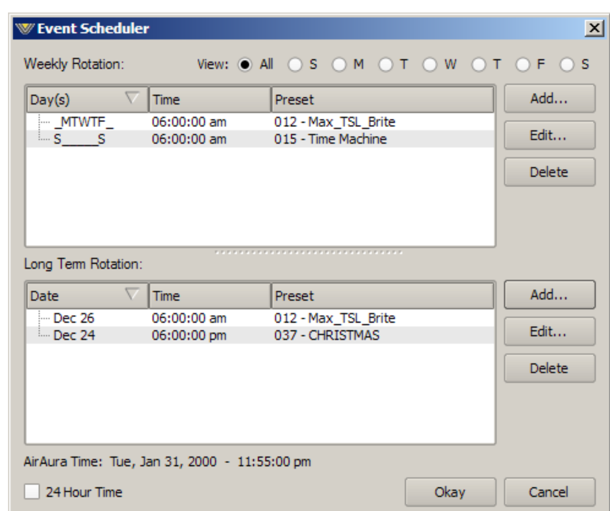
If a one-time preset change is required, such as might be necessary for a special holiday or other event, the *Long Term Rotation* routine should be used.

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 appears as follows:



Deleting a Scheduled Event

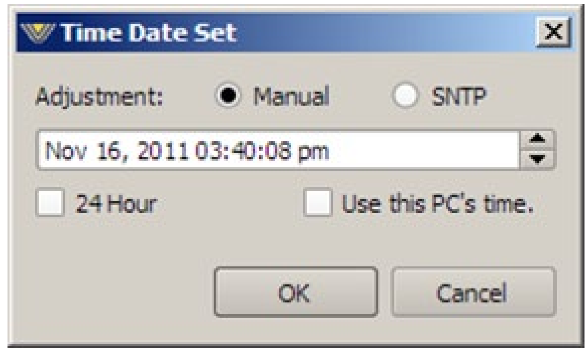


Deleting a scheduled preset change is as simple as adding one. Open the *Event Scheduler* window by navigating to the System menu and clicking *Events*. The *Event Scheduler* window will open.

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 that there is no confirmation prior to the actual deletion of the event!

Setting System Time



Allows the system time of AirAura's onboard computer to be set or changed. AirAura's system time can be synchronized to external time servers (SNTP, or Simple Network Time Protocol) if desired. 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.

SNTP (Simple Network Time Protocol)

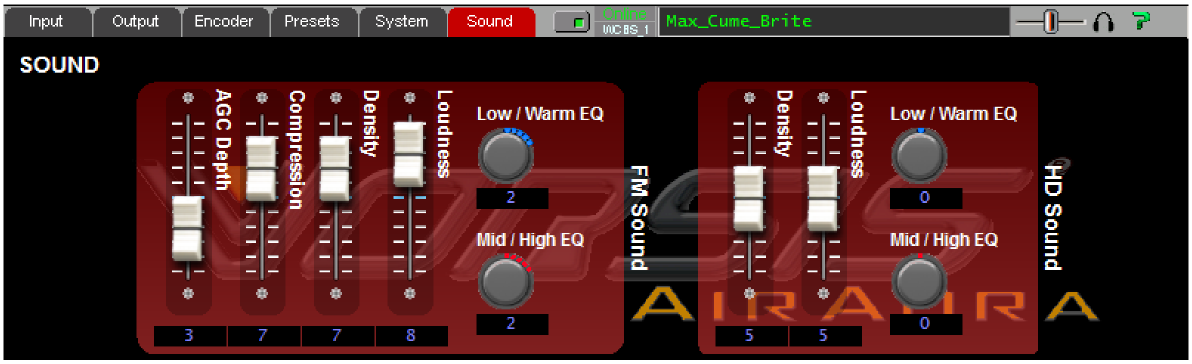
AirAura supports automatic and unattended system time updates via SNTP. Using SNTP can keep AirAura's onboard clock set to within milliseconds of actual time. Various "time servers" are available via an internet connection and a partial list of available time servers is below. All servers shown are located within the United States.

IP Address	Name	Location
129.6.15.28	time-a.nist.gov	Maryland
129.6.15.29	time-b.nist.gov	Maryland
132.163.4.101	time-a.timefreq.bldrdoc.gov	Colorado
132.163.4.102	time-b.timefreq.bldrdoc.gov	Colorado
132.163.4.103	time-c.timefreq.bldrdoc.gov	Colorado
128.138.140.44	utcnist.colorado.edu	Colorado
192.43.244.18	time.nist.gov	Colorado
131.107.1.10	time-nw.nist.gov	Washington (Microsoft)
69.25.96.13	nist1.symmetricom.com	California
216.200.93.8	nist1-dc.glassey.com	Virginia
208.184.49.9	nist1-ny.glassey.com	New York
207.126.98.204	nist1-sj.glassey.com	California
207.200.81.113	nist1.aol-ca.truetime.com	California
68.216.79.113	nist1.columbiacountyga.gov	Georgia

NOTE: If AirAura's NTP function is enabled the network that AirAura is connected to must have a path for retrieving the correct time, whether it is from a local NTP server at the station or one that is reachable over the Internet. The NTP time function cannot work if AirAura is not connected to a network, or is connected to a network with no path to an NTP server.

Sound Menu

Selecting the *Sound* tab will reveal the ten controls of Guru GUI, six for FM and four for HD. It is these simple to use but quite powerful controls that can morph the sound of any Factory preset into over one million different sounds and textures.



There are two sections to the *Sound* screen. The left side contains the controls that modify the sound of the FM signal path while the far right side hosts the controls for adjusting the sound of the HD signal path. Note that the FM and HD signal paths share the common five band AGC/SST structure so as to maintain a consistent spectral texture as an HD receiver blends from analog to HD. The *AGC Depth* and *Compression* sliders are *shared* between the FM and HD signal paths.

Each sound slider is intelligently connected to myriad controls (over 425) that are accessible via the full-control "Expert" level Professional GUI. As each Sound slider is adjusted to the eleven positions within its 0 to 10 range, many complementary adjustments are occurring behind the scenes to modify the sound according to the naming and purpose of the controls. The following is an explanation of what each FM Sound slider control does, again keeping in mind that the *AGC Depth* and *Compression* slider are shared between the FM and HD processing chains.

FM Sound Adjust

AGC Depth

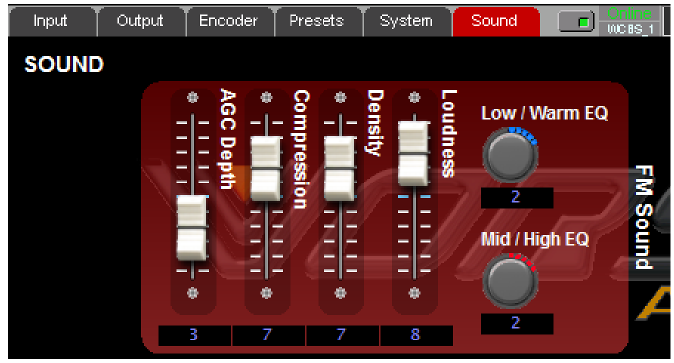
The *AGC Depth* control adjusts how deep the AGC's *long term* gain reduction will be with typical program material and with appropriate input levels. Higher settings cause deeper long-term gain reduction and the additional "reserve" gain caused by raising the setting of this control will tend to hold up fading songs longer. Conversely a lower setting of the control causes gain reduction that is less deep, leaving less gain available to hold up program levels of material (such as songs) as it fades.

Note that the *AGC Depth* slider does not modify the time constants within the AGC/SST section. That is the job of the *Compression* slider, discussed next.

Compression

The *Compression* control adjusts the medium term program density and how obvious the processing will sound. Lower settings cause the processing to be virtually inaudible as it corrects levels and spectral balance as incoming program elements change. Adjusting the slider to higher numbers makes the gain controlling action more audible. Higher Compression settings also cause the audio to "fill up" and become more "full."

Density



The *Density* control adjusts how much short-term limiting is permitted. Higher settings will make the audio more dense and "processed" sounding while lower settings will be less so. Stations looking for a "compressed" sound will find the *Compression* and *Density* controls quite useful for tailoring the "feel" of their audio. Note that increasing the Density setting will not cause the final clipper to be overdriven because Guru GUI's intelligence makes other adjustments behind the scenes to prevent it. In other words, distortion will not increase simply because Density was increased.

Low/Warm EQ (we're leaving Loudness for last!)

The *Low/Warm EQ* control adjusts how powerful low frequencies sound on the air. When the control is at "0" and has not been touched since a preset was last taken the factory assigned equalization for that preset is being applied. Adjusting the control either side of "0" increases or decreases the factory equalization settings. In general, setting the control to lower numbers reduces on-air bass while moving it to higher numbers increases it.

Mid/High EQ

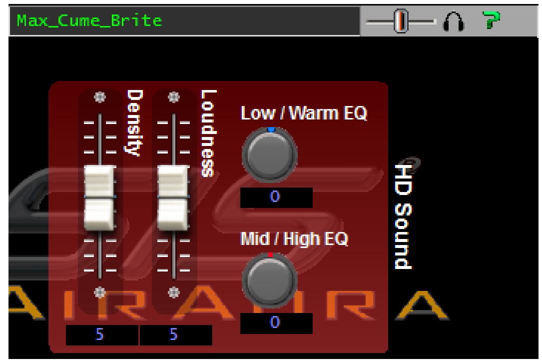
The *Mid/High EQ* control adjusts how the higher frequencies sound on the air. When the control is at "0" and has not been touched since a preset was last taken the factory assigned equalization for that preset is being applied. Adjusting the control either side of "0" increases or decreases the factory equalization settings. In general, setting the control to lower numbers reduces on-air highs while moving it to higher numbers increases them.

Loudness (finally!)

The *Loudness* slider is used to set the desired on-air loudness. The most complex control within the entire Guru GUI structure, it intelligently communicates with three powerful sections of AirAura's processing; the 31-band limiter, the FM Limiter/Clipper and the MPX processing. Its intelligent hand on these other processing sections automatically optimizes the distortion vs. loudness tradeoffs as the *Loudness* slider is adjusted.

HD Sound Adjust

Density



The HD *Density* control adjusts how much short-term limiting is permitted within the multiband and broadband limiting sections. Higher settings will make the audio more dense and "processed" sounding while lower settings will be less so. Stations looking for a "compressed" sound on their HD signal will find both the *Compression* and *Density* controls quite useful for tailoring the "feel" of their audio. Note that increasing the Density setting will not cause the lookahead limiter to be overdriven because Guru GUI's intelligence will make other adjustments to prevent it. Distortion and/or audible pumping will not increase simply because Density was increased.

Low/Warm EQ

The *Low/Warm EQ* control adjusts how powerful low frequencies sound on the HD signal. When the control is at "0" and has not been touched since a preset was last taken the factory assigned equalization for that preset is being applied. Adjusting the control either side of "0" increases or decreases the factory equalization settings. In general, setting the control to lower numbers reduces on-air bass while moving it to higher numbers increases it.

Mid/High EQ

The *Mid/High EQ* control adjusts how powerful the higher frequencies sound on the HD signal. When the control is at "0" and has not been touched since a preset was last taken the factory assigned equalization for that preset is being applied. Adjusting the control either side of "0" increases or decreases the factory equalization settings. In general, setting the control to lower numbers reduces on-air highs while moving it to higher numbers increases them.

Loudness

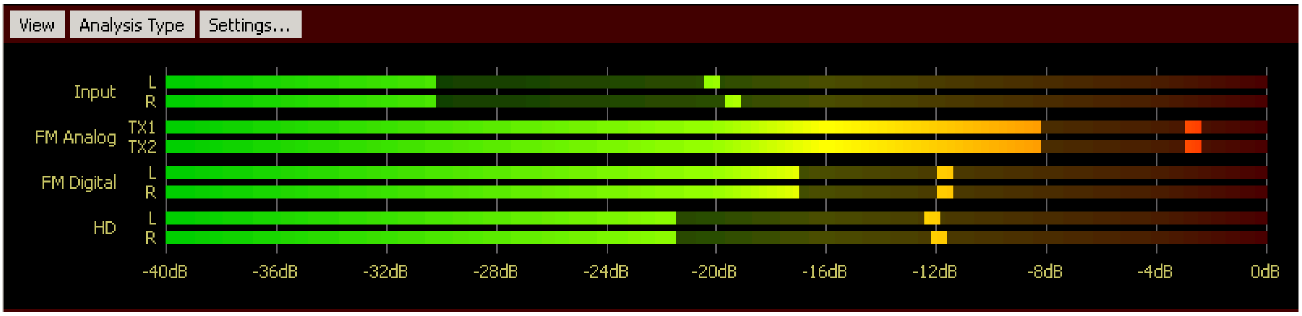
Once the previous controls have been adjusted for the desired sound texture, the *Loudness* control can be used to set the desired HD loudness. The Loudness slider intelligently communicates with the HD processing's 31-band limiter and its lookahead limiter.

As the *Loudness* control is adjusted upwards and away from "5" the sound will get denser and louder. The opposite occurs when the control is adjusted counterclockwise away from "5."

Note that the Loudness control's interaction with other processing sections has been tuned to avoid undesired artifacts such as audible pumping and/or distortion when the control is set to either extreme.

Audio Input and Output Metering

AirAura provides high resolution level metering for all input and output signals.



The topmost pair of horizontal bargraph meters show the applied input levels of the selected audio source, after that source's Gain control. Like many meters in AirAura the input level meters are calibrated in dBFS, for decibels Full Scale. The right hand "0dB" marking represents the absolute level where there are no more digital bits left for defining the input signal – any signal above this will cause severe distortion. The left hand "-40dB" marking represents the audio level that many of the silence sense/ audio failover circuits use for making their logic decisions.

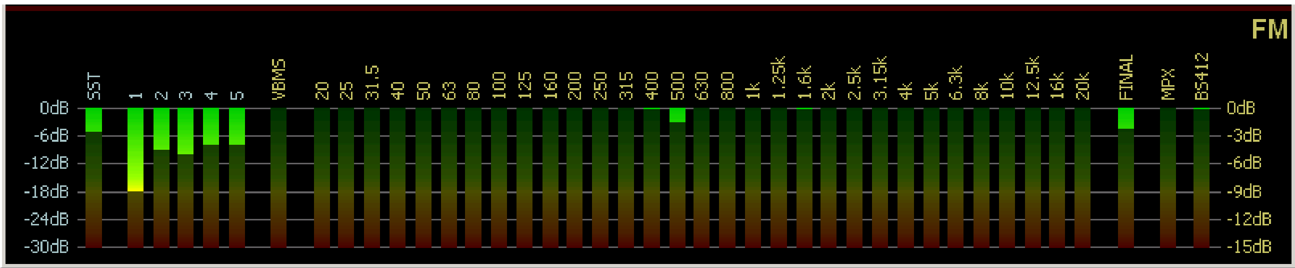
So, what are normal input levels? Because we are working in the digital domain great care must be taken to ensure that audio levels aren't run so high as to cause them to run out of headroom. Good engineering practice dictates that levels arriving from the studio or STL should be peaking between -20dB and -12dB on the input meters and no higher. This allows headroom for the occasional "hot" audio level.

Operating the Input meters at lower indications won't bring up noise or other artifacts that we might be familiar with in analog circuits. With AirAura's 144dB internal dynamic range there is virtually no chance of increasing noise by running with "light" audio input levels, though users would likely find that they may need to increase the setting of the AGC Drive control to compensate.

AirAura's FM Analog, FM Digital and HD output levels are also shown on the bargraph level meter display and their scales match that of the Input meter display. The big difference in the output meters is that because the output levels they are displaying are "post processing" peak levels are not only known, they are very well controlled. Therefore there is no need to leave headroom for output signals as must do for input signals.

Audio Processing Gain Reduction Displays

At the bottom of the GUI display are the vertical bars representing the amount of activity occurring in each of AirAura's processing sections.



The meters are arranged in seven logical groups as follows:

- SST Activity meter;
- Five-band AGC gain reduction;
- VBMS Activity meter;
- 31-band Limiter gain reduction activity;
- Final Limiter gain reduction or Clipper activity;
- MPX Limiter gain reduction or Clipper activity;
- BS-412 MPX Power gain meter.

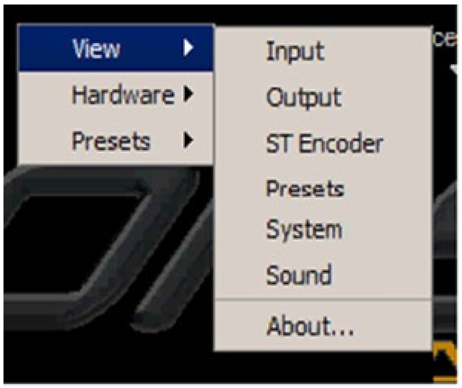
Important Gain Reduction Display Metering Notes:

- The SST and five band AGC utilize a meter scale that goes from 0dB at the top to -30dB at the bottom.
- The 31-band limiter, main FM and MPX limiter/clippers and BS-412 algorithm utilize a meter scale that goes from 0dB at the top to -15dB at the bottom.
- The VBMS meter does not need nor does it have a calibrated scale.

The 31-band limiter does not need to do much work to gain its advantages. In fact, less is more. Gain reduction deeper than 5-6dB is not only unnecessary, but doesn't generate any further gains in loudness. Unlike the limiters in all other broadcast audio processors, when 6dB, 10dB, or even more is required in order to generate consistency and competitive loudness, the 31-band limiter's best sound quality and loudness results when its bands are just being "tickled."

Accessing Menu Options

Right clicking anywhere on the AirAura Control Panel will open a pop up menu tree with access to various items. The first item in the list is *View* and it presents the user with several choices for what he wishes to view.



Input: Makes the Input menu screen visible.

Output: Makes the Output menu screen visible.

ST Encoder: Makes the Stereo Encoder screen visible.

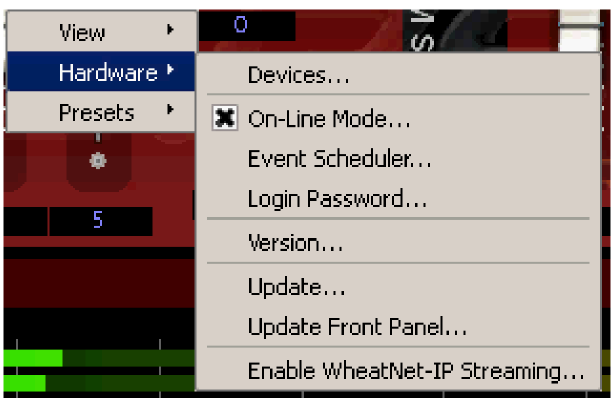
Presets: Makes the Preset management screen visible.

System: Makes the System screen visible.

Sound: Makes the Sound adjustment screen visible.

About...: Displays the *GUI* software version.

Next in the list is *Hardware*. When selected, the *Hardware* item opens up a list of six items.



Devices...: Displays the Devices dialog where the GUI is configured to communicate with AirAura hardware.

On-Line Mode...: Shows or changes the On-Line status of the GUI.

Event Scheduler...: Opens the Event Scheduler for viewing or editing.

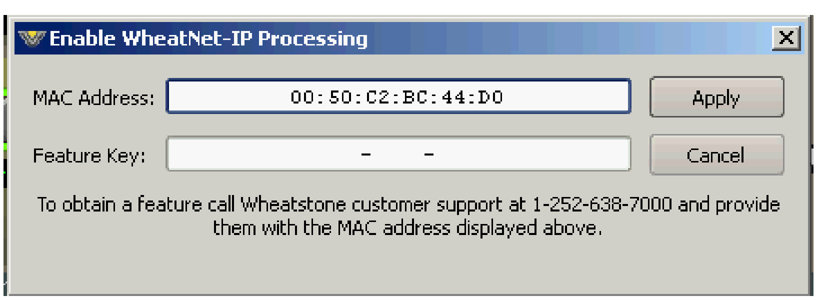
Login Password...: Sets the Login password for the GUI when connecting remotely.

Version...: Displays the software and firmware versions currently loaded in the *hardware*.

Update...: Opens the hardware update dialog where a .VBN update file is selected for download to AirAura.

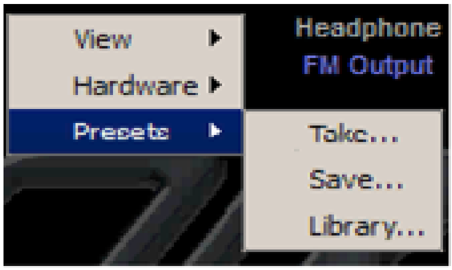
Update Front Panel...: Opens the front panel software update dialog. Here, new LBN files may be downloaded to the front panel to enable software updates.

Enable WheatNet-IP Streaming...: Opens the dialog box shown below to enable WNIP audio input and output streaming.



Note that a WheatNet-IP Feature Key must be obtained from Wheatstone to enable this feature. The Feature Key is unique to each instance of AirAura hardware and Feature Keys are not transferable.

Next in the drop down list is *Presets*. When selected, the *Presets* item opens up a list of three items:

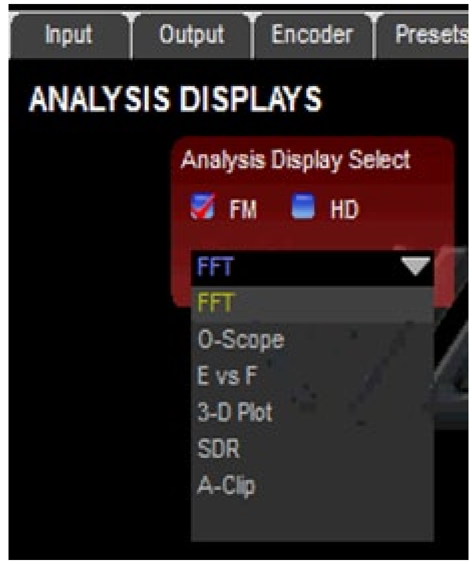


Take...: Opens the Take Preset dialog to allow the current preset to be changed.

Save...: Opens the Preset Save dialog with prompt for the name and preset slot for the current settings to be saved to.

Library...: Opens the preset Library Manager which allows presets to be packaged and moved to/from the hardware and host PC, etc.

Analysis Menu



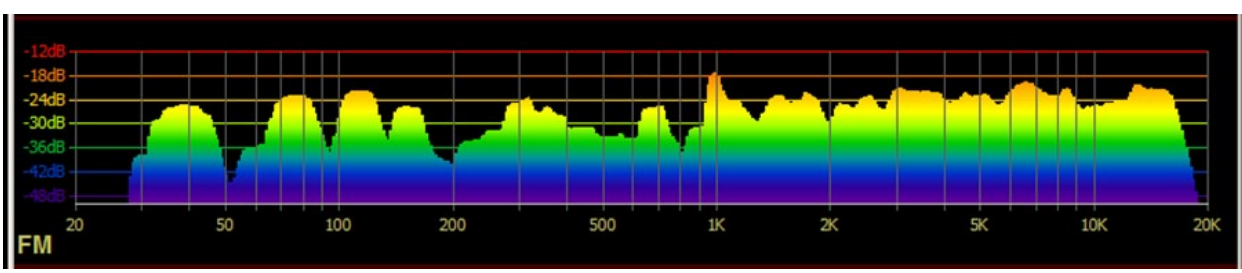
AirAura is equipped with extensive processing analysis features that show what the processing is doing and by how much and when. The front panel and Guru GUI both have drop down menus where the analysis to be viewed may be selected. There are six options in this drop down menu:

- **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 AirAura.
- **O-Scope** – Displays the audio waveforms at the output of the FM or HD processing chains.
- **E vs F** – Displays the measured RMS Loudness within each of the 31 FM or HD limiter bands.
- **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.
- **SDR™** – Spectral Dynamic Range™ (a Vorsis exclusive). This display shows the amount of *dynamic range* remaining in each of the 31 FM or HD limiter bands, after processing.
- **A-Clip** – This is AirAura's final FM clipper activity display (another Vorsis exclusive) 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.

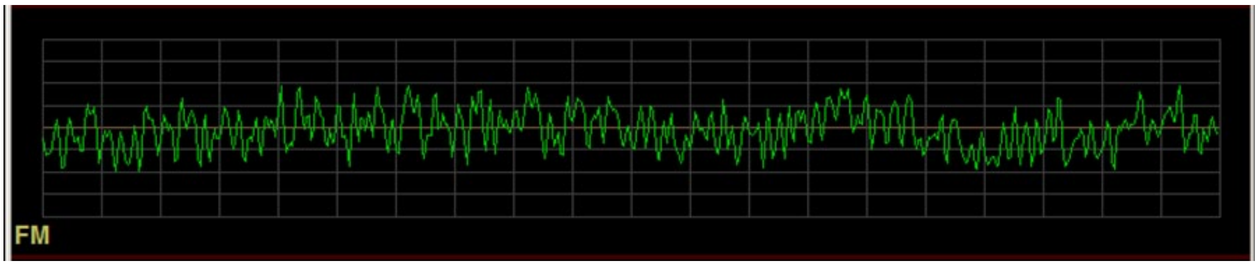
FFT

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.



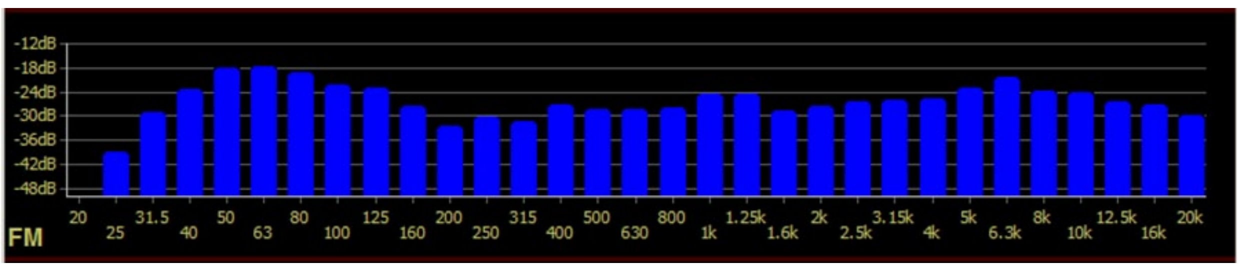
O-Scope

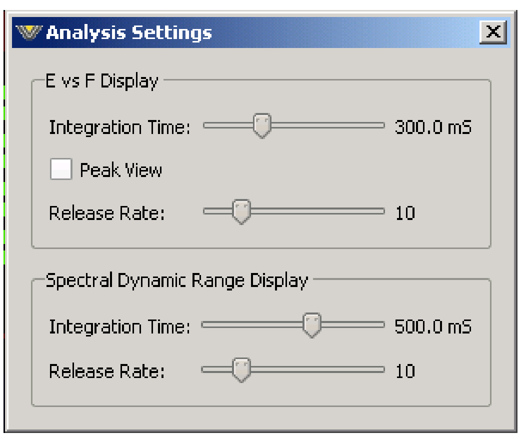
This is a time-domain (oscilloscope-type) display of the audio at the output of the FM and HD final peak control sections. The FM O-Scope display shows what the pre-emphasized and clipped audio looks like. The HD path, with its more "gentle" lookahead limiter, will appear less "aggressive" than the FM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



E vs. F

This display represents the RMS Loudness generated in each band of the FM or HD limiter.

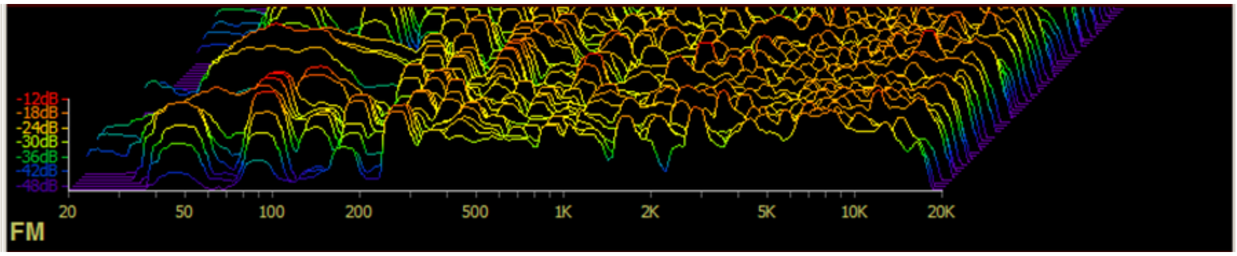




The E vs. F display has a submenu (shown left) that allows customization of the RMS Integration Time, the display Release Rate, and whether or not peak levels are to be displayed. The factory default Integration Time is 300mS – the same as a standard VU meter – which is also a fair estimator for how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

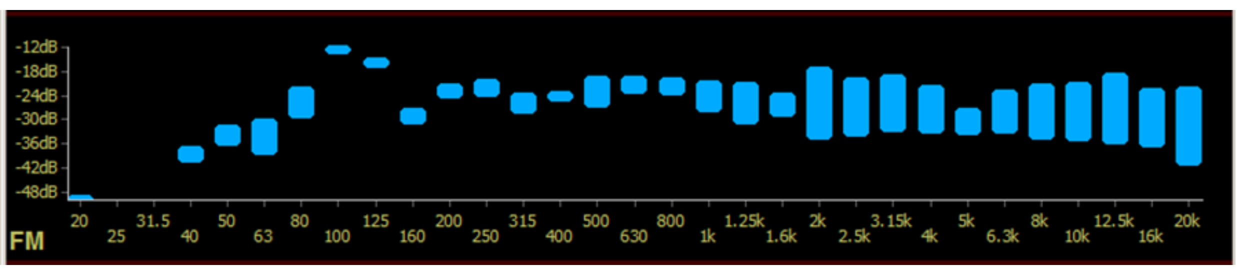
3-D Plot

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.



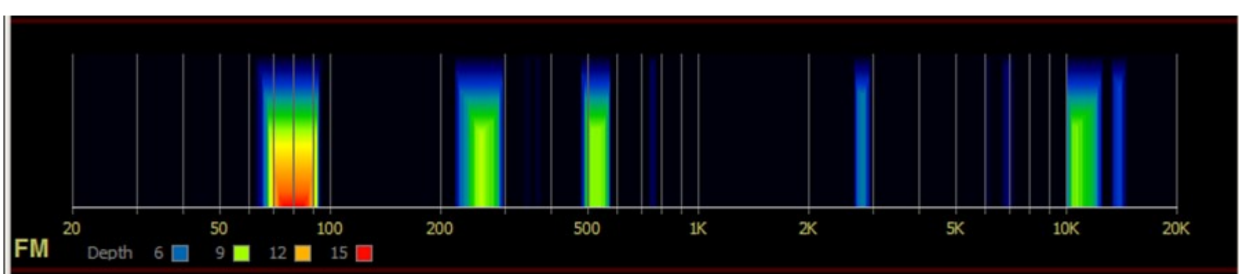
SDR™

During development of the AirAura product we invented a new and useful processing analysis tool, the *Spectral Dynamic Range™ Meter*. This special analyzer was designed to measure the dynamic range exiting each of the 31 band limiter bands (i.e., how loud the audio is at the output of each band). 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 a limiter 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.



A-Clip

During development of our new AirAura FM clipper we invented a way to watch it work. Known as the A-Clip display, it shows in detail *where* in the audio spectrum AirAura's distortion management is working. The 'ghosts' that appear show where in the audio spectrum the distortion management is working and the color of the display shows *how much*. Like weather radar images, the 'redder' the ghosts are, the more active the distortion manager is in that part of the audio spectrum.



AirAura Pro GUI

Chapter Contents

AirAura Pro GUI

The *AirAura Professional Graphical User Interface* (or Pro GUI) is an advanced user interface which reveals all of the controls existing within the FM and HD processing structures – more than 400 of them. 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 AirAura Pro GUI allows you to adjust the processor and see, in real time, how it is reacting to the audio being processed. All Wheatstone Pro GUI's, have been carefully designed so that no adjustment is more than two mouse clicks away.

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

Another option is the ability to connect to AirAura using a PC with a wireless adaptor. Connecting to the AirAura using its wireless adaptor will be explained later on.

Getting Started

The AirAura Pro GUI is a Windows™ based program designed to be intuitive and straightforward to use. The Pro GUI installer is supplied on the product CD as an executable program. Though the GUI may be installed to any folder on the host computer, the default path is: "C:\Program Files\Vorsis\Airaura." After installation, the GUI can be started by clicking on the AirAura item in the Windows Start menu.

Connecting With The Pro GUI

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

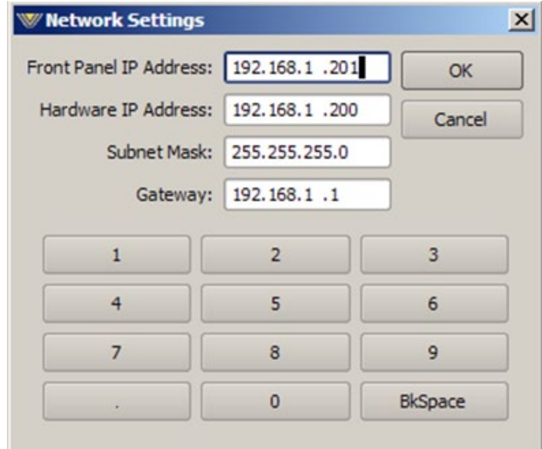
The AirAura should be installed, powered-up, and verified to be operating normally. A CAT5 Ethernet cable connected to one of the rear panel's two 100baseT LAN ports should be connected to the Ethernet port on the host PC. AirAura is Auto-MDIX so either a straight through or crossover cable may be used.

Configure the desired hardware TCP/IP address for the AirAura by using the front panel's "Right Click\File\Settings" navigation to get to the Network Settings window.

On the next page you will see the following four entries (default settings shown):

Front Panel IP Address

This is the reserved address that the front panel (not the remote GUI on the PC!) uses to communicate with the AirAura 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 AirAura's "Hardware" address, but on the majority of networks should never end in "0" or "255."



Hardware IP Address

This is the address of the AirAura's DSP array and is the address that remote GUIs running on host computers will use to connect to and control AirAura 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 AirAura will use to reach connections outside of its own internal network. If you are on a local LAN segment or connected to the AirAura by a direct Ethernet connection, the Gateway address is not usually necessary.

After determining the IP address that you wish the AirAura to use, enter that address in the AirAura's *Hardware IP Address* field using the navigation buttons. Please clear the entire IP address field using the backspace key before entering new data to ensure there are no leading or trailing blanks in the address field. Using the period key to navigate from one IP address field to the next will ensure no bank spaces or leading zeros.

Next, using the same sequence as above enter a new *Front Panel IP Address* by adding "1" to the IP address entered for the hardware in the previous steps.

Example:

If 192.168.0.200 was entered as the hardware address, the correct Front Panel GUI address will be 192.168.0.201. Note that the GUI address is not utilized externally by the AirAura, so there is no traffic to/from this address external to the AirAura. However this address will appear as a valid device on the local network so this address should be reserved for the AirAura. Note that another device on the network configured for the same IP address as AirAura's front panel may cause the front panel to not be able to communicate with the host hardware.

Once entries to AirAura's Network Configuration screen are completed, click on the *OK* button at the top right corner of the dialog. Next, when prompted to reboot the AirAura for the changes to take effect, wait five seconds and then interrupt power to the unit for five seconds. When power is reapplied AirAura will boot up and the new IP address configuration will be valid.

* Using a Gateway address on the wired side of the WLAN module can cause a local connected network to become confused if it and the gateway addresses of the module are the same. It is sometimes best to program the WLAN module's Gateway address to an unused address on the locally connected network. If an invalid entry is used for the WLAN module Gateway that address can no longer be used for the module's web-based configuration and the wireless address will need to be used instead.

Configuring The Pro GUI

Now that the AirAura has a valid network address the Pro GUI must be configured so that it can find AirAura on the network. Assuming the GUI is now installed, start the program to bring the GUI on the screen. Next, on the right side of the GUI locate the *Devices* button and click on it. This will open the *Edit Device* dialog which is used to tell the GUI what IP addresses and names AirAuras may have.

The Name field can be left blank or used to give AirAura 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 AirAura and then click OK.

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

It will be green when the AirAura connects to indicate that the GUI is now communicating with the AirAura. Once the GUI has been made aware of the AirAura's existence in this manner, it will always appear in the list of "AirAura Devices" and it will be instantly accessible from anywhere, any time.

When connected to an AirAura the status bar will indicate *Online* and the "Devices" text will show the name of the AirAura that it is currently communicating with (in this example, the AirAura has been named *WCDY*). The status bar will also display the name of the preset that the AirAura 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 red 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.

Connecting Directly Without a LAN

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

NOTE: The controlling PC and the AirAura'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).

Wireless Connection

Many users will want to make adjustments to the AirAura "wirelessly" while in a remote location such as their home, office, or car. This can be accomplished in a variety of ways, but normally the GUI and host computer will be configured to connect to

AirAura's own wireless interface. AirAura can also be connected to a wired network and accessed wirelessly if that network has a wireless access point connected to it and the host PC has permission to connect to that network.

Please see the section on AirAura's wireless features for more information on how to configure AirAura's built-in wireless network.

The AirAura and Internet Security Concerns

AirAura 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 AirAura and use it as a pathway to the rest of the network to which it is connected.

About DHCP and The AirAura

AirAura 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's Network Protocols and Ports Used

The AirAura uses both TCP and UDP protocols to communicate with the remote GUI. TCP is used from GUI to AirAura because its high reliability ensures that all control changes sent to the AirAura will be received with 100% accuracy. The UDP protocol is used by AirAura 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 AirAura requires these ports to be open through any firewalls for successful connectivity.

Using The AirAura Pro GUI

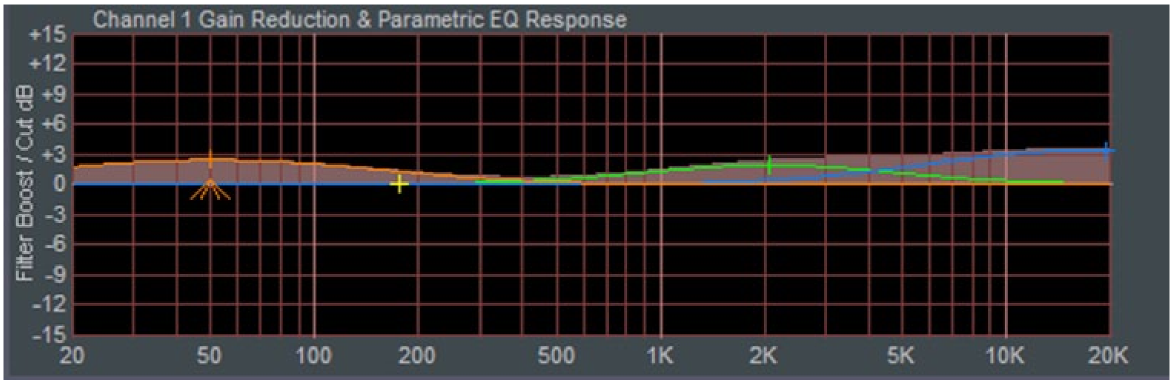
The GUI may be positioned on the host computer's screen by left-clicking and-holding the "Vorsis" logo, and dragging the GUI to the desired position. In normal "Windows" fashion, the GUI may be minimized on the desktop (taken off the screen) and/or closed by way of the familiar controls at the extreme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

Descriptions of the features and functions of the GUI will follow different discrete regions of the screen; it will begin with the most eye-catching region – the Dynamic Displays – and then we'll explain the Control Area above it, followed by the Side Bar buttons, and lastly the Title Bar.

Dynamic Displays Region

Frequency-Domain Graph

The most dominant display on the main screen is a large graphical area depicting the audio frequency range of 20Hz – 20kHz arranged in logarithmic fashion with gradations at selected frequencies. The scale of the vertical axis changes according to its context; however the overall vertical range remains at 30dB with 3dB graduations.



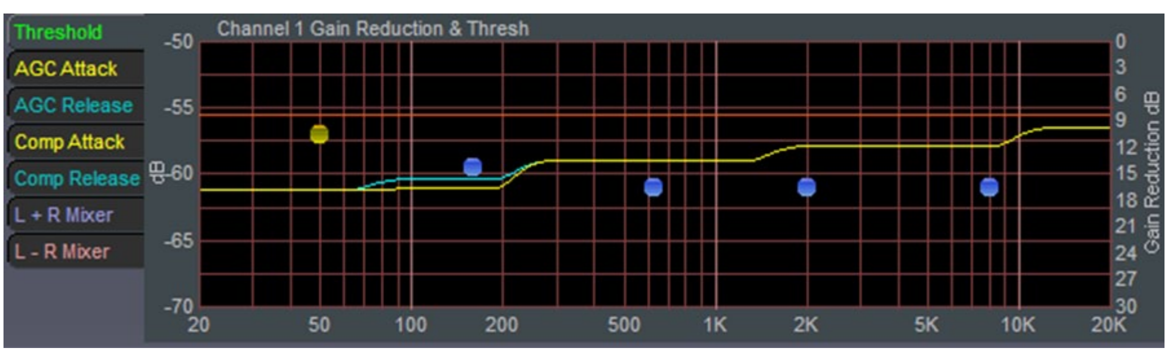
AirAura Graphic EQ Display

Managing the Blue Dots - Tutorial

We've worked very hard on the design of the Wheatstone graphical user interface to make it intuitive, friendly, and easy to navigate and interpret, in spite of how many controls the user has access to (well over 400).

Each of the processing section's screens has a graphical area where not only visual representations of processing activity are shown, but in some cases individual adjustments of different sections of the processing can be made.

All of the operating parameters of the five band AGC and the 31-band HD and FM limiters can be individually adjusted by manipulating the positions of various "Dots" on that section's graphical screen. Each "dot" represents one processing "band." By double clicking a blue dot (to highlight it and turn it Orange) and then dragging it with the mouse a particular parameter for a band can be adjusted.



AirAura 5 Band AGC/Compressor Adjustment Display

By right clicking on the graphical display area with your mouse the following options will appear:

By default, all adjusters (the blue dots) are Grouped. That is they are coupled to each other so that if you grab one with the mouse (single left click and hold) and then slide the mouse up and down, all blue points move up and down together.

When a blue dot is double-clicked, it turns orange to indicate that it has been selected. It then may be moved individually, and to anywhere along the vertical scale.

If another blue dot is clicked somewhere else on the line, it too will turn orange….

Now if you move the mouse to a point between the two dots selected in the above steps, right click the mouse, and select *UnGroup Adjusters*, all the dots between the first two dots that were double clicked to turn them orange will also turn orange.

At this point all of the dots including and between the first two dots may be individually adjusted up or down. If you now place the mouse over the same graphical area, right click again and this time select *Group Adjusters* the points between the first two that you highlighted will turn back to blue. If you then grab the line between these points with the mouse again (left click and drag) you can move the entire line connecting the two highlighted end dots up and down, with the two end points remaining anchored in place. Cool, eh?

Right clicking on the graphical area and selecting *Group All Adjusters* removes all highlighted dots turning them blue again. Conversely, right clicking and selecting *UnGroup All Adjusters* turns all (remaining) blue points orange and they can now each be individually adjusted. The *Flatten* function, although next in the list, is best explained last….

Highlight a single dot again … any dot will do. Then right click the graphical area and select *Proportional Drag*. Now left click any dot to the left or right of the dot you highlighted and drag it up or down. See how the line behaves? It pivots at the point where you highlighted the dot, and it bends at the point where you grabbed it with the mouse. Virtually ANY curve can be created by following these steps. Now right click again and this time select *Flat Drag*. Grab the line again… anywhere (left click and drag)… and move it up and down. See how the line pivots at the highlighted point but remains flat above or below it as you raise and lower the line?

After these exercises the line is probably all messed up. You want to put it back to where it was or at least make it all flat again without having to move every single point back, right? But how do you do that? It's easy ... right click on the graphical area again and select *Flatten*. If there were any points still highlighted orange you'll want to first click *Group All Adjusters* before you select *Flatten* in order to turn them off. In any case, when you finally click *Flatten* the line will return back to flat and you may move it up and down again with the mouse – this time as a whole.

To review:

Group Adjusters – Gangs the **blue** dots *between* two highlighted **orange** points into a commonly behaving entity.

UnGroup Adjusters – Disconnects any **blue** dots that were grouped by "Group Adjusters."

Group All Adjusters – Gangs all the dots, turning them **blue**, *regardless* of their previous highlighting state.

UnGroup All Adjusters – Un-gangs the adjustment dots and turns them **orange**, regardless of any previous highlighting.

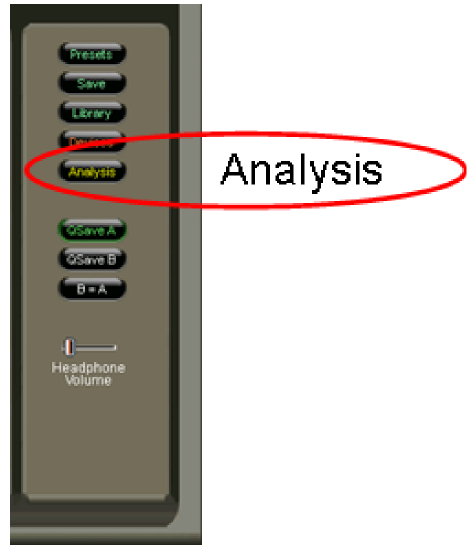
Flatten – Flattens into a horizontal orientation any dots not set to orange.

Proportional Drag – Allows the dragging of adjustment dots to create slopes anchored if desired to a single orange set point.

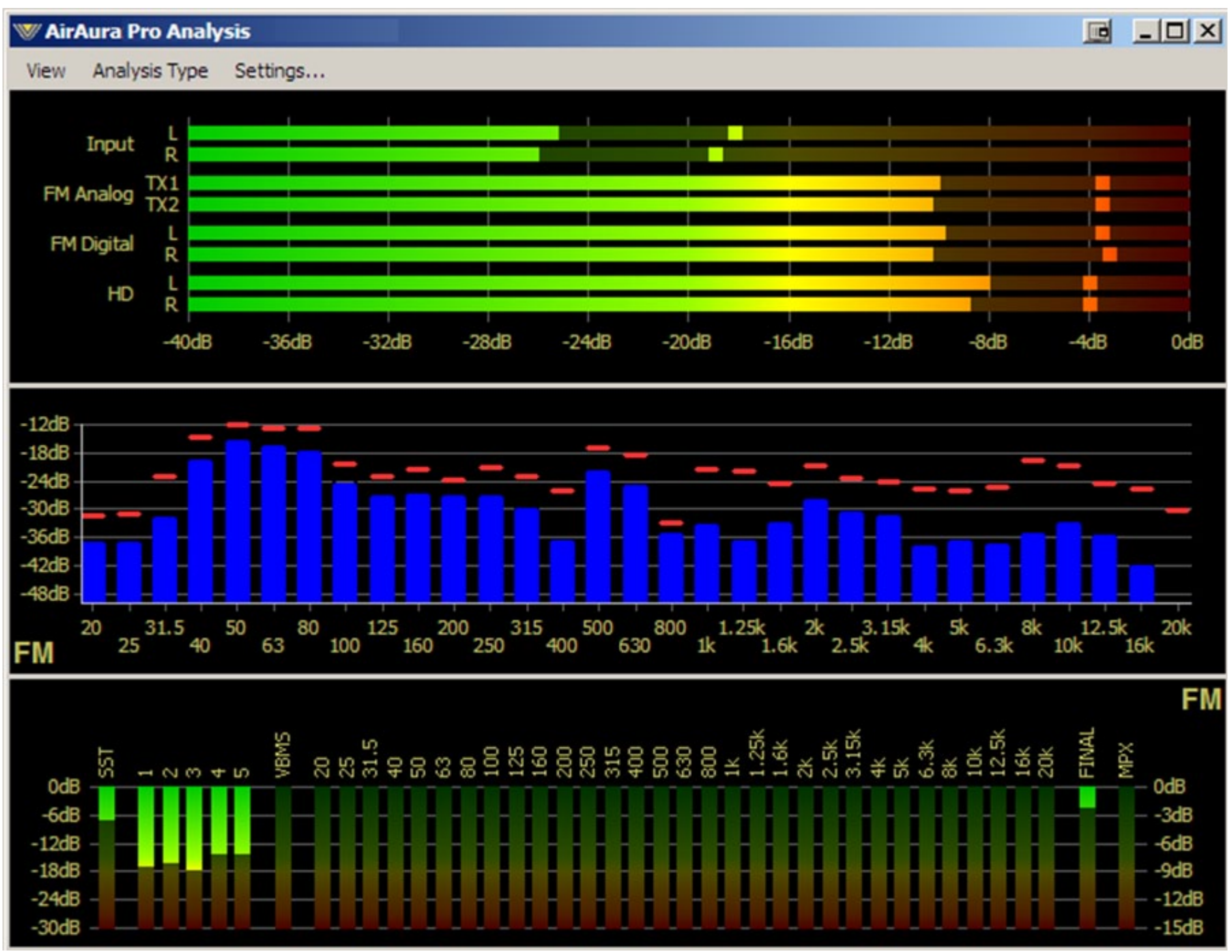
Flat Drag – Allows the dragging of dots together in a "flat" orientation.

Metering and Analysis

The AirAura Pro GUI offers the same analysis options as the Guru GUI. You can open a new graphics window to access the analysis features by clicking on the *Analysis* button in the sidebar region.

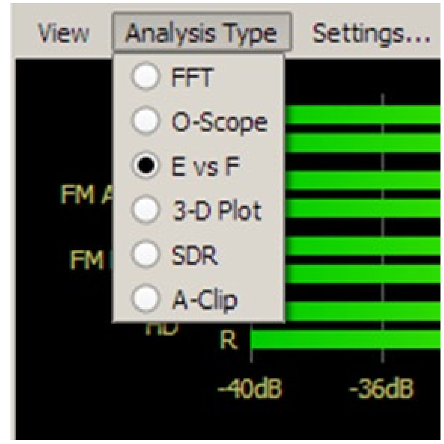


The analysis window opens to a separate window which may be resized, minimized on the desktop (taken off the screen) and/or closed at will using the familiar "Windows" controls at the extreme top right.

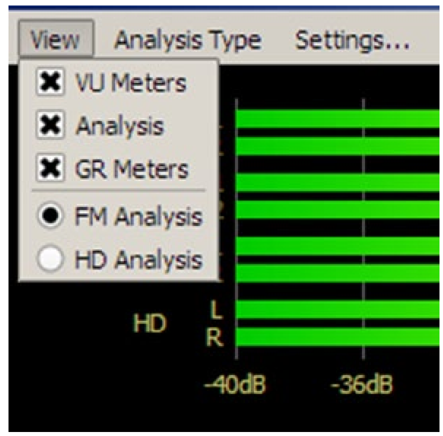


AirAura PRO GUI Analysis Window

To learn more about the different types of analysis available, please see the section on Analysis for an in-depth look at what the different displays mean and how they can help visually adjust the sound of AirAura. Accessing the different visualizations is done through a drop down menu accessed at the top of the analysis window. Click on *Analysis Type* and a drop down menu opens showing the analysis options.

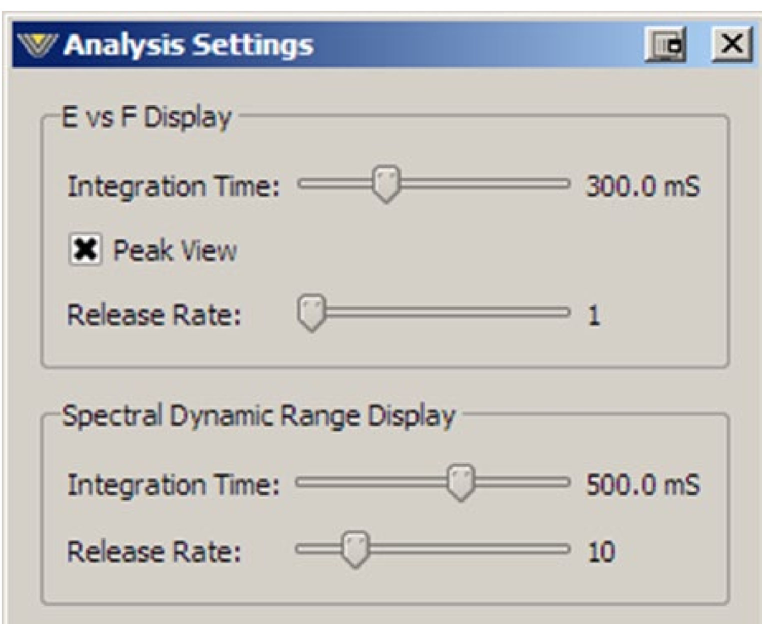


It should be noted that whatever analysis display is chosen by the remotely connected GUI, the same analysis will be displayed on the front panel of the AirAura hardware. Clicking the analysis type on either the front panel or the remote GUI tells the hardware what kind of analysis is being requested. The hardware then 'crunches the numbers' and sends the resulting data to all connected GUIs.



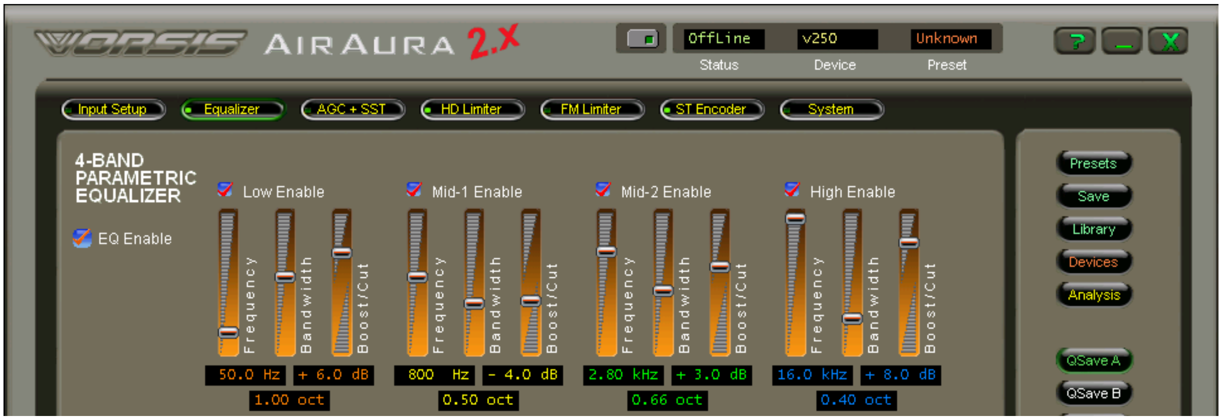
You can customize the Analysis display by choosing which panels you want to monitor. The default is the I/O metering on top, the analysis in the middle window and the gain reduction meters in the bottom window. With the *View* tab, you can turn on or off and of those displays. As seen on the right you can also choose which processing path you want to view... FM or HD.

The *Settings...* button allows you to adjust the E vs. F display to customize its 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.



Control Area Region

This is the large enclosed area directly above the graph and meters. The EQ section is shown here as an example.



All of the knobs, faders, switches, and checkboxes affecting the sound of the audio processing are located within this half of the GUI. To operate them, either:

Place the cursor over the desired control, and use the mouse's scroll wheel to increase or decrease the control's position.

OR

Click on the control, and slide the mouse to move the control – up or right increases the control's indicated position, down or left decreases it. When using the mouse in this fashion (as opposed to using the scroll wheel), the resolution of the control variation can be increased – made finer – by holding down the keyboard's Ctrl key while adjusting the control as above.

OR

In most cases you can double click on the text that indicates the setting value (for example, the window that reads X.XdB under the *Line Gain* knob on the *Input Section*) and a box will pop up where you can enter an exact value and click *OK*.

From left-to-right across the upper portion of the graphical area are a series of buttons corresponding to logically grouped processing sections in signal flow fashion in Input to Output order.

A small green indicator at the left end of each button indicates if any signal processing within that block is active. Left-clicking a button opens the associated control panel to gain access to the controls for that block of signal-processing

INPUT SETUP MENU

This is the first tab on the top far left of the Pro GUI. (Note: the screen that is currently being displayed is indicated by a green background on the selection tab).

Input Screen

The *Input* screen contains controls for the input source selection: *Automatic Input Failsafe*, Analog and Digital input gain controls, input L/R gain balance, *Phase Rotator*, *Auto Mono Threshold* and its checkbox, and the high pass filters which may be operated in Stereo or Sum/Difference mode.

Input Source

The Input Sources can be selected as either Analog or Digital. There is one analog input source, AirAura's balanced audio inputs on its rear panel. There are two types of Digital sources however, standard AES3 and WheatNet-IP.

The currently selected input is shown by a red checkmark.

Input Signal Presence

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.

Input Failsafe

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

If the primary source was digital, 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.

Analog Input Gain

The Analog audio input gain can be adjusted by clicking on the knob and dragging the mouse or touchpad to the right or left. The *Analog Input Gain* level can be adjusted over a +/- 24dB range.

Digital Input Gain

The Digital audio input gain can be adjusted by clicking on the knob and dragging the mouse or touchpad 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.

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.

Auto Mono Threshold

The Auto-Mono threshold can be enabled by clicking on its checkbox and setting the *Auto-Mono Threshold* knob for the desired level when automatic mono blend occurs. Normal settings of this control will fall between about -22dB and -30dB.

What is Auto Mono?

Auto Mono is a feature that automatically and continuously analyzes the L+R/L-R ratio of the incoming program material to determine if the content in the L-R difference channel is valid. If the Auto Mono feature is enabled and the input signal is not true stereo, then the L-R information is discarded and the output of the processor is pure mono.

This feature is most useful in formats that play music such as Oldies, Solid Gold, or other programming that was primarily mastered in mono or that is intended to be mono but might contain small left/right channel balance or phase errors. When the input material has these errors the L-R difference channel contains meaningful signals and that information may be reduced or eliminated with no audible detriment to the sound.

In fact in most cases the Auto Mono algorithm will significantly *improve* the sound of aired material containing such level or phase errors.

A lot of oldies music was originally mastered in mono. In recent years many of the original songs may have been ripped into playout systems using audio compression schemes such as MPEG Layer II or MP3. Plus, although the songs may have been originally recorded in mono, they were likely ripped into the playout system while it was in its stereo record mode, so there could be uncorrelated stereo information in the L-R channel which is basically “junk.”

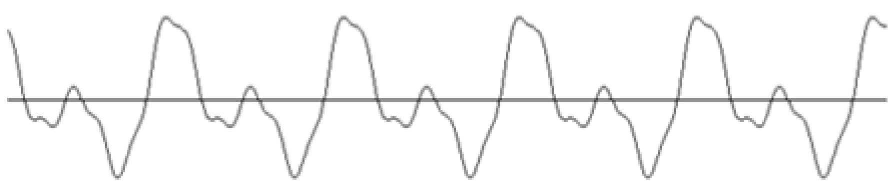
The Auto Mono circuit has been designed to be unobtrusive in its action while removing such “junk.” It may typically be left enabled so that it has an opportunity to correct any material that may pass through AirAura which would sound better if the difference channel were cleaned up.

A typical operating threshold setting for the Auto Mono circuit with most program material is usually between -24dB to -36dB. Higher settings (less than -20dB) may cause unnecessary falsing on certain material while the lowest setting (-40dB) may allow suspect material to pass uncorrected. The control has a range of -40dB to 0.0dB.

Phase Rotator

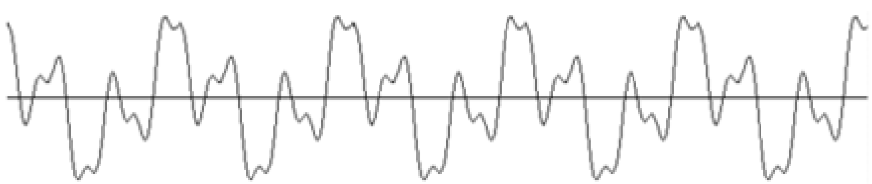
Human voice is usually asymmetrical by nature, which means that it is usually “peakier” in one polarity than the other, hence asymmetrical. What the Phase Rotator does is fix these asymmetrical peaks, and the way it works is by changing the phase of signal harmonics compared to their fundamental frequencies. 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.

Ordinarily, human voice looks like the waveform below. Notice how the width and amplitude of the upper and lower signal peaks are different? This is “asymmetry”, and it makes the processing of voice waveforms more difficult.



Human voice before *Phase Rotator* algorithm

After passing through the Phase Rotator the same voice appears like the waveform at left. Now the upper and lower portions of the waveform have equal distribution. In the process no sound is lost and no distortion is created. Only the signal’s phase relationships are rearranged.



Human voice after *Phase Rotator* algorithm

HPF - 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 *High Pass Filter* has three operating modes:

Off – No *High Pass Filter* boxes are checked. The High Pass Filter is not enabled.

Stereo – If *Stereo* is checked the High Pass Filter operates on both left and right audio channels equally by separate high pass filters operating at the frequency set by the single frequency tuning control. In this mode the Diff Freq slider is not visible.

Sum/Diff – *Stereo* is not checked, opening a second tuning slider to reveal adjustments for the stereo difference channel. Separate checkboxes, one for the *Sum Freq* slider and one for the *Diff Freq* slider are visible. Checking a box activates a high pass filter on that channel, sum or difference (L+R or L-R). Separate frequency sliders are available for adjusting the high pass filter corner frequency for each signal path. The mono (L+R) and stereo (L-R) signal paths may have different high pass filter cutoff frequencies.

Why Different High Pass Filter Operating Modes?

In most program material there is very little very low frequency energy in the difference (L-R) signal. In fact, what low frequency energy is there isn't typically correlated with the program material (it's hum, etc.). Therefore by setting the Difference channel to a higher filter cutoff frequency than the Sum channel this noise can be rejected. Good starting settings for the Sum and Difference sliders when in the Sum/Difference High Pass Filter mode is 30Hz and 180Hz respectively. If Stereo mode is selected a High Pass Filter frequency of 30Hz is a good setting.

The purpose behind setting the High Pass Filters as just described is that the Sum channel carries most of the bass energy in the majority of popular program material. The Difference channel on the other hand rarely contains bass or other low frequency signals and what signals there are, are probably not desired.

EQUALIZER MENU

Parametric Equalizer Screen

This powerful *4-Band Parametric Equalizer* has four identical and independent bands. Each of the four bands can be switched in or out independently, and is fully adjustable in three ways:

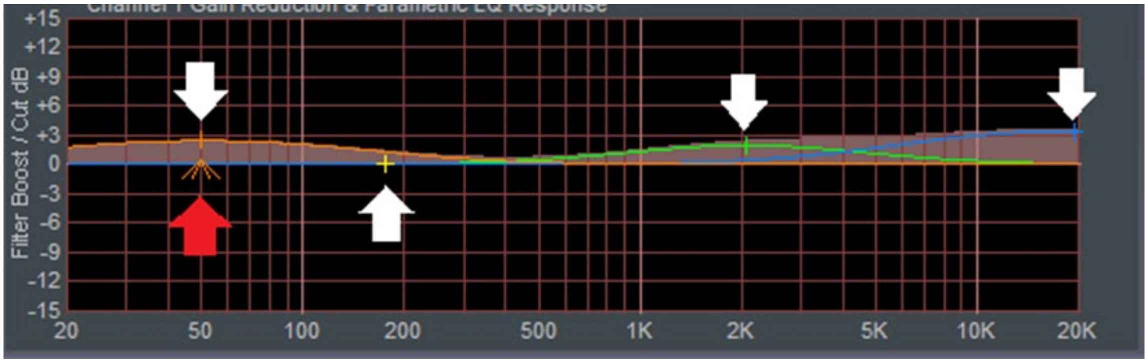
Center Frequency	**—**	**20 Hz to 20 kHz.**
Bandwidth	**—**	**0.2 to 3.0 octaves.**
Boost/Cut	**—**	**+/- 14.0dB.**

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.
2. The second method of adjusting the parametric equalizers is by directly manipulating the actual curves on the graphical screen below using the mouse (the Controls checkbox must be checked to see the curves). See the next page for how to manipulate the curves directly.
3. 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.

Manipulating Equalization Curves via Mouse

Double clicking on the small crosshair at the top (or bottom if it's below the line) of the curve turns that section on and off (white arrow)



Clicking on the crosshair 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).

Clicking on the small "tent" underneath (see red arrow), and holding the left mouse button down while sliding the mouse left or right allows changes the bandwidth, or "Q," of the equalizer section.

EQ Enable

This is the master Equalizer In/Out button. When this box is checked the equalizer section is "IN," even if no other section of the equalizer has been enabled by clicking the *Low Enable*, *Mid-1 Enable*, *Mid-2 Enable*, or *High Enable* buttons. When any of these checkboxes is checked the controls for the EQ section below it or enabled.

EQ Position

The parametric EQ can be patched in so that it is BEFORE or AFTER the five band AGC/Compressor. We always recommended that the EQ be used ***BEFORE*** the five band AGC/Compressor. The reason for this is that if radical EQ settings are being used they will tend to be smoothed out by the action of the multiband AGC. We feel this is the EQ's most 'harmless' but also most useful setting.

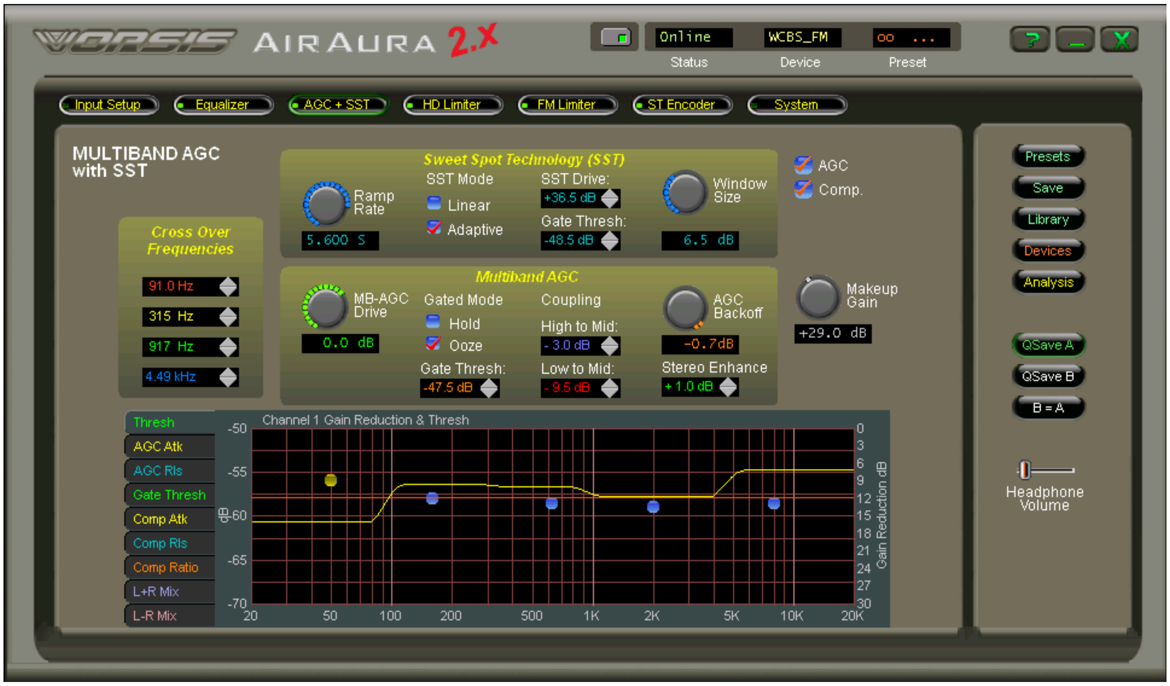
If the EQ is placed AFTER the multiband AGC there is only the 31-band limiter remaining to manage any wild EQ curves that may have been dialed in. Post AGC EQ may negatively impact the consistency of the sound simply because the 31-band limiter is NOT another multiband AGC. It is a limiter that rarely does any work, and when it does work it is only to control signal peaks. Using the EQ after the multiband AGC and running the 31-band limiter "hard" in order to try to compensate for this will not increase loudness. Instead, and like any overdriven limiter, it will cause listener fatigue.

Remember the Post-AGC Mixer!

Because the five band AGC has an output mixer there is already an "EQ" on the output of that stage which may be used to modify the spectral balance. It is important to realize that if the Parametric EQ is placed after the multiband AGC *and* the AGC output mixer is being used to modify the spectral balance, there are now <u>two</u> equalizers in series. Great care must be used when operating the parametric equalizer and multiband AGC output mixers in tandem because depending on control settings it can be very easy to wind up with too much boost or cut in one part of the audio spectrum.

AGC + SST MENU

The AirAura has three sections dedicated to dynamics control, the Automatic Gain Control (AGC), Compressor, and Sweet Spot Technology™ (SST). All three are inextricably linked by DSP algorithms which logically connect and control them. All three sections will be collectively discussed in one chapter rather than separately.



Multiband AGC with SST Screen

What is SST?

Unique to Vorsis processing is a special algorithm, SST, or Sweet Spot Technology™ . The SST operates by incorporating data derived from the input audio, the multiband crossover, the AGC and Compressor gain control behaviors, and the settings of the user controls. This data is used to derive special control signals that manage the multiband AGC/Compressor so that it operates within the desired gain reduction and spectral balance window. SST is quite adept at maintaining the medium and long term "sound" of the processor as incoming program levels and spectral balance wander about.

The intelligence of SST enables the AGC and Compressor to maintain a consistent spectral and dynamics texture over widely ranging signal input conditions. Without the benefit of SST, the AGC and compressor would have little advantage over other audio processors, which all suffer from the inability to *invisibly* manage medium and long-term spectral consistency and density.

A "broadband AGC" is never used prior to the AGC and Compressor in Wheatstone processing. Such simplistic gain control devices work 'blind' because they only see their own input and output audio. In fact they know *nothing* about the output of the multiband AGC and it is *this* section of processing that determines the overall texture of how a radio station sounds on the air. SST works smarter. Through a multiplicity of analysis generated control signals, SST calculates and applies corrections that perfectly manage the multiband AGC so that it creates a consistent signature sound across a *very* wide range of input sources and audio levels.

The automatic gain control (AGC) itself operates in five frequency bands, and its medium-term operation is a special derivative of the program-related processing occurring in the Compressor algorithms. While the SST section operates in Left/Right mode, the AGC and Compressors always operate on the program content's Sum and Difference – the highest signal, whether it's in the sum or difference channel, determines the amount of processing to be applied to *both* channels.

SST Controls Overview

SST Drive

The *SST Drive* control adjusts the amount of analysis signal being fed to the SST algorithm which measures the dynamic contributions made by the five band AGC. The higher the *SST Drive* setting the deeper the amount of correction that the SST is allowed to contribute. The SST metering has been arranged to *look* like gain control metering in order to make its operation and adjustment more intuitive even though that's not how it works. Think of the SST meter as showing how much correction it is *allowed* to make to the multiband AGC to keep things sounding nice.

A typical target operating value for the SST meter is between -12 and -15dB. Note that there is no harm in seeing more or less as long as one understands that higher numbers affords a greater long term AGC correction range than lower numbers. A built-in dynamic clamp prevents the SST from governing more than 20dB of correction and the meter indication will simply stop at that value if driven too hard.

SST Gate Threshold

The SST algorithm has been provided its own freeze Gate and Gate Threshold control so that its operation may be optimized for the program format. Typically the SST *Gate Thresh* should be set several dB higher (less negative) than that of the five band AGC to allow the AGC to operate over its design range. The overall SST *Gate Thresh* control range is -70.0dB to -6.0dB.

Note that the SST Gate Threshold sets the audio input level above which all SST correction is turned off and its so-called "gain" is frozen.

Ramp Rate

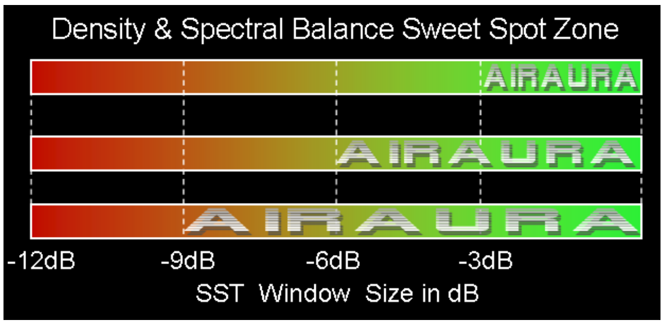
In order for the SST to compensate for all types of programming it should have its speed of correction able to be tailored specifically for the application, and it can – this is what the *Ramp Rate* control does. The *Ramp Rate* control is calibrated in units of time which describe the speed at which the SST corrects the operation of the five band AGC/Compressor.

Note that the setting of the *Ramp Rate* control also influences the behavior of SST's Linear and Adaptive modes, covered later.

A good typical starting value for SST *Ramp Rate* is around 4.000 seconds. The full control range is 2.000 to 7.000 seconds.

Window Size

Because the SST is responsible for maintaining the operation of the five band AGC in its "Sweet Spot" over a very wide range of operating conditions, the size of the sweet spot should be able to be set, and this is what the *Window Size* control does. See the graphic below for some examples:



The graphic at left is a visual for how the *Window Size* control works at settings of -3dB, -6dB and -9dB.

The size of the "AirAura" text inside the shaded bars is how much audio density and/or spectral balance are allowed to "wander" before the SST makes its corrections. By design the SST does not control errors if they go "above zero" because that is handled by the multiband AGC itself. The shaded areas show that the larger the Window Size, the larger the "error" allowed before corrections go into effect.

Window Size settings below 4dB can force the SST to unnecessarily try to correct "normal" dynamic variations which should probably be left alone. Likewise, very large Window Size settings can allow large variations to go uncorrected. Settings lower than 4dB and higher than 7.5dB are not particularly useful, and a normal Window Size setting for the majority of program formats will be between 4.5dB and 7.5dB.

SST Linear/Adaptive Options

The two final controls are behavioral options for the SST and provide choices for Linear or Adaptive. Only one can be chosen. In the Linear mode, the SST always makes adjustments to the five band AGC/Compressor at the speed set by the *Ramp Rate* control.

In the Adaptive mode, SST behavior is just as different from the Linear mode as the name suggests. In this mode the SST is permitted to *momentarily* make *quick* adjustments to the multiband AGC at a rate that is *faster* than the Ramp Rate setting. However, after this brief escape to make corrections it then settles down to the Ramp Rate setting. While this momentary change in behavior is primarily triggered by a dramatic change in program density and/or spectral balance, it is programmed to also trigger under certain other conditions when warranted.

The use of the Linear mode is advised for most programming formats because it is suitable for the widest range of programming. On the other hand, the Adaptive mode is by design, more "aggressive" and might be more appropriate for formats where program levels vary widely and/or there is a desire to remove many of the natural dynamics from the program material.

The SST may be defeated by not checking either the Linear or Adaptive options. When defeated the SST does not modify the multiband AGC in any way and its control range is frozen to compensate. When defeated the SST meter will remain at -12dB even though program elements may change.

A Word About Outboard Processors

Several popular outboard broadband and multiband pre-processors on the market have been used in front of “all-in-one” processors to achieve one or all of the following:

- **A different sonic texture than what is available in the main processor.**
- **The ability to bypass functions in the main processor (usually located at a transmitter) and bring the “meat” of the controls to the studio end for endless tweaking**
- **Compensation for a poorly designed AGC in the main processor.**
- **Overdrive protection for an STL path**
- **Encode level enhancement for ratings encoders.**

Field experience has shown that the SST algorithms in AirAura eliminate most of the concerns mentioned above. By controlling the multiband AGC with the SST as we do, the need for a wideband AGC in front of the main processor vanishes. Further, because SST only *manages* the multiband AGC it behaves like a “multiband broadband” AGC if there was such a thing. By *thinking* like a multiband AGC it is not fooled by excessive energy in one part of the spectrum over another. By *operating* like a broadband AGC it maintains a completely natural and balanced sound *regardless* of how much or how little it needs to do. The multiband AGC is always operating in its “Sweet Spot”... thus “Sweet Spot Technology™ ” or “SST.” No other all-in-one processing solution processes with the accuracy of SST.

We believe that the amount of control in our Pro GUI allows the user to virtually set the AirAura to whatever sonic texture they can think of, eliminating the need for an outboard box to emulate the same.

Internet connectivity allows processors in even the most remote locations to be easily adjusted via a remote GUI. In most cases, the need for the processor to be at the studio “…so it can be accessed…” is a thing of the past.

When audio levels in the plant are properly calibrated, there is no need for an AGC in front of an STL path because almost all processional broadcast gear that may be in-line between the console and the STL will have as much if not more headroom than the broadcast console. If non-broadcast or semi-professional gear is in the audio path, it should be replaced. Consumer grade equipment in the program line of a broadcast station is usually not good engineering practice and can spoil the financial investment you have made in AirAura.

There may be some cases where, no matter what, an outboard processor is needed or desired and three of these situations follow.

- **Ratings Encoders** – We’ve found that ratings encoders need a “steady diet” of incoming level in order to be function properly. Improper levels, especially those that are too low, can be problematic for stations that are automated where a human can’t make compensating adjustments to source levels on the console. While SST can easily compensate for audio levels more than 10dB different from source to source, a 10dB drop in level at the input to the ratings encoder (which is prior to the processing) will not keep a ratings encoder happy over the course of a 3-4 minute song. It may even set an audio alarm and email and page the station’s staff.

The BEST way to address the level consistency problem in a facility is to make sure there are standards in place for getting proper levels recorded into the playout system and refrain from using outboard pre-processors. Unfortunately, that isn't always practical, so if a station feels they must use outboard pre-processing ahead of AirAura we suggest setting AirAura's multiband Gate control to a setting that is between -30dB and -40dB, use the *Hold* gating mode, and operate the SST in its Linear mode. If aggressive pre-processing is applied we suggest setting AirAura's AGC Backoff control near 0dB.

- **Reverb** – Reverb was originally used as a way to increase modulation density on older AM transmissions when the audio processing of the day simply wasn't sophisticated enough to do it. The effect of reverb not only did the trick, it also added a pleasant sonic texture to the audio, one still desired by many stations today. The challenge for most reverbs is that variations in their input levels can have a very large effect on how they sound on the air if installed directly into the program chain.

  A pre-processor is traditionally used ahead of a reverb unit which will protect it from generating excessive reverb if (when) its input levels are too high. The use of a pre-processor ahead of a reverb also helps sustain the reverb effect. As in the case of adding pre-processing to boost encoding levels in a Ratings Encoder, the same steps should be taken to adjust AirAura to "mate" with the outboard processor. If reverb is side-chained into a microphone processor and not directly in the audio path, no special adjustments of AirAura need to be made.

- **Analog Telco STL or Aural STL** – Older analog telco lines sometimes need some pre-processing in order to keep signal to noise ratios high. Some users have indicated the need for pre-processing before an aural STL to prevent overshoot. Again, if you feel the need to use a pre-processor in either of these situations, adjustments to the AirAura should be made in concert with the suggestions above.

It is not recommended that SST be defeated in the presence of an outboard AGC. If you do choose to defeat the SST, its influence on the Multiband AGC will become fixed at -12dB and adjustment of the 31-band limiter drives on the FM and HD side may need to be made for any offset that the fixed level creates. It is because of these changes, and because AirAura was designed as a "system," that slowing down SST and the five band AGC/Compressor should be tested first <u>before</u> sections of AirAura are purposely bypassed.

Multiband AGC Controls

MB-AGC Drive

The *MB-AGC Drive* control adjusts the amount of signal at the input to the AGC/Compressor which then determines the depth of gain control (AGC/Compression) achieved. Driving the AGC 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 AGC 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.

AGC Gated 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 *Ooze* is selected and the audio input falls below the AGC Gate Threshold, rather than the bands "holding" their current gains, their gains will slowly increase towards 0dB. The *Ooze* setting is useful for formats playing material with very wide dynamic range or for certain speech-based formats.

When *Hold* is selected and the audio falls below the AGC 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 AGC Gate Threshold again. The Hold mode prevents the AGC from increasing its gain in the absence of audio which minimizes the increase of background noise.

AGC Gate Threshold

The AGC *Gate Thresh* control sets the audio level for when the AGC enters one of the Gated modes explained above. The control can be set to OFF to defeat the Gating entirely, or adjusted over the range of -79.0dB to -20.0dB.

Band Coupling (Inter-band Coupling)

There are two *Coupling* controls associated with the AGC, *High to Mid* and *Low to Mid*. These controls set by how much the outer AGC bands (1, 2 and 4, 5) are permitted to have more gain than Band 3, the reference band. In essence, band Coupling prevents the outer bands from adding gain when the signal in those bands is not sufficient to cause gain reduction. Without the benefit of Coupling, the outer bands could add more gain than the mid band which could bring up noise and allow the audio spectrum to assume an unnatural sounding shape.

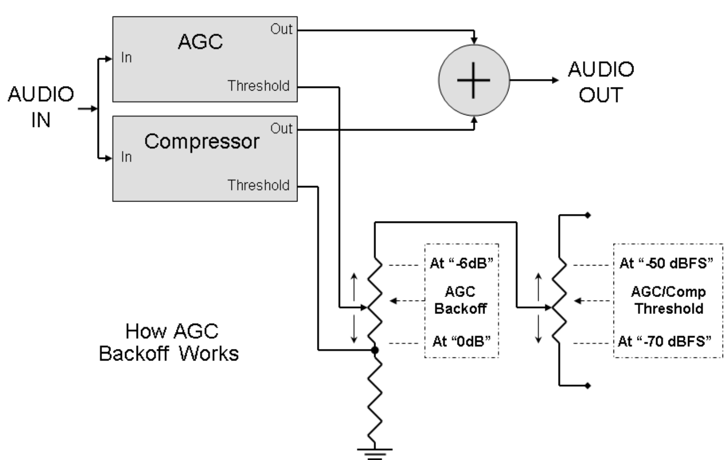
Band 3 was chosen as the "master" band for the high and low coupling control because that band is in the approximate center of our audible hearing spectrum, and coupling the outer bands to that band sounds more natural than any other combination.

The *Coupling* control may be adjusted over the range of "0dB" to "-6dB." When both coupling controls are set to "0dB" bands 1, 2, 4, and 5 cannot have more gain than band 3. 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 the High to Mid and -2.0dB to -6dB for the Low to Mid. Decoupling the AGC's outer frequency bands beyond -6dB can allow their gains to "wander" too much, causing unnatural spectral buildup on some material. Therefore the maximum decoupling is 6dB.

AGC Backoff

The *AGC Backoff* control adjusts the processing balance between what occurs due to slower AGC action and what occurs due to faster and more audible compression. It does this by manipulating the threshold of the AGC relative to that of the compressor. When the *AGC Backoff* control set at "0," the thresholds of the AGC and Compressor are equal, making the combination of their two operations more towards slower AGC and less compression. The schematic-like graphic at left may shed some light on what is going on as the *AGC Backoff* control is adjusted.



Each band has a compression threshold control which affects the operating threshold of the AGC and Compressor and that control is labeled AGC/Comp Threshold in the graphic.

The "output" of each band's threshold control feeds a virtual voltage divider with the *AGC Backoff* control as the top resistor in the divider. As the *AGC Backoff* control is adjusted away from the "0dB" position (where AGC and Compressor thresholds are equal) the AGC's Threshold is raised compared to the Compressor's. The effect? The AGC starts doing less work and the Compressor does more. The further the *AGC Backoff* control is adjusted away from "0dB" the more the average levels and detail increase, and because the AGC/Compressor operate in the Sum and Difference signals (L+R/L-R) the perceived stereo sound field also seems to widen.

It is important to understand that the Compressor may never have more gain than the AGC. The Compressor always starts its gain reduction journey *from* whatever the current AGC gain reduction level is, and when the Compressor releases, it can only release *back* to whatever the AGC gain reduction value is at that moment. This means that *most* of the gain control inside the AGC/SST section occurs via slower AGC action, not faster Compression, which dramatically reduces certain distortions caused by fast-acting gain controllers, and without losing the benefits of faster gain control when needed.

When the *AGC Backoff* control is set all the way to the -6dB setting much of the gain control is due to the faster Compressor action and short term dynamics will be aggressively controlled. What this means is that it is the setting of the *AGC Backoff* control alone that primary governs the "sound" of the audio being processed as far as 'hearing it working' is concerned. The *AGC Backoff* control is therefore the "Compression Texture" control since the more it is turned away from "0," the more audible the processing becomes.

This single control, *AGC Backoff*, allows processing texture to be highly modified from a gentle, unobtrusive and invisible control (Backoff between "0dB" and about minus 2.0dB), to more obvious compression (Backoff control at "-3.0dB") or even greater that sounds similar to, but is cleaner than, the vintage compressors and limiters from the 1960's.

Note that regardless of the setting of the *AGC Backoff* control, the AGC and Compressor stages operate with their algorithms linked mathematically. When less control from the AGC is desired, simply reduce the compression ratios slightly or lengthen the AGC attack times.

Stereo Enhancement

AirAura contains a multi-pronged stereo enhancement technique that has proven to be very effective in the field. This approach has a side benefit in that it minimizes receivers' reactions to multipath interference without degrading the audible stereo separation. This multi-pronged approach includes:

- Separate AGC Sum and Difference (L+R/L-R) mix levels.
- Overall post-AGC L-R signal level manipulation.
- Automatic Multipath Limiter incorporated into the stereo encoder.

The first control is actually a set of controls within the *L-R Mix* tab on the *Multiband AGC with SST* screen and on the left hand side of the graphical area. Each of the five AGC band's output mix levels may be adjusted via the *L+R Mix* and *L-R Mix* tabs.

On the *L-R Mix* tab a blue dot, one for each AGC band, can adjust the output level of the L-R processing section by +/-6dB.

On the main page of the *Multiband AGC with SST* screen and located below the *AGC Backoff* control is a *Stereo Enhance* control which adjusts the overall output level of the L-R processing section on a broadband basis (all five bands together).

It may now be obvious that the *L-R Mix* and *Stereo Enhance* controls can be used in combination to arrive at virtually *any* style of stereo enhancement. For instance, after passing through a codec many program sources contain extra midrange energy in the L-R and because there is less audio energy in that signal to mask such artifacts they can become more audible in stereo. By slightly reducing the Band 3 and Band 4 L-R Mix settings (perhaps 1dB to 3dB) these artifacts can be smoothed out and made less obvious.

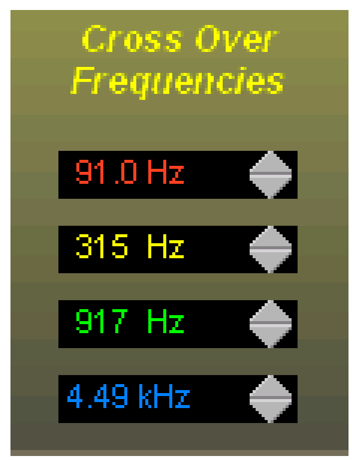
To further mask such artifacts and/or to make the stereo sound field big, wide, *and* warm, one might slightly reduce the Band 3 and Band 4 L-R Mix levels while also slightly increasing the L-R Band 2 Mix level. The Band 3 Band 4 settings help subdue codec artifacts while the increase in lower midrange energy provides a comfortably wide sound. Because high L-R frequencies aren't also being enhanced, the susceptibility of receivers to the blending artifacts during multipath is reduced, giving the audience in questionable coverage areas a better listening experience.

The *L-R Mix* and *Stereo Enhance* controls each have a +/-6dB control range. Because they work in concert, they add or subtract to each other's settings. For instance, if the *Stereo Enhance* control is set to +2dB and, in the L-R mixer band 4 is set for +1, a total of 3dB of enhancement occurs in Band 4. Conversely, if the *Stereo Enhance* control is set for +2dB and the Band 5 L-R Mixer is set at -3dB, the effective enhancement in band 5 is actually -1dB. (2dB + -3dB = -1dB).

Makeup Gain

The *Makeup Gain* control adjusts the output level of the AGC/Compressor into the following limiter stages. It simultaneously adjusts the Sum and Difference channels. Any AGC/Compressor which is a feed-forward design, such as that in AirAura, requires the gain removed by compression to be "made up" afterwards to return the audio back to normal levels. For example, if the AGC is performing 20dB of gain reduction the *Makeup Gain* control should be set to +30dB (+20dB to make up the gain reduction plus another 10dB for headroom allowance). The *Makeup Gain* control range is (+24.0dB to +36.0dB).

Crossover Frequencies



The AGC/Compressor is a five band design utilizing 48dB/octave 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 Ranges

Band 1 to Band 2 - Super-Low to Low:	20Hz – 149Hz
Band 2 to Band 3 - Low to Mid:	162Hz – 687Hz
Band 3 to Band 4 - Mid to Presence:	728Hz – 2,180Hz
Band 4 to Band 5 - Presence to High:	2,590Hz – 20,000 Hz

Note that although the bands cannot be made to overlap, the crossover can be operated as a three, four or five band AGC as follows:

For Three Band Mode: Move the BAND 1<>2 crossover down to 20Hz

AND

Move the BAND 4<>5 crossover up to 20,000Hz

Three band mode is suggested for Sports and Classical formats. In Sports formats, especially those that carry a lot of live events, the three band mode helps prevent unnatural crowd noise. In Classical formats, the three band mode allows the AGC and Compressor to sound natural by minimizing spectral rebalancing, keeping the spectral balance true to the original source.

For Four Band Mode: Move the BAND 1<>2 cross-over down to 20Hz

<u>OR</u>

Move the BAND 4<>5 crossover up to 20,0000Hz

In the four band mode, either Band 1 or Band 5 is tuned beyond the range of program material. Formats desiring a very smooth and mellow high end might see some benefit in moving Band 5 up out of the way so that very high frequencies never become exaggerated by the AGC. Formats that don't need extra bass support, or talk formats where studio acoustics can cause unnatural bass buildup making voices sound tubby, can take advantage of the flexible crossover design to move Band 1 down out of the way so that very deep bass resonances caused by such room acoustics are never exaggerated.

For Five Band Mode: Make sure BAND 1<>2 crossover is at least at 40Hz or higher

<u>AND</u>

Make sure BAND 4<>5 crossover is at or lower than 12,000Hz

Because of the intelligence of AirAura's AGC/SST the full five band mode should be used in most situations and is even perfectly fine for many talk/classical based formats. Only those formats where special needs are required should consider using the three or four band mode.

AGC/Compressor Sidebar Menu

In Vorsis processors the controls for all processing bands may be adjusted by first selecting the desired control type tab, and then using the blue dot navigation method covered previously to adjust individual band parameters. For more information on how to adjust these controls, please see the "Blue Dots" section of this manual.

Threshold

The *Thresh* control determines at what level processing will begin to take effect in a particular AGC 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 AirAura's AGC Threshold controls is -70 to -50 **<u>dBFS</u>**.

Before we venture further, it is important to point out that *most* of AirAura's controls are calibrated in dBFS in order to show their true relationship to digital signal levels and their relationship to "0dBFS." "0dBFS" is when there are no more digital bits available and severe distortion results. With the exception of the MPX Output level controls this is how external levels in and out of AirAura are also calibrated.

Now, back to the subject of AGC Thresholds…

Sometimes, Threshold settings are seen down around -60dB in some of our factory presets. They might *look* like very low threshold settings but in fact they aren't. The -60dB Threshold setting is actually -60dBFS, and that's for "peak" levels, so consider this; if we leave the standard 20dB of headroom below 0dBFS and fudge that factor by another 20dB to make sure there is enough peak headroom for *any* possible scenario coming in from the studio, we're now down at -40dBFS. Now, if we're processing for an aggressive

format and want the AGC to have at least 20dB of level makeup available in the AGC for bringing up the low stuff, we'll need a threshold setting that is another 20dB below that … now we're at -60dBFS. A -60dBFS threshold actually *does* equal the usual compressor threshold settings in traditional compressors and AGC devices.

AGC Attack

The *AGC Atk* control determines how quickly the AGC will respond to upward level changes in the audio. The control range is 50ms to 500ms. A setting of 300ms corresponds well to our perception of average loudness and is therefore a very good starting point.

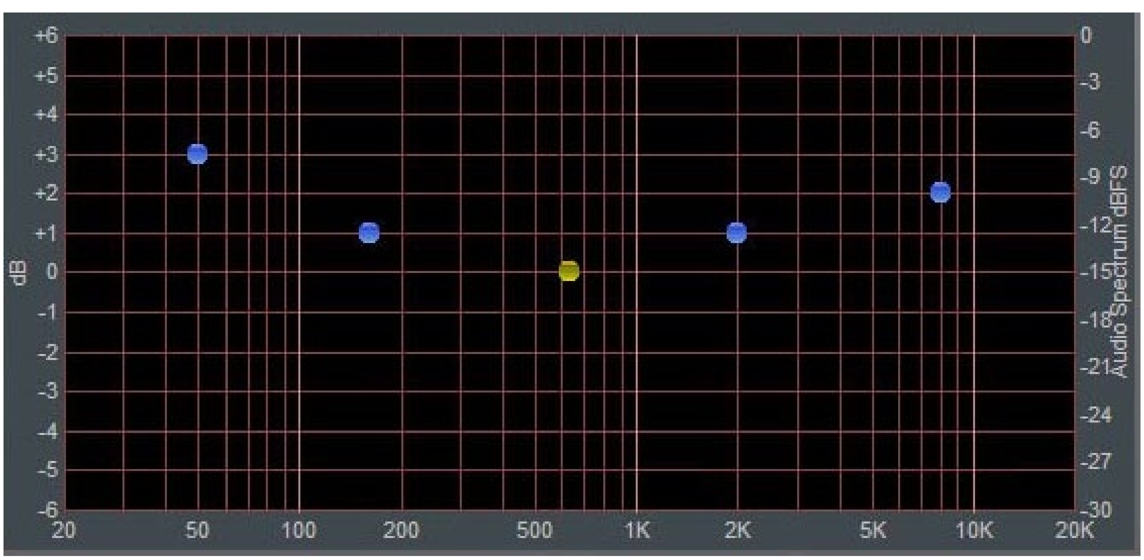
AGC Release

The *AGC Rls* control determines the rate at which the AGC will increase gain when the audio levels fall, as long as they are still above the Gate Threshold setting discussed previously. The adjustment range of the control is from 1.0 seconds to 7.0 seconds. The 7 second setting is considered VERY slow and is recommended for fine-arts or classical programming (or, if you're Jeff Keith, for Classic Rock processing when he wants the on-air dynamic range to mimic the old AOR formats). The 1 second setting is very fast and is recommended only if a certain "pumpy" texture is desired. The recommended setting for most formats is between 2.5 seconds and 6 seconds.

Gate Threshold

The AGC *Gate Thresh* control sets the audio level below which the audio gain reduction values are held until audio returns. This control can be set to OFF or adjusted over the range of -79.0dB to -20.0dB. In each of the five bands there is also a Gate Threshold Trim control allowing offsets of +/-6dB to be made to each band referenced to the main Gate Threshold setting. This allows precise Gate Thresholds on a band by band basis as appropriate for the program format requirements.

Example: Suppose the AGC Gate Threshold control is set to -40.0dB and the controls for the individual bands are set as in the example below:



In this example, Band 1's Gate Threshold has been offset by +3db, which means its actual gate threshold is now -37dB, or 3dB less sensitive than band 3 (which is still set at the "0" reference line). This will cause band 1 to need a higher audio level before it will "un-gate" and allow the gain to increase. Setting the gate this way prevents unnatural increases in low frequency gain that might be undesirable. The Gate Thresholds may be offset this way in any of the bands as the application warrants.

Compressor Attack

The *Compressor Atk* control determines how quickly the Compressor responds to rising audio levels not caught by the AGC because of its longer attack time. The compressor's purpose is to handle medium term peaks only, and by doing so it completely frees the AGC from having to worry about controlling them. The recommended Attack time setting for sounding "musical" is between 10 and 30ms. Compressor settings faster than 10ms can make the audio sound constricted, especially with Ratio settings above about 4:1. Conversely, settings slower than 30-50ms can prove to be ineffective on certain percussive material and even though it is essentially inaudible while doing so, it can make the 31 Band limiter work harder than it needs to.

Compressor Release

The *Compressor Rls* control determines how quickly the compressor returns gain to the processing "floor" established by the AGC's current value of gain reduction. As mentioned before, the 5 Band AGC and establishes the platform gain reduction from which the compressor operates. It accomplishes what adding a compressor after the AGC would accomplish, but without needing it. The result is cleaner and more detailed audio without the compression artifacts that can creep into aggressive processing or multi-box solutions when ultimate loudness is the goal.

The Compressor Release time's adjustment range is 50ms to 1 second (1000ms). The best starting point is usually to set the *release time* to the *attack time* of the AGC. Thus, if the AGC attack time is 300ms, the Compressor release time would be 300ms. This is not a rule, but a good starting point.

Ratio (Compression Ratio)

The *Comp Ratio* determines the how tightly the multiband AGC controls the audio. The range of adjustment is from 1.0:1 to 20:1, with 1:1 being "no control," to 20:1 which behaves like a leveler or very slow limiter. The Compression Ratio for each band may be set individually, and the Ratio settings apply to both the AGC and Compressor simultaneously.

Compression Ratio is an easy term to explain. As above, if the Ratio is 1:1 it means there is no audio control because the Output increases 1dB for every 1dB of input increase. Similarly a ratio setting of 3:1 means that for every 3dB increase in input level, the output level increases by only 1dB. At a ratio of 20:1, it would take an input level increase of 20dB (almost the entire scale of a standard VU meter) to make the input level rise by only 1dB. This is very tight control, such as what a limiter might have.

Sensible Compression Ratio settings result in very natural gain control; ratios between about 2:1 and 6:1, depending on format, with more aggressive formats sounding more consistent if ratio settings lean toward the latter value.

L + R Mix(er)

The *L+R Mix* may be used to tailor the mix levels of the L+R, or "mono" content of the processed signal. Very dramatic EQ changes can also be made using these controls. Generally we recommend that the L+R Mix controls be set flat and the parametric EQ used for EQ tailoring. If more than the parametric EQ is required, *small changes* can be made here if needed, +/-3dB at most, for best on-air consistency.

L – R Mix(er)

We'll cover this a little more again here even though much of it was already covered in the section on Stereo Enhancement.

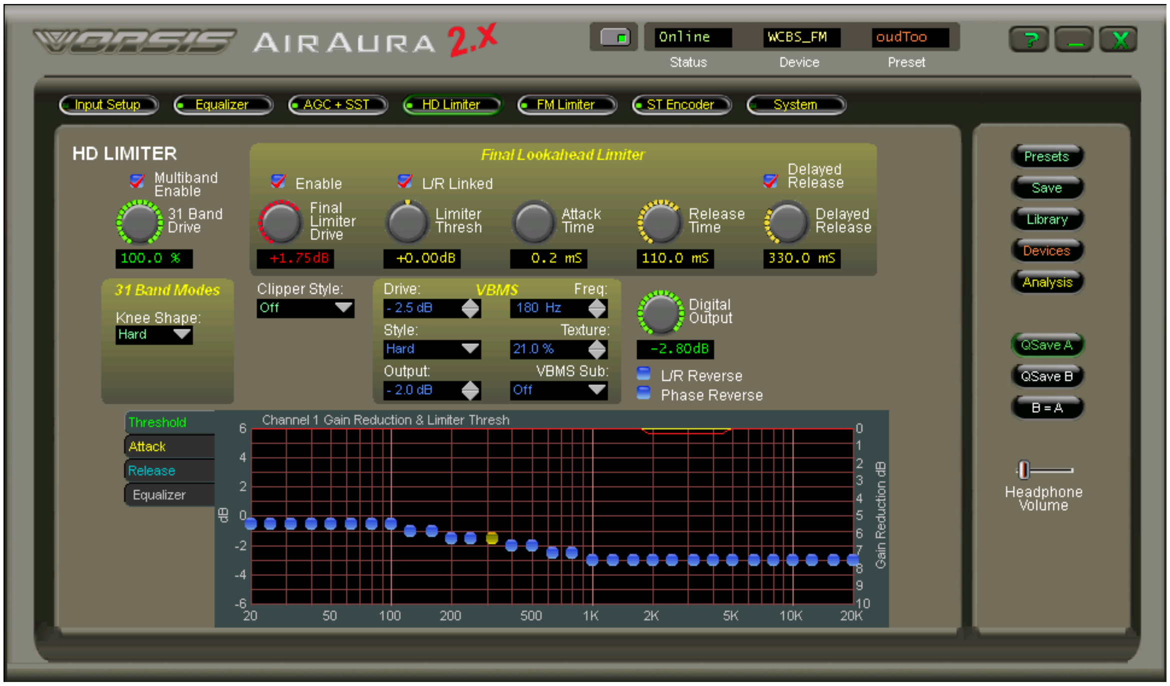
The *L-R Mix* controls the difference, or "stereo" content of the processed signal. By carefully using these controls custom stereo enhancement effects may be added to specific parts of the audio spectrum *without* raising the overall L-R gain and without increasing the potential for multipath interference.

To use this control to its full benefit, the *Stereo Enhance* control (explained earlier) should be set at 0dB and adjustments made in this section to add and/or tailor the sound of stereo enhancement. The most dramatic changes will be heard in Bands 2, 3 & 4 and this is somewhat dependent on the AGC's crossover frequencies. A good 'medium' setting for stereo enhancement would be to raise the L-R Mixer settings on Bands 2 and 3 to perhaps +3 dB and Band 4 to perhaps +2dB.

If the Stereo Enhance control is set above "0" then its setting will need to be added to the settings for each band of the L-R mixer to determine the total enhancement in effect. For instance, if the Stereo Enhance control is at +2dB and Band 3 is at +3dB, there is then +5dB of enhancement in Band 3.

HD LIMITER MENU

The 31-band *HD Limiter* screen contains all the controls necessary for fine tuning the sound of the HD output while constraining its absolute peak output level into the next device in the chain.



HD Limiter Screen

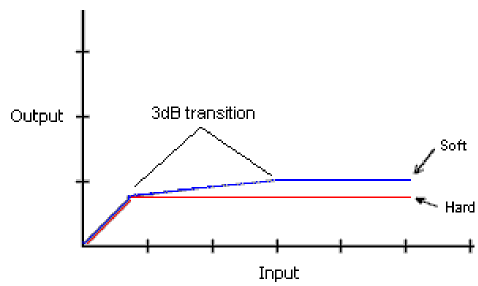
31 Band Drive

The *31 Band Drive* control determines the amount of input signal being applied to the 31-band HD limiter. Lower drive settings create a more gentle sound while still controlling peaks while higher settings can make the sound more aggressive by increased short term density. Extremely high drive settings which cause more than 4dB to 6dB of frequent gain reduction are wholly unnecessary, can become fatiguing, and may create a less than desirable operating condition for a perceptual codec such as the one used in HD transmission. The *31 Band Drive* is adjustable from 80 to 100% and is calibrated in percent, not dBFS, because its setting is also dependent on the setting of the AGC *Makeup Gain* control covered previously.

31 Band Modes

There are two Mode settings available, Soft and Hard, which adjust the 31-band limiter's transfer function. In each mode the behavior of the threshold circuits is modified to alter how the limiters behave once their limit thresholds are reached. In the *Hard* mode infinite ratio limiting begins exactly at the value set by the Limiter Threshold controls. In the *Soft* mode, limiting begins to occur at a softer ratio 3dB below the value set by the limiter controls, and then gradually transitions into infinite ratio limiting over a 3dB range.

We recommend the Hard setting if the HD output is used to feed a perceptual codec. This is because the codec responds more favorably to the type of peak control exercised by the 31-band limiter when operating in this mode.



Comparison of Hard and Soft Limiter Modes

The *Soft* setting is gentler sounding, with gain reduction beginning to take place 3dB below a band's actual threshold setting with limiting gradually getting harder until it reaches infinite ratio. The curve at left illustrates this behavior.

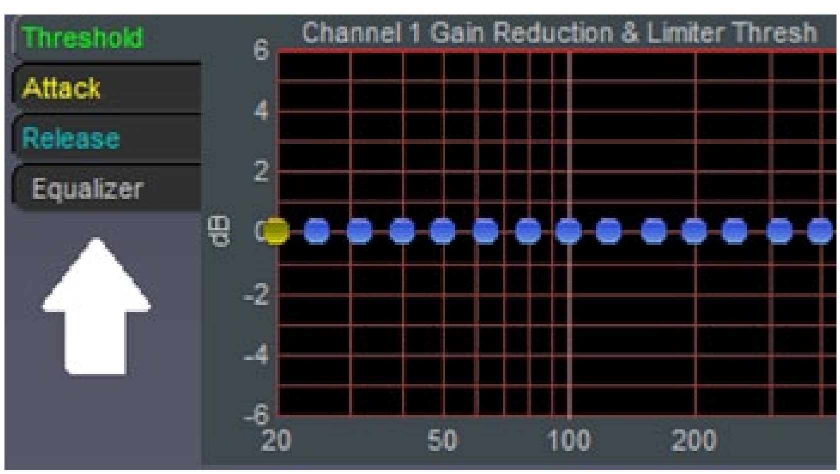
Because the Soft setting begins having a contribution to the gain 3dB below where the knee becomes infinite, this setting is intentionally more audible in its action. Because it creates softer-sounding peak control, it may be desirable for certain program formats. While we recommend the Hard mode for use with perceptual codecs, it's always best to choose the mode that sounds best in *your* application.

As mentioned previously, the Soft setting is intentionally more obvious in its action and if desired, could be used to generate a lot of extra RMS density (loudness). This is because the more time the audio spends within a nonlinear transfer function, the higher the RMS energy generated. The Hard setting is less obtrusive in its action; in fact it's essentially invisible because peaks within individual frequency bands spend more time below threshold than above. Further, the bands are so narrow that our ears do not perceive the gain control occurring within a single band, mainly because that limiting is masked by audio present in adjacent limiter bands that is not undergoing limiting.

Feel free to see how each setting sounds and how it can help your station achieve the sound it desires, keeping in mind that we recommend the Hard setting for perceptual codecs.

31 Band Limiter Sidebar

Like the five band AGC/Compressor, the 31-band limiter graphical area hosts a sidebar with adjustment parameter tabs for gaining access to the controls for tailoring the sound of the 31-band limiter. Remember the previously described method for adjusting these controls (the blue dots) because the same method explained there is also used here.



Threshold

The Threshold controls determine at what audio level limiting will begin in a particular band. The AirAura 31-band limiter threshold controls are adjustable over a 12dB range. The absolute threshold of each limiter band is determined not only by the settings here, but also by the Multiband Mode (“Soft” or “Hard”) as explained above.

Attack

These controls determine how quickly the Limiter bands respond to audio peaks once they exceed the limiter threshold. Recommended settings are between 30ms and 10ms. Longer attack times are possible, and even recommended, for the limiter bands that fall below the setting of the VBMS Frequency control since doing so will assist with bass “punch.” Overall, longer attack times will cause the Lookahead Limiter to work harder.

Release

The release time controls determine how quickly the limiters will return to their 0dB resting values once audio has fallen back below the limiter threshold. Faster release times will yield a louder sound while slower times will yield a smoother sound at the expense of some loudness. Very fast release times, faster than the attack times (!), allow the freedom to create very controlled intermodulation “texture” and “slam,” and only in certain spectral ranges, when desired

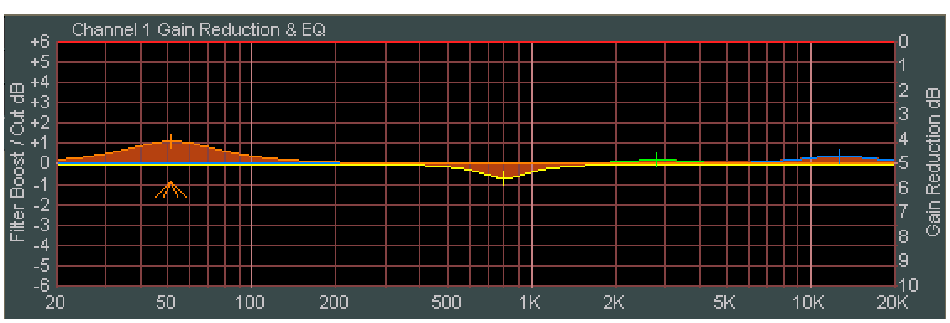
Because of the way the 31-band limiter works, there is a lot of leeway in adjusting the attack and release times and the ranges of the controls have been limited to prevent “driving off a cliff” though using nonsensical settings in any audio processor can yield unacceptable sound.

Equalizer

This EQ is a four-band parametric like the one near the AGC section and is situated in front of the HD section’s lookahead limiter. EQ applied here is akin to adjusting a final mixer at the outputs of the 31-band limiters. The adjustment ranges have been made large enough to create the desired effect but limited in scope to prevent overuse.

The recommended settings in this equalizer section are flat because any EQ changes *should* have been made upstream in the AGC L + R mixer or in the parametric EQ. If a LITTLE more of something is needed, these controls have been provided to accomplish that. But they are also easily abused because of where this equalizer is located. Boost and Cut are limited to +/-3dB.

Adjusting this final EQ is the same as adjusting the Parametric EQ by directly manipulating the curves.



Double clicking on the little crosshair at the top (or bottom if it’s below the line) of the curve turns that section on and off.

Clicking on the crosshair and holding the left mouse button down as the mouse is dragged moves the curve to a new frequency (mouse drag left or right) as well as adjusting the curve's height above or below the 0dB reference line (mouse drag up or down).

Clicking on the "tent" underneath the curve and holding the left mouse button down, allows the bandwidth, or "Q," of the equalizer to be changes by sliding the mouse left or right.

Clipper Style

These clippers are provided mainly for applications that do not involve the HD perceptual codec. If this feature is in use and the outputs are feeding an audio codec, great care is required in adjusting the depth of clipping, the clipping style, and the audio signal density, to mitigate unavoidable and undesirable artifacts! Again – this clipper is supplied solely for use in creating a specific sonic effect that cannot be achieved via other means.

There are three choices available in the HD Clipper Style list box.

Off

Bypasses the clipper altogether. This setting is highly recommended when AirAura's HD output is being used with a perceptual bit-reducing codec such as the one used for HD Radio.

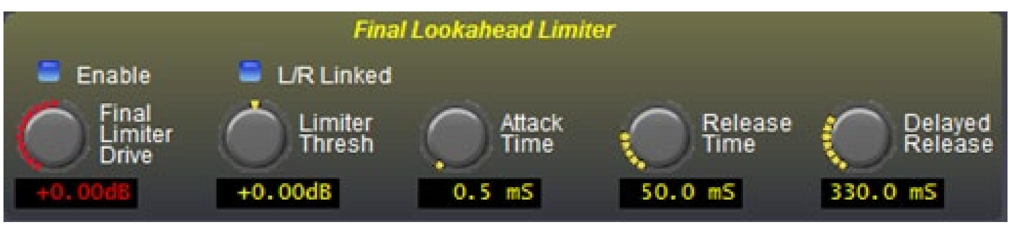
Round

The Round style adds a bit of "roundness" and a small amount of "edge" to the audio without sounding harsh or outright distorted.

Deep

The Deep style sounds "warmer" and "bigger" than the Round style because it is allowed to generate more distortion harmonics.

Lookahead Limiter



The HD section's final peak control is accomplished by a special high-speed feed forward limiter with a 0.5 millisecond lookahead time. The control path, not the audio path, is oversampled in order to achieve extremely precise peak control. Because the goal of oversampling is better peak control through finer definition of the audio waveform, oversampling of the audio signal itself is not only unnecessary, but wastes DSP resources.

Limiter Enable

When the *Enable* checkbox is checked the Lookahead limiter is enabled. When the Lookahead Limiter is enabled all of its controls are functional and will have an impact on the sound.

L/R Linked

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.

Final Limiter Drive

The *Final Limiter Drive* control sets how deep limiting will be for normal program material.

Limiter Threshold

The *Limiter Thresh* control adjusts the threshold of the Lookahead Limiter relative to the setting of the *Final Limiter Drive* control. It is normally adjusted until the Lookahead Limiter gain reduction bargraph *just* shows the onset of limiting with normal program material. The Lookahead Limiter threshold may be adjusted over a +/- 6dB range.

Attack Time

The Lookahead Limiter *Attack Time* control sets the attack time of the Lookahead limiter, or how fast it reacts to and then controls signal peaks. Limiter attack times from 0.2 milliseconds through 0.5 milliseconds allow the limiter to "lookahead" in time and react to signal peaks before they arrive at the limiter's gain control algorithm. This results in a "prediction limiter" that prevents output peaks before they can occur. At *Attack Time* settings of 0.5 milliseconds and higher the lookahead has no effect and the limiter then operates conventionally.

NOTE: The 0.5 millisecond look ahead is always active. If the *Attack Time* control is set at 1.0 millisecond the actual attack time is 0.5 milliseconds. The lookahead value of 0.5mS must be subtracted from the control setting to know the actual attack time.

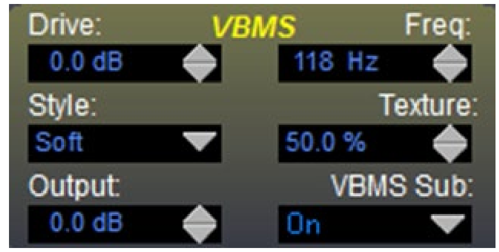
Release Time

The *Release Time* control sets the primary (fast) recovery time of the lookahead limiter. Slower settings (longer release times) cause the peak limiter to recover more slowly from limiting. Release times set too slow may allow undesirable signal ducking to occur on certain program material. Fast release times increase signal density and decrease limiter audibility – up to a point. Release times that are too fast may increase intermodulation distortion. The control range is 33.0 to 330.0 milliseconds.

Delayed Release

The *Delayed Release* control sets the secondary (slow) recovery of the look ahead limiter. When Delayed Release is engaged, the limiter will release the first 3dB of gain reduction at the (faster) Release setting, with the remainder of the Release occurring at the Delayed Release timing. Using Delayed Release drastically reduces intermodulation distortion and allows more limiting depth without incurring a distortion penalty. The control range is 100ms to 1.0 seconds.

VBMS (Vorsis Bass Management System)



The Vorsis Bass Management System allows extremely low program frequencies to be processed without causing dynamic gain changes and without generating undesirable intermodulation artifacts in higher frequency signals. Its primary purpose is to allow the thumps of kick drums and other percussion instruments to sound natural in their "percussiveness" without pumping or other undesired artifacts. The algorithm operates in left/right mode treating both channels identically.

Our laboratory research on codec behavior rather surprisingly revealed that it is perfectly acceptable to process very low frequency material using the VBMS algorithm provided that the harmonics generated during the process are well controlled spectrally. Since our bass algorithm does not allow significant harmonic levels above the third or fifth (somewhat dependent the VBMS Style setting) there is no audible downside to using it with coded audio. Furthermore, the ability to utilize the VBMS on both the FM and HD sides of the processor allows the sound to be more evenly matched as receiver-based digital to analog crossfades are encountered.

Drive

This control adjusts the amount of Input Drive to the VBMS algorithm. Higher drive settings make it work harder and generate higher levels of harmonics. The *Drive* control adjusts the amount of drive to the VBMS and is adjustable over a range of +/- 10dB. Adjustment is best done by ear with percussive source material.

Style

Two VBMS operating styles, plus Off, are available. The *Hard* setting has a transfer function occupying approximately 1dB of transition amplitude. The waveform control is fairly abrupt and generates low order harmonics.

The *Soft* setting operates with a transfer function occupying approximately 2dB of transition amplitude. Therefore the control is less sharp than the hard setting, generating more low order harmonics. While the two settings may appear to be almost the same, the 1dB difference in transfer ratio creates a *significant* difference in how the two VBMS styles *sound* on actual program material.

Output

The VBMS *Output* control adjusts the output of the VBMS algorithm and can be thought of as a "low bass mix" control. Settings between -6dB and -3.0dB add extra bass to the overall mix, while settings lower than -6.0dB reduce it. Both the VBMS *Drive* and the VBMS *Output* controls are best adjusted by ear using percussive bass material. Care should be exercised in the setting of the VBMS Output control because settings above +0dB may cause bass to reach the main clipper which will add distortion. Like VBMS Drive, the VBMS Output is adjustable over a range of +/- 6dB.

VBMS Frequency

The VBMS *Freq* control tunes the VBMS algorithm's frequency of operation from 60Hz to 300Hz. Program energy not passing through the algorithm is time aligned for coincidence regardless of the VBMS settings. VBMS operates *only* on frequencies *at and below* the VBMS Frequency setting.

Texture

The *Texture* control sets the ratio of odd to even harmonics in the VBMS circuit. Normally the VBMS adds only odd harmonics so some mechanism is needed to add some even harmonics too so that smaller speakers seem to have better bass than they are physically capable of reproducing. This is what the VBMS *Texture* control does. Higher settings are recommended for rock/classic hits based formats while Urban, R&B and many CHR formats may wish to operate with the control set lower. Note that settings above about 35% may sound unnatural on some program content.

Sub Bass

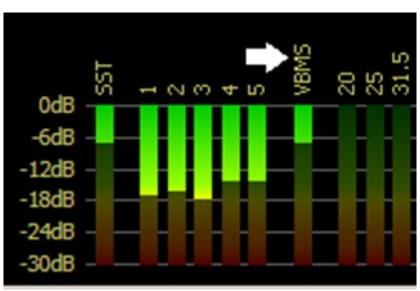
The *VBMS Sub* Bass control has an On/Off function that enables and disables a dynamic low frequency extension circuit that enhances the presence of subsonic energy like a subwoofer does and is useful for formats that require extra deep bass, such as Urban and CHR.

NOTE #1: We **do not** recommend *VBMS Sub* being enabled if substantial bass equalization has been dialed in using the parametric EQ or the L+R EQ mixer in the AGC/Compressor. We also recommend listening to the station with an audio system *capable* of reproducing sub bass frequencies while experimenting with *VBMS Sub* because its effect will be *essentially inaudible* on systems that are incapable of reproducing very low frequencies.

NOTE #2: When *VBMS Sub* is enabled we recommend operating the High Pass Filter (Input Menu) at settings no lower than 30Hz. In fact slightly higher settings, such as 40Hz, might be more appropriate, depending on the format. Failure to heed this advice may allow undesirable subsonic signals to get on the air.

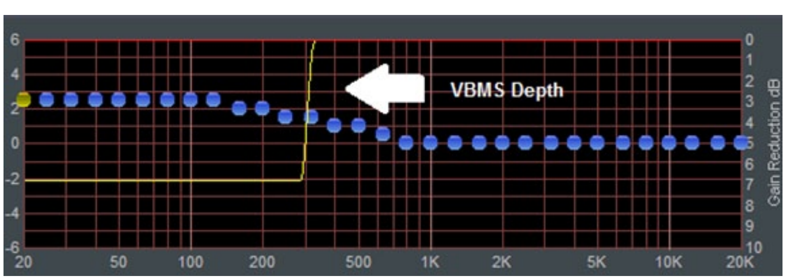
NOTE #3: Remember that the majority of consumer sound systems cannot reproduce audio signals below about 40-50Hz. Exceptions to this are systems specifically equipped with subwoofers and/or large air mass speaker systems. It is important to recognize the capabilities of the station's typical listeners and whether or not they might be equipped with audio systems capable of reproducing sub bass. This of course suggests a highly format dependent situation!

VBMS Metering



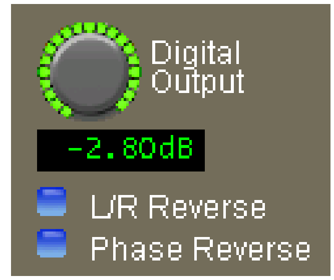
There are two ways to monitor the amount of processing being done by the VBMS

VBMS can be seen on the *Analysis* Screen on the Pro GUI, in the Guru GUI, or on the front right screen panel of the AirAura. VBMS can also been seen in Pro GUI in the Limiter gain reduction graph. Its activity is indicated by the yellow line.



Depending on how hard the VBMS is driven, one may or may not observe much activity on the VBMS metering. In fact, only when a lot of bass is present in source material will VBMS show any activity on the meters, however this does not mean VBMS is not working.

Digital Output



The *Digital Output* control sets the audio level at the Digital and Analog outputs of the HD processing in AirAura. There is no further processing after this point. The control range is -79.95dBFS to 0dBFS with 0.05dB increments allowing very fine settings of the maximum peak level into the next piece of equipment. Muting of the HD outputs will occur when this control is dialed fully counterclockwise and the display shows “Off.”

L/R Reverse

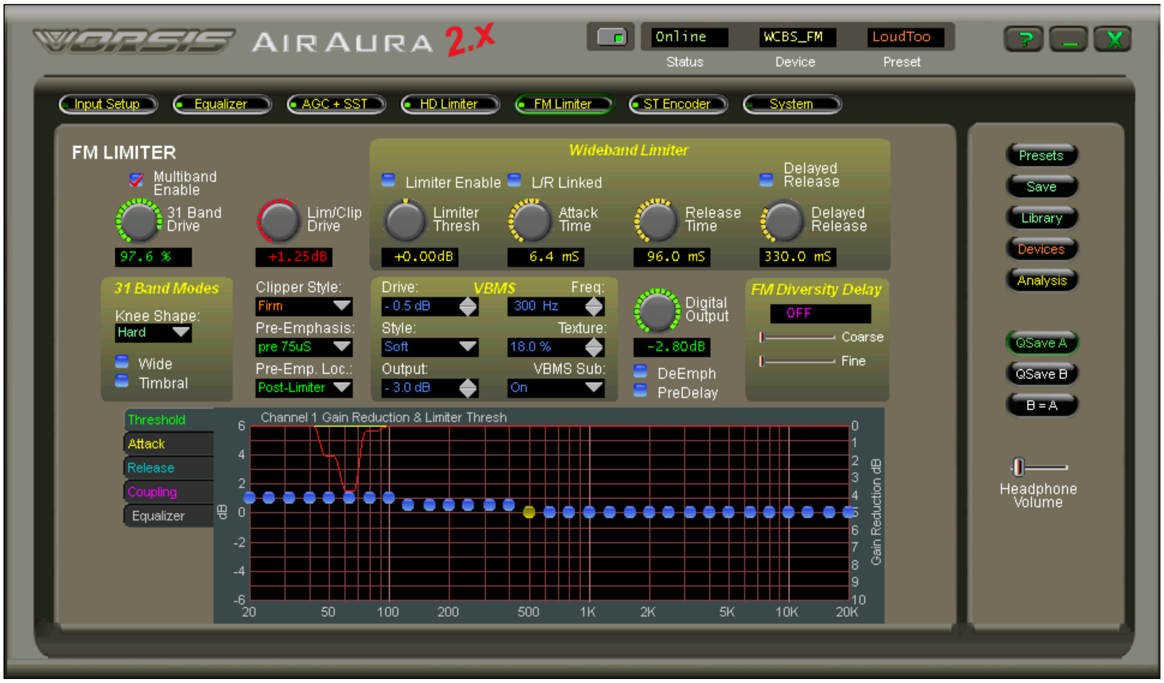
The *L/R Reverse* checkbox allows the left and right channels of the Digital and Analog outputs to be swapped (left becomes right, right becomes left) which can be useful during troubleshooting equipment that is external to the AirAura.

Phase Reverse

Certain software versions within the iBiquity HD Radio system have an inadvertent signal polarity inversion putting the analog and digital signals 180 degrees out of phase resulting in audible flanging as the two signals crossfade in a listener’s HD receiver. When the *Phase Reverse* checkbox is checked, it flips the polarity of the HD Radio AES/EBU output signal by the 180 degrees to compensate for this anomaly.

FM LIMITER MENU

The 31-band *FM Limiter* screen contains all the controls necessary for fine tuning the sound of the FM output and adjusting its peak control method. These controls allow the user to make many tradeoffs between loud and clean, in ways that have never been available before in an FM peak limiter.



FM Limiter Screen

31 Band Drive

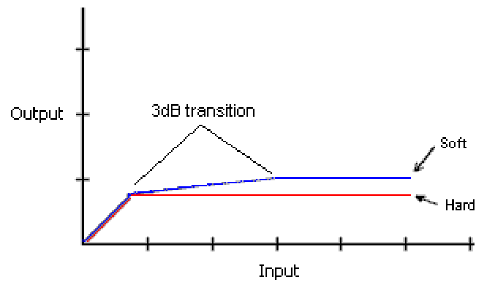
The *31 Band Drive* control determines the amount of input signal applied to the 31-band limiter – less drive creates a more gentle sound while still controlling peaks while higher settings increase short term density and loudness. Note that unlike limiters with fewer bands, the 31-band limiter *does not* require 10-15dB of continuous peak limiting to do its work. In fact, *31 Band Drive* control settings that result in more than 4-6 dB of limiting are not only unnecessary, but serve no purpose. The *31 Band Drive* control is adjustable from 80 to 100%.

31 Band Modes

The 31-band Limiter may be operated in one of two modes; Soft and Hard. Each mode applies a slightly different transfer function to all 31 bands which then modifies how they behave at the threshold of limiting.

In the *Soft* mode, limiting begins at 3dB below a band's actual threshold setting, and as the audio level continues to increase the limiter transitions into a hard limit ratio over a 3dB window. The curve below illustrates this behavior.

Because the Soft setting begins having a contribution to the gain 3dB below where the knee becomes hard, this setting is intentionally more audible in its action and creates a softer sounding limiting that might be desirable for some program formats.



Comparison of Hard and Soft Limiter Modes

The *Hard* mode is the most abrupt transfer function with no gain control occurring until a band's set limit threshold is reached. The curve on the left illustrates this behavior. Notice that once the threshold is reached (depicted by the rising line terminating at the flat top) there is no increase in level and therefore the compression ratio of the limiters is extremely high.

As mentioned previously, the Soft setting is intentionally more obvious in its action and if desired, could be used to generate a lot of extra RMS density (loudness). This is because the more time the audio spends within the limiter's nonlinear transfer function, the higher the RMS energy generated. The Hard setting is less obtrusive in its action because peaks within individual frequency bands spend more time below threshold. Because limiting does not occur until the thresholds are reached, the multiband limiter has no contribution to the sound until the audio is above threshold. Once above threshold the limiters operate with infinite ratios and the audio is truly "limited."

Both limiter settings have their purpose, and there is no right or wrong setting. Please feel free to experiment with each mode to see how each sounds and how it can help your station achieve the on-air sound it desires.

Special 31 Band FM Limiter Operating Modes – Wide and Timbral

The 31-band FM limiter operation varies according to the settings of the *Wide* and *Timbral* checkboxes. When these boxes are not checked each band of the 31-band limiter operates in its Discrete mode where the operation of each limiter band is completely independent of all the others.

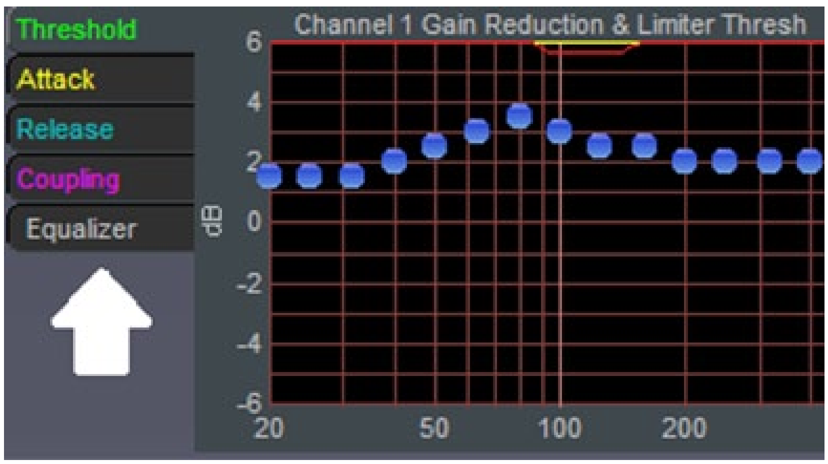
If the *Wide* mode checkbox is checked the 31-band limiter operates as a pseudo-10-band limiter with its bands ganged together in what we call "Triplet Coupling." In this mode the limiter operates as ten "center bands" by dividing the limiter bands into groups of three. For instance, Band 1 becomes coupled to Band 2 and Band 3, while Band 4 becomes coupled to Band 5 and Band 6, and so on across the audio spectrum. This mode is useful when one wants to "hear" the limiters working in order to create a particular sonic effect. Because the 31 individual bands are so narrow, when the limiters are operating in the Discrete mode it is quite difficult if not impossible to hear individual limiter bands work.

When the *Timbral* checkbox is checked the limiters will operate in a special 'mathematically coupled' mode. Here, analysis of the incoming audio and the behavior of individual limiter bands cause them to operate in an extremely complex way. When limiting occurs at a fundamental frequency, the limiter algorithm calculates what the second and third harmonics of that frequency are and causes the limiter bands on or near those harmonic frequencies to perform the same amount of gain reduction as that occurring at the fundamental. The purpose of this mode, and the end result of it, is the maintenance of a very accurate timbral balance as the limiters operate. This mode has all the advantages of a multiband limiter while from a spectral balance standpoint it sounds like a broadband limiter (or no limiter at all). This mode is especially useful in Jazz and Classical formats because the gain at the upper harmonics of solo instruments stays balanced.

If both *Wide* and *Timbral* are selected the 31-band limiter operates as a pseudo-ten band limiter with the Timbral algorithm running on top of it. This mode may be useful if one wants to create a special limiting effect though in our opinion the limiter is best operated in one of its other three modes (Discrete, Wide or Timbral)

31 Band Limiter Sidebar

Just like the five band AGC/Compressor, there is a sidebar with access to adjustments for tailoring the sound of the 31-band limiter. All limiter bands may be adjusted as a group or independently in the sidebar menus. Please see the section of the manual on "The Blue Dots" for more information on how to adjust the controls in the GUI's graphical areas.



Threshold

The *Threshold* control determines at what level limiting will start in a particular band. The range for the AirAura limiter threshold controls is adjustable over a 12dB range and they work in concert with the 31-band Limiter Drive to set the individual band's operating behavior. The threshold of the limiters is determined not only by the settings created here, but also by the Multiband Mode ("Soft" or "Hard") mode.

Attack

The *Attack* control determines how quickly the Limiter in each band responds to audio peaks once they exceed the setting of the limiter threshold. Recommended Attack Time settings are between 30 and 10ms. Longer attack times are possible for frequencies below the VBMS frequency setting however longer attack times may also cause the Lookahead Limiter to work harder and/or overshoot if it is being used.

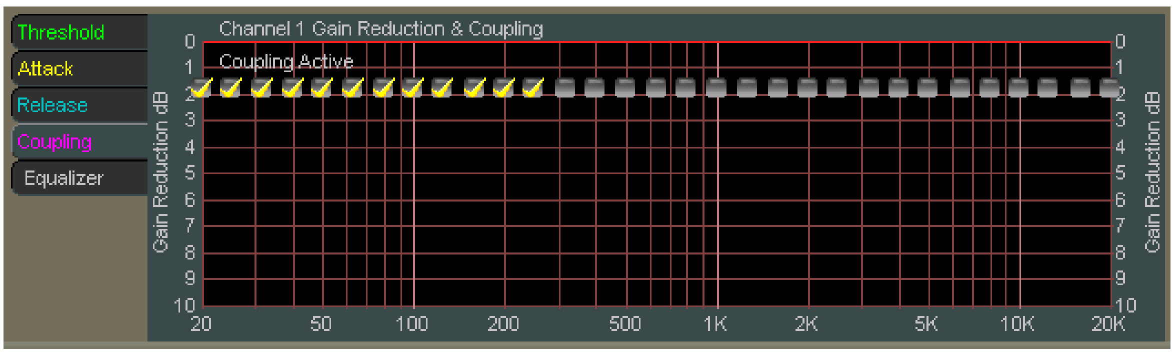
Release

The *Release* control determines how quickly the limiter will recover to 0dB once audio has fallen back below the set limiter threshold. Faster release times will yield a louder sound while slower times will yield a smoother sound, somewhat at the expense of loudness. Very fast release times, faster than the attack times in fact (!), allow the freedom to create very controlled intermodulation "texture" and "slam," and only in certain spectral ranges, when desired.

Because of the way the 31-band limiter works, there is a lot of leeway with attack and release times and the ranges of the controls have been set so that the end user can't get into a lot of trouble. Though the controls are "forgiving" in that sense, improper settings may yield undesired sound, so these controls should also not be adjusted haphazardly.

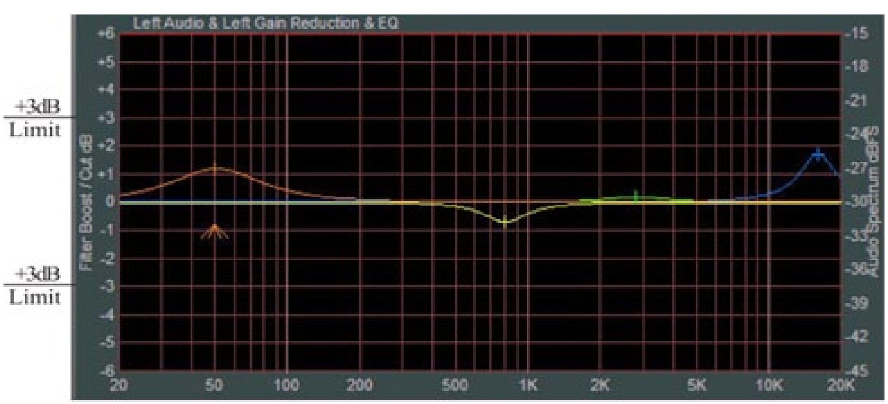
Coupling

When the 31-band limiter is not operating in its Discrete mode (*Wide* or *Timbral* is checked) the individual limiter bands may be selective decoupled from the Wide or Timbral algorithm if desired. For instance, to achieve a certain sound the user might discover that he gets the sound he wants if he operates the limiter bands above 300Hz independently but allow the bands below 300Hz to operate according to the Wide or Timbral mode he selected. The graphic below shows how the Coupling controls would appear in his scenario.



Equalizer

The post-limiter *Equalizer* serves as the final place where one should make only *slight* adjustments to the sound of the final processed output. Adjustments in this section have been purposely limited to a small range to prevent problems due to maladjustment. Our recommendation is to leave these equalizer settings set to "flat" because EQ changes should normally have been made upstream in the L + R mixer or in the parametric EQ. On the other hand if only a little more of something is wanted, these controls have been provided. Because of the sensitive nature of where this EQ is located, boost and cut have been restricted to a modest +/- 3dB.



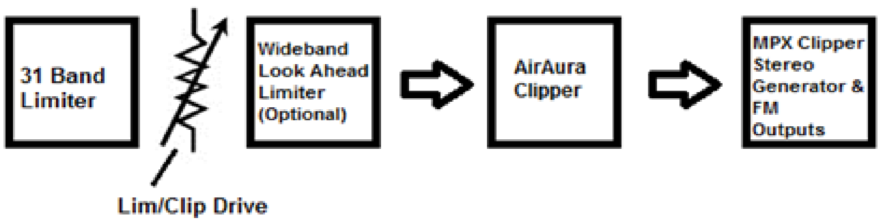
Adjusting this final EQ is the same as adjusting the Parametric EQ when directly manipulating its curves using the mouse. Double clicking on the little crosshair at the top (or bottom if it's below the line) of the curve turns that section on and off (white arrow). Clicking on the crosshair and holding the left mouse button down as you drag the mouse moves the curve to any frequency (left or right move) as well as adjusts the curve's height above or below the 0dB reference line (up or down move). Clicking on the little "tent" underneath (red arrow), and holding the left mouse button down while sliding it left or right changes the bandwidth, or "Q," of the equalizer section.

Lim/Clip (Limiter/Clipper) Drive

The *Lim/Clip Drive* control sets the drive level of both the Lookahead Limiter (if enabled) and the AirAura FM clipper. The adjustment range is +/-6dB and the adjustment resolution is 0.05dB, fine enough for the most exacting clipper or limiter drive adjustments.

Where Is The Lim/Clip Drive Control?

The *Lim/Clip Drive* is at the output of the 31-band limiter and just before the Wideband Limiter. It is called “Limiter/Clipper” because it drives both the Wideband Lookahead Limiter (if enabled) followed by the AirAura FM clipper. If the Wideband Limiter is not enabled, this control directly drives the AirAura Clipper.



Clipper Style

There are three final clipper styles offered in the AirAura Clipper, Hard, Firm, and Round. Each has a different sound texture. These choices enable the end user to create any on-air sound desired. All three styles offer excellent peak control because the main clipper is followed by an overshoot calculator that removes peaks that might cause the main 15kHz lowpass filter to overshoot. The Main Clipper’s drive level is determined primarily by the setting of the *Lim/Clip Drive* control as well as the amount of activity occurring in the 31 Band section.

Hard

The *Hard* clipper style has an infinite ratio with no increase in peak levels once its threshold is achieved. This is the least “forgiving” of the clip styles as far as clipper drive levels are concerned. This clip style is however the “cleanest sounding” of the three styles because unlike the other two, no clipping whatsoever occurs BELOW the clip threshold.

The other reason this is a “cleaner” clipper is because the distortion spectrum created by this style is more easily manipulated by the distortion cancellation scheme. The *Hard* style creates predominately harmonic distortion up to approximately 1.5dB of drive, depending on the incoming program material.

Firm

The *Firm* clipper style has a transfer function that allows progressively higher ratios over approximately a 0.75dB transition. It is a softer sounding clipper than the *Hard* style and is also more forgiving in drive level. But it *can* sound grainy on some material because the distortion spectrum is richer and not as easily managed by the distortion cancellation scheme. In essence it sacrifices a little cleanliness for additional ‘fatness’ and ‘meat’, plus more easily heard details in the midrange and highs.

The *Firm* clipper style creates both harmonic *and* intermodulation distortion with the balance leaning towards harmonic rather than intermodulation.

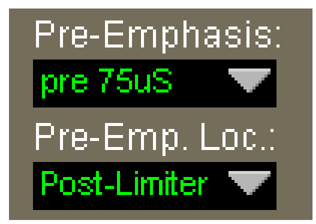
Round

The *Round* clipper style has a transfer function that allows progressively higher ratios over approximately a 1.5dB transition region. This is most forgiving of the three clipper styles as far as drive levels are concerned and contributes quite a bit of "roundness" without sounding overly harsh.

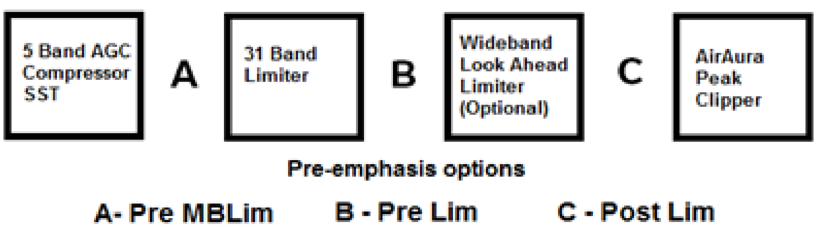
However … the *Round* style is the least "distortion cancelled" of the three styles because of the richer harmonic spectrum created. The *Round* style creates both harmonic *and* intermodulation distortion with the balance leaning towards intermodulation. This is the "warmest" and "thickest" sounding of the three clipper styles and works quite well with bass on Urban based formats – not so much for others.

Clipper Drive Note: The *Lim/Clip Drive* control will need to be readjusted when switching back and forth between the different clipper styles because due to the changing transfer function the perceived "amount" of clipping *increases* as the style is switched from Hard, to Firm, to Round. For example, if the Clipper Style is switched from Hard to Round, the setting of the *Lim/Clip Drive* control will need to be reduced to arrive back at the same perceived density and distortion. The amount of reduction required is highly dependent on how hard the 31-band limiter is being driven.

Pre-Emphasis



FM *Pre-Emphasis* is selectable as Off, 50 or 75 microseconds. Pre-emphasis may be in one of 3 locations in the signal chain:



Pre-MBLim

In Pre-MBLim pre-emphasis is placed *before* the 31-band limiter which then causes the 31-band limiter to operate on pre-emphasis curve as a frequency conscious multiband limiter. This setting is best used when "softer highs" are the goal albeit with some loss of transient detail.

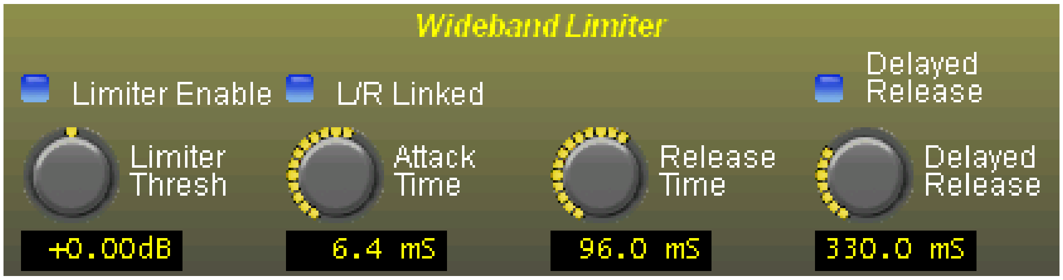
Pre-Limiter

In Pre-Limiter pre-emphasis is inserted *after* the 31-band limiter but is before the Lookahead limiter and main clipper. In this position the Lookahead Limiter may be operated as a HF limiter. Though this option is provided for convenience, if strong frequency selective HF Limiting is desired, we encourage the use of the Pre-MBLim option instead.

Post-Limiter

In this position pre-emphasis is *after* the 31-band limiter and Lookahead limiter and directly prior to the main clipper. This setting is the most invisible in operation and allows for the most accurate high end and transient impact.

Wideband Limiter



The output of the 31-band Limiter can be peak controlled by a special high-speed feed forward limiter with a 0.5 millisecond lookahead time. The control path, and not the audio path, is oversampled in order to achieve extremely precise peak control. Because the goal of oversampling is better peak control through finer definition of the audio waveform, oversampling of the audio signal itself is not only unnecessary, but wastes DSP resources.

Limiter Enable

When this checkbox is checked the Lookahead limiter is enabled.

L/R Linked

When this box is checked the left and right channel control signals are made common and the greater of the two controls *both* channels, preventing unnatural stereo image shifts when one channel needs to reduce the signal to control peak excursions and the other doesn't.

Limiter Threshold

The *Limiter Thresh* control sets the relative threshold of the lookahead limiter compared to the setting of the *Lim/Clip Drive* control. It is adjusted until the onset of limiting is reached with normal program material as indicated by the FM gain reduction bargraph. The Limiter Threshold is adjustable over a +/- 6dB range.

Attack Time

The *Attack Time* control sets the attack time of the look ahead limiter. Attack times from 0.2 milliseconds through 0.5 milliseconds allow the limiter to "look ahead" in time and react to signal peaks *before* they arrive at the limiter's gain control algorithm. Attack time settings greater than 0.5 milliseconds and up to the limit of 30 milliseconds allow the limiter to operate as a conventional, non-lookahead limiter.

NOTE: The 0.5 millisecond look ahead is always active. If the attack time control is set at 1.0 millisecond the actual attack time is 0.5 milliseconds. The lookahead value of 0.5mS must be subtracted from the control setting to know the actual attack time.

Release Time

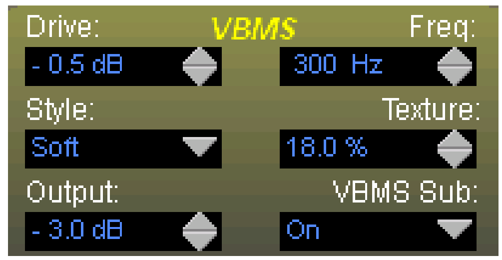
The *Release Time* control sets the primary (fast) release time of the lookahead limiter. Slower settings (longer release times) cause the peak limiter to recover more slowly from limiting. Release times set too slow may allow undesirable signal ducking to occur.

Fast release times increase signal density, and release times that are too fast result in increased intermodulation distortion. The control range is 33.0 to 330.0 milliseconds.

Delayed Release

The *Delayed Release* control sets the secondary (slow) release of the look ahead limiter. When Delayed Release is engaged, the limiter will release the first 3dB of gain reduction at the (faster) Release setting, and the remainder of the Release at the Delayed Release timing. This will drastically reduce intermodulation distortion and allow more limiting depth without incurring a distortion penalty. The control range is from 100ms to 1.0 seconds.

VBMS (Vorsis Bass Management System)



The Vorsis Bass Management System allows extremely low program frequencies to be processed without causing dynamic gain changes and without generating undesirable intermodulation artifacts in higher frequency signals. Its primary purpose is to allow the thumps of kick drums and other percussion instruments to sound natural in their "percussiveness" without pumping or other undesired artifacts. The algorithm operates in left/right mode treating both channels identically. Field research revealed that it is perfectly acceptable to process very low frequency material using the VBMS prior to all modern FM exciters.

Drive

This control adjusts the amount of Input Drive to the VBMS algorithm. Higher drive settings make it work harder and generate higher levels of harmonics. The *Drive* control adjusts the amount of drive to the VBMS and is adjustable over a range of +/- 10dB.

Style

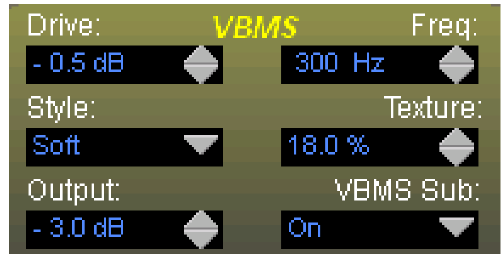
Two VBMS operating styles, plus Off, are available. The *Hard* setting has a transfer function occupying approximately 1dB of transition amplitude. The waveform control is fairly abrupt and generates a fair amount of low order harmonics.

The *Soft* setting operates with a transfer function occupying approximately 2dB of transition amplitude. Therefore the control is less sharp than the hard setting, generating more low order harmonics. While the two settings may appear to be almost the same, the 1dB difference in transfer ratio creates a *significant* difference in how the two VBMS styles *sound* on actual program material.

Output

The VBMS *Output* control adjusts the output of the VBMS algorithm and can be thought of as a "low bass mix" control. Settings between -6dB and -3.0dB add extra bass to the overall mix, while settings lower than -6.0dB reduce it. Both the VBMS *Drive* and the VBMS *Output* controls are best adjusted by ear using percussive bass material. Care should be exercised in the setting of the VBMS Output control because settings above +0dB may cause bass to reach the main clipper which will add distortion. Like VBMS Drive, the VBMS Output is adjustable over a range of +/- 6dB.

VBMS Frequency



The VBMS *Freq* control tunes the VBMS algorithm's frequency of operation from 60Hz to 300Hz. Program energy not passing through the algorithm is time aligned for coincidence regardless of the VBMS settings. VBMS operates *only* on frequencies *at and below* the VBMS Frequency setting.

Texture

The *Texture* control sets the ratio of odd to even harmonics in the VBMS circuit. Normally the VBMS adds only odd harmonics so some mechanism is needed to add some even harmonics too so that smaller speakers seem to have better bass than they are physically capable of reproducing. This is what the VBMS Texture control does. Higher settings are recommended for rock/classic hits based formats while Urban, R&B and many CHR formats may wish to operate with the control set lower. Note that settings above about 35% may sound unnatural on some program content.

Sub Bass

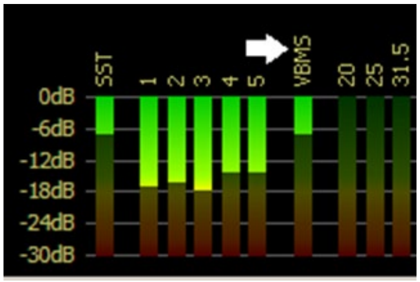
The *VBMS Sub* Bass control has an On/Off function that enables and disables a dynamic low frequency extension circuit that enhances the presence of subsonic energy like a subwoofer does and is useful for formats that require extra deep bass, such as Urban and CHR.

NOTE #1: We do not recommend *VBMS Sub* being enabled if substantial bass equalization has been dialed in using the parametric EQ or the L+R EQ mixer in the AGC/Compressor. We also recommend listening to the station with an audio system *capable* of reproducing sub bass frequencies while experimenting with *VBMS Sub* because its effect will be *essentially inaudible* on systems that are incapable of reproducing very low frequencies.

NOTE #2: When *VBMS Sub* is enabled we recommend operating the High Pass Filter (Input Menu) at settings no lower than 30Hz. In fact slightly higher settings, such as 40Hz, might be more appropriate, depending on the format. Failure to heed this advice may allow undesirable subsonic signals to get on the air.

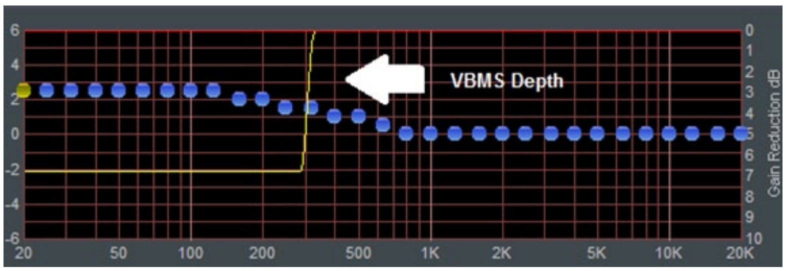
NOTE #3: Remember that the majority of consumer sound systems cannot reproduce audio signals below about 40-50Hz. Exceptions to this are systems specifically equipped with subwoofers and/or large air mass speaker systems. It is important to recognize the capabilities of the station's typical listeners and whether or not they might be equipped with audio systems capable of reproducing sub bass. This of course *suggests* a highly format dependent situation!

VBMS Metering



There are two ways to monitor the amount of processing being done by the VBMS

VBMS can be seen on the Analysis Screen on the PRO GUI, in GUI Guru or on the front right screen panel of the AirAura. VBMS can also been seen in PRO GUI in the Limiter gain reduction graph. Its activity is indicated by the yellow line.



Depending on how hard the VBMS is driven, one may or may not observe much activity on the VBMS metering. In fact, only when a lot of bass is present in source material will VBMS show any activity on the meters, however this does not mean VBMS is not working.

Digital Output

The *Digital Output* control sets the absolute peak output level in dBFS after all processing.

The front panel and remote GUI bargraph meters display the peak output levels in dBFS.

The control has an Off position and levels can adjusted in 0.05dB steps over the range of -79.95dB to +00.00dBFS.

De-Emphasis

When *De-Emph* is checked the appropriate de-emphasis is applied to the FM digital output according to any pre-emphasis which may be in effect. If no pre-emphasis is being applied this checkbox has no effect.

PreDelay

When the *PreDelay* box is checked the digital FM output will always be sourced ahead of the FM Diversity Delay.

FM Diversity Delay

AirAura 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 in steps of 100 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.

Adjustment of Diversity Delay

Front panel adjustment of the Diversity Delay using the trackpad has enough resolution to get the diversity delay in the ballpark, and finer adjustments (down to 100 microsecond steps) can be made using GUI to set the delay.

For best resolution of the Diversity Delay controls they are best adjusted using the host PC's keyboard and mouse and using the following table as a guide.

COARSE ADJUST SLIDER		FINE ADJUST SLIDER	
Slider only	500 mS	Slider only	0.9 mS
Slider plus Shift key	1700 mS	Slider plus Shift key	8.6 mS
Slider plus Ctrl key	50 mS	Slider plus Ctrl key	0.1 mS (100μSec)

Note that the smallest increment of delay time adjustment is 100 *microsecond*.

Entering Delay Values Manually

Double-clicking on the delay value window opens up a dialog where the numeric delay value in milliseconds may be entered. This is useful when you know roughly how much delay is required or wish to try various values quickly.

STEREO ENCODER MENU

The *Stereo Encoder* screen contains the controls required to configure the AirAura's internal DSP-based precision stereo generator.

Stereo Encoder Screen

MPX Processor Style

AirAura 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. Note that we do not mathematically remove the higher order odd harmonics in this clipper algorithm like we do in the VBMS section because those harmonics are *required* in order to make the waveform have a tightly defined peak amplitude.

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 look ahead and extremely high sample rate precisely controls composite waveform peaks on a cycle by cycle basis. It works similarly to the oversampled look ahead peak limiter in the FM Limiter screen, only at a much higher sample rate.

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

MPX Process Drive

The *MPX Process Drive* control sets the input drive level to either the composite clipper or lookahead limiter as selected. High settings create higher clipper or limiter drive settings while lower settings do the opposite. We recommend that the Composite Clipper Drive be kept under +2.0dB for best MPX spectrum cleanliness (all composite clippers create MPX spectrum "trash").

Automatic Multipath Limiter

AirAura'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.

Automatic Multipath Limiter Threshold

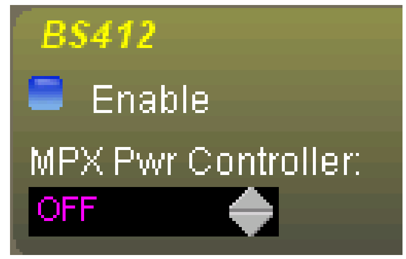
There are eleven operational choices available under the *Automatic Multipath Limiter Threshold* control settings: 10% to 100% in 10% percent steps, and an Off setting. Off is the setting immediately above the setting for 100%.

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 Automatic 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.

ITU BS412 MPX Power Controller

AirAura's BS412 *MPX Pwr Controller* has 15 possible settings, plus OFF: +8dB, +7dB, +6dB, +5dB, +4dB, +3dB, +2dB, +1.5dB, +1.0dB, +0.5dB, 0, -0.5dB, -1.0dB, -1.5dB, and -2.0dB. When the control is in any position except Off it is measuring and displaying the current MPX Power level on the BS412 meter on the front panel and GUI. Calculated corrections to the MPX power are only made when the BS412 *Enable* checkbox is checked, and when it is checked, modifications to the MPX power will take place immediately!



IMPORTANT!

The BS412 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 BS412 Controller <u>do not</u> check the BS412 *Enable* checkbox. Turning on the BS412 MPX Power Limiter can result in up to a 5dB loss in loudness.

The 0.5dB resolution near the control's zero setting is to allow small adjustments in the MPX Power's reference level to compensate for unusual things a user might do to create his on air sound. Though the MPX Power Controller is not easily fooled we provided a small amount of tweaking near the "0dB" reference level "just in case."

TURNING ON THE BS412 MPX POWER CONTROLLER

Any time the BS412 MPX Power control is not set to Off, the BS412 algorithm is measuring and displaying the current MPX power on the front panel and GUI BS412 meters. When the BS412 *Enable* checkbox is checked 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 BS412 control. The control's "0dB" setting conforms to the current ITU-R BS.412-7 Multiplex Power standard.

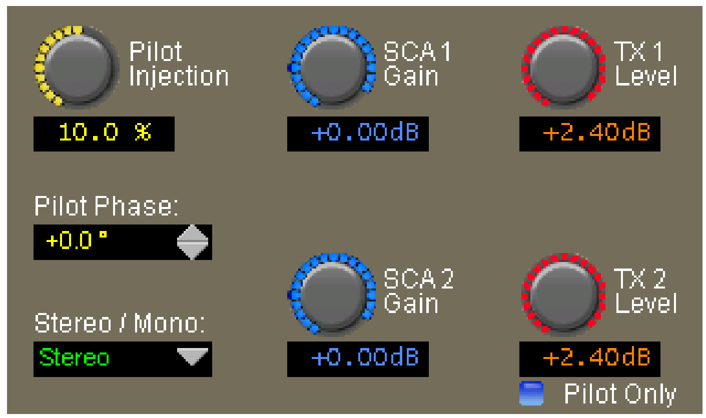
TURNING OFF THE BS412 MPX POWER CONTROLLER

The BS412 Controller may be immediately taken off line by un-checking the *Enable* checkbox.

IMPORTANT!

If your station is <u>not</u> required to comply with a Multiplex Power standard where it is licensed to operate, the BS412 control should remain unchecked!

Pilot Injection



The *Pilot Injection* 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.

Pilot Phase

Small corrections to the 19kHz stereo pilot phase may be made in 0.5 degree steps up to +/- 22.5 degrees. The ability to vary the stereo pilot phase relative to the 38kHz subcarrier allows AirAura to compensate for nonlinear time delays in the transmission system. Such time delays skew the phase relationships between the 19kHz stereo pilot and the 38kHz stereo subcarrier and degrades stereo separation.

Stereo/Mono

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 completely turned off.

SCA 1 & 2 Gain

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 -79.95dB and +10.0dB and in 0.05dB steps.

TX 1 Level

The *TX 1 Level* control sets the peak output amplitude of the MPX 1 output on the rear panel BNC connector. The control is used to set total modulation in the presence of audio and can be set to OFF or to levels between -79.95dB and +6.00dB and in 0.05dB steps. The control setting for approximately 3.5V P-P (1V RMS) into a 10k ohm load is +2.40dB.

TX 2 Level

The *TX 2 Level* control sets the peak output amplitude of the MPX 2 output on the rear panel BNC connector. The control is used to set total modulation in the presence of audio and can be set to OFF or to levels between -79.95dB and +6.00dB and in 0.05dB steps. The control setting for approximately 3.5V P-P (1V RMS) into a 10kohm load is +2.40dB. If TX 2 *Pilot Only* is checked, the TX 2 output becomes a 19kHz sine wave synchronizing source for RDS generators.

Analog Output Type

Please pay careful attention to the following analog output description as it is different from those found in all other broadcast audio processors!

Composite MPX

When selected, the male XLR outputs and the female BNC connectors assigned to TX 1 and TX 2 carry wide-band, line level composite stereo audio and, as in the L/R case above, the XLR outputs are electronically balanced. These balanced outputs can be used to gain a signal to noise advantage with exciters having a balanced composite input (most modern ones do). The adjustment range of the analog outputs, regardless of the operating mode, is -80dB to +6dB in 0.05dB steps, covering virtually every conceivable operating need in the field

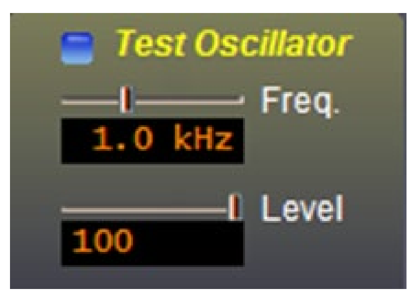
Left/Right

The male XLR outputs and the female BNC connectors assigned to TX 1 and TX 2 carry line level audio. In the case of the XLR outputs the audio is electronically balanced. If pre-emphasis has been selected in the FM Limiter screen then these outputs are pre-emphasized also.

Left/Right De-Emphasized

The male XLR outputs and the female BNC connectors assigned to TX 1 and TX 2 carry line level audio but are de-emphasized complementary to any pre-emphasis that has been selected in the FM Limiter screen.

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 *Test Oscillator* checkbox is selected.

A special frequency, "Bessel," is included. This is a 31.250kHz test signal which can be used for modulation monitor calibration using the first Bessel null.

SYSTEM MENU

The *System* screen allows system-wide settings to be configured. In the first third of the screen is a very important checkbox labeled *Special Options* with two choices.

Input/Output Settings Follow Preset Takes

When this box is unchecked, 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 AirAura SYSTEM, or global, parameters, which are saved with presets but not restored when presets are taken, unless the *Input/Output Settings Follow Preset Takes* checkbox is checked. When this box is checked, the following control settings are recalled whenever a preset is recalled:

INPUT Screen:

Current Audio Source (Analog or Digital)

Analog Gain

Digital Gain

Balance (common to both Analog and Digital Inputs)

Phase Rotator

FM Output Screen:

DeEmphasis

PreDelay

Digital Output

Stereo Encoder Screen:
Pilot Level
Pilot Phase
SCA 1
SCA 2
TX 1
TX 2
BS412
Pilot Only
Analog - L/R, L/R Deemph, and MPX output style choices

The Input/Output Settings Follow Preset Takes choice allows AirAura to modify its input and output gain and other system 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 *Input/Output Settings Follow Preset Takes* checkbox is unchecked.

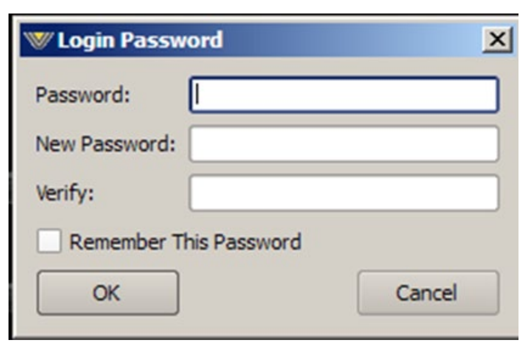
Metering Data Over TCP/IP

By default, AirAura 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 AirAura 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 AirAura 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.

Remote Login

When *Remote Login* is clicked, a password prompt box will open as shown on the right.



Remote Login is for password restricting Pro GUI access to an AirAura over Ethernet. When a GUI TCP connection session is initiated, the AirAura accepts the connection and then asks the remote Pro GUI for a password.

Unless and until the login password can be verified the AirAura will not send its current settings to the remote GUI nor will it accept any changes from the connecting GUI.

The dialog box show above also allows a user to change the login password if one has been set. The current password will need to be known beforehand, and it will need to be entered twice for verification. The password can be any combination of characters, numbers, or symbols. If the *Remember This Password* check box is checked, the GUI stores the password on the local PC. If you do not check the check box you will be asked for the password each time you connect (since the first round of verification will fail).

When asked for a password while logging in you can also check the “remember” box to save what you enter. That way you don’t need to enter the password every time you login.

Occasionally you may see the word “pending” temporarily appear in the connection status box of the remote GUI. That means the AirAura and Remote GUI are handshaking on the password string that is being sent.

Pro GUI Access

The AirAura’s Security System allows you to grant access at certain levels of permission in order to control the privileges of those accessing the various features of the AirAura. Privileges are granted by *Access Code*, not by user type. In fact, user types are created *through* the generation of access codes for those user types.

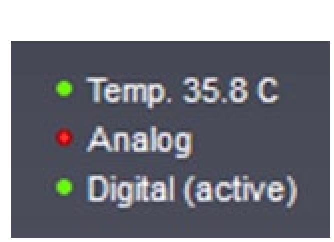
Examples:

- Suppose *Access Code* 0000 is created with *Preset Takes* being the only box checked. This means that a user which has been assigned access code 0000 can *only* take presets.
- Suppose *Access Code* 5678 is created with *Preset Takes* **and** *Setting Changes* both checked. This means that a user with access code 5678 cannot only *take* presets, but can also *make* changes to the settings within those presets. Note that because *Preset Saves* was *not* checked when this access code was created, that user may *make* changes to a preset but may not *save* them!

Pro GUI Lock

In the event that security has been enabled by configuring the *GUI Access Code & Permissions* Dialog (above), then selecting *GUI Lock* will immediately lock the GUI on the host PC, prohibiting access until it has been unlocked with the proper access code.

Status Indicators



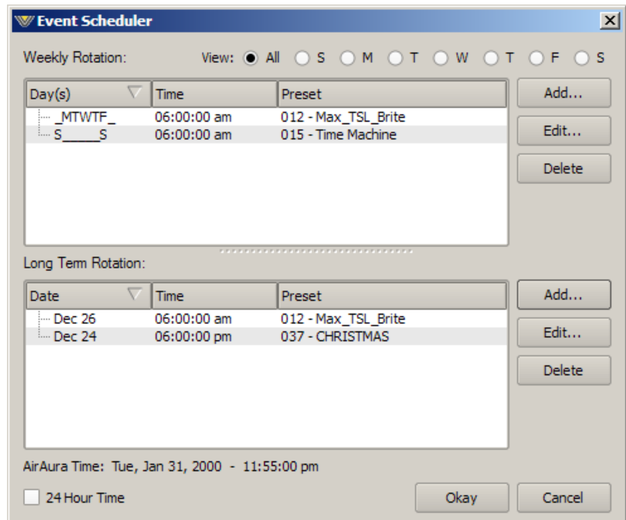
Temp – When the *Temp* indicator is green, the operating temperature inside the AirAura is satisfactory. When it is red, the temperature is too high. The displayed value is the current operating temperature in degrees Celsius of the AirAura motherboard. Normal temperatures should be in the range of 25ºC to 50ºC, and will vary somewhat depending on ambient temperatures near the AirAura.

Analog – When green, this indicator shows that audio is present on the analog line inputs and that its level is above -24dBu. When the indicator is red it means that audio is missing on one or both channels or is too low to be detected.

Digital – When green, this indicator shows that valid data is present on the AES3 digital input, and that any audio present is above -48dBFS. When the indicator is red it means that valid digital audio data is not presently being detected, audio data *could* be present but is perhaps corrupted in some way, or that audio is present but is simply too low to be detected.

Scheduler

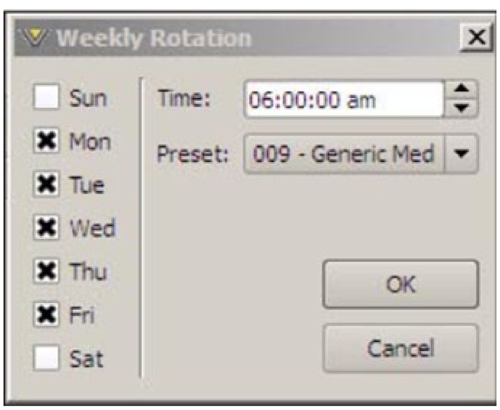
The AirAura contains a comprehensive scheduling utility that allows presets to be automatically changed on desired dates and times. There is also a *Long Term Rotation* scheduler that allows preset changes to be scheduled for any time in the future. To create, edit, or view entries in the *Event Scheduler* click the *Scheduler* button on the System Screen.



The *Event Scheduler* screen shown on the right will open and the events list will be empty when opened for the first time:

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 below.



2. On the left column, select the days that the preset should be selected. Any combination of checkboxes may be chosen.

3. Next select the time that the preset should change.

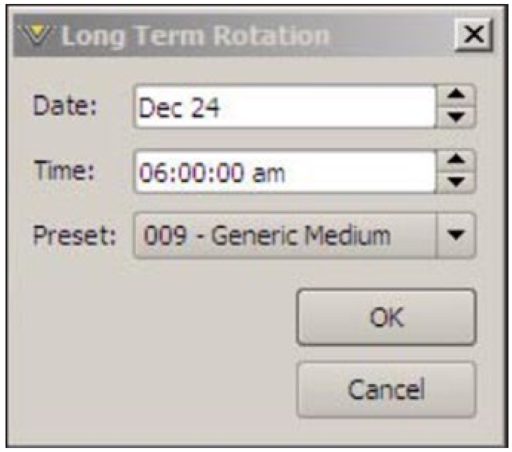
4. Next choose the preset that should be selected at the time programmed in the previous step. When finished editing, click *OK*.

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.

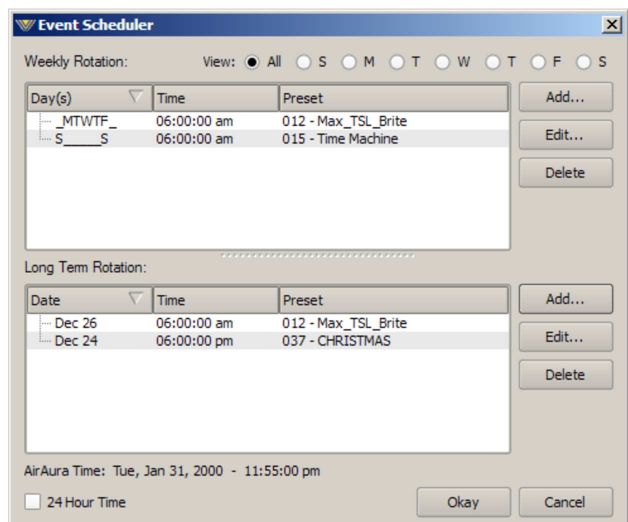
Creating a Long Term Rotation Schedule

Adding a new event in the *Long Term Rotation* scheduler works in a similar manner. The *Long Term Rotation* entry window appears as follows:



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.

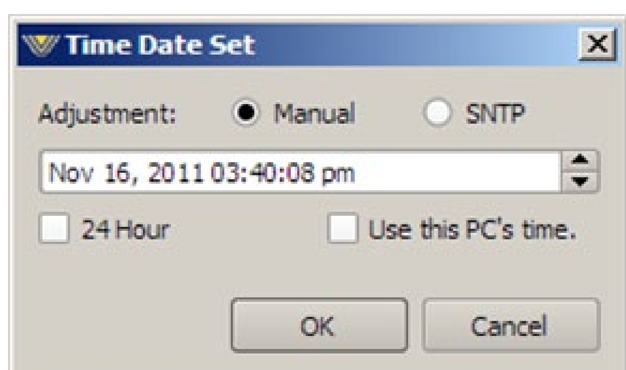


To delete an item in either the *Weekly Rotation* or *Long Term Rotation* schedules simply highlight it in the list and then click the appropriate *Delete* button on the right side of the Event Scheduler window. The highlighted scheduled event will be removed from the list.

Note that there is no confirmation prior to the actual deletion of the event!

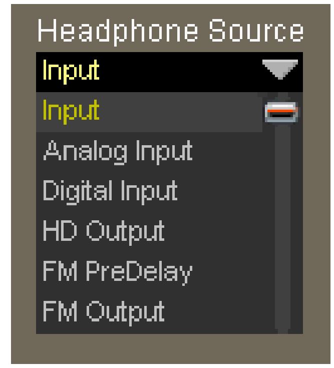
System Time

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.



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.

Headphone Source



The headphones may be driven by a number of selected points inside the signal processing chain as follows:

Input: Output of the audio input selector, analog or AES.

Analog Input: Audio appearing on the analog line input.

Digital Input: Audio appearing on the AES3 digital input.

HD Output: HD output of the processing chain.

FM PreDelay: Output of the FM limiter before the diversity delay.

FM Output: FM output of the processing chain.

NOTE: The AirAura headphone routing selector allows you to monitor an unselected audio input *without* having to put it on the air. This can be handy for confirming that an audio source is present without actually switching to it!

Headphone Control

On the left side of the front panel of the AirAura is a quarter inch female TRS jack for headphone use. The volume control for the headphone output is located on the lower right side of the Pro GUI.

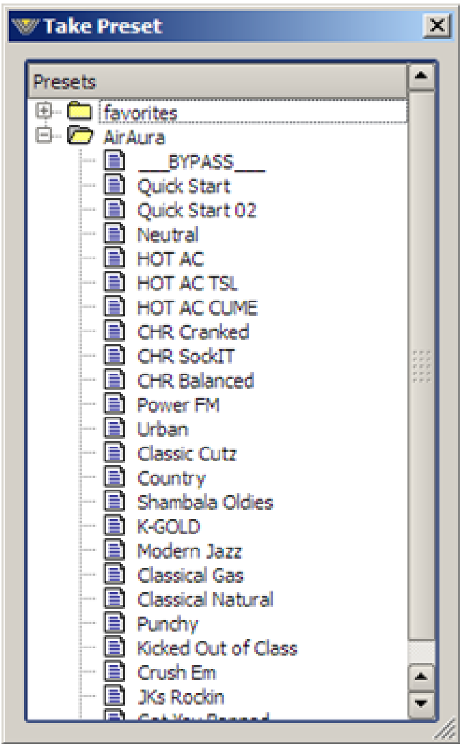
To adjust the headphone volume slide the fader right to increase volume and left to decrease it.

Note: The headphone outputs of the AirAura are driven by National Semiconductor LM675 power operational amplifiers capable of >3A into any reasonable load. Actual headphone impedance is not a concern; however headphone circuit current limiting is provided at 160mA in order to protect external devices and also to keep you from setting your hair on fire from too high a headphone listening level!

Side Bar Region

To the right of the control area is a vertically disposed row of buttons. We will explain each button in turn.

Presets



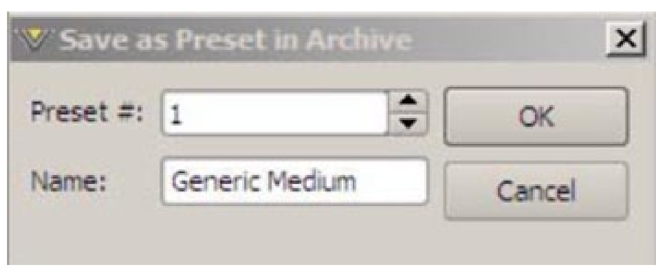
When left-clicked, a dialog box appears, showing folders that contain presets, typically favorites and AirAura, which lists all presets stored on the hardware.

Double-clicking on a preset brings it immediately into use in the AirAura. 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. ***It is important to remember to save the present settings as a preset before invoking others, or they will be lost.***

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

Save

Clicking on the *Save* button opens a dialog box (shown on the right), prompting the user for a name for the preset 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. Note that it is possible to save the present settings in over an existing preset and doing so erases the previous preset's contents.



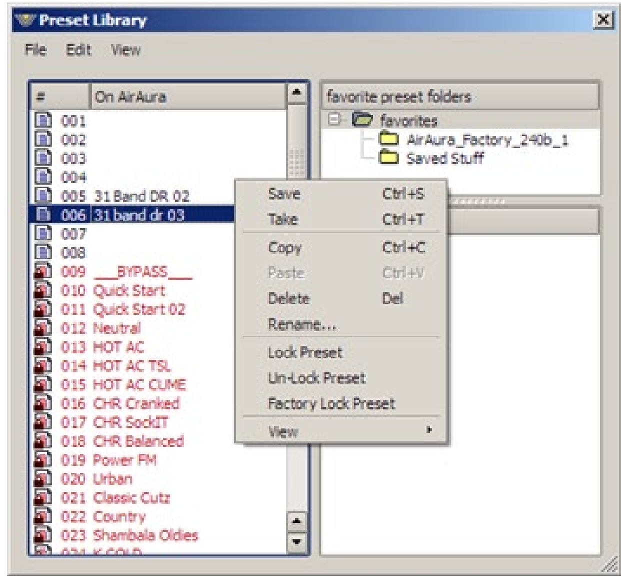
Factory presets are protected from being inadvertently overwritten.

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 AirAura.

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 AirAura 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:

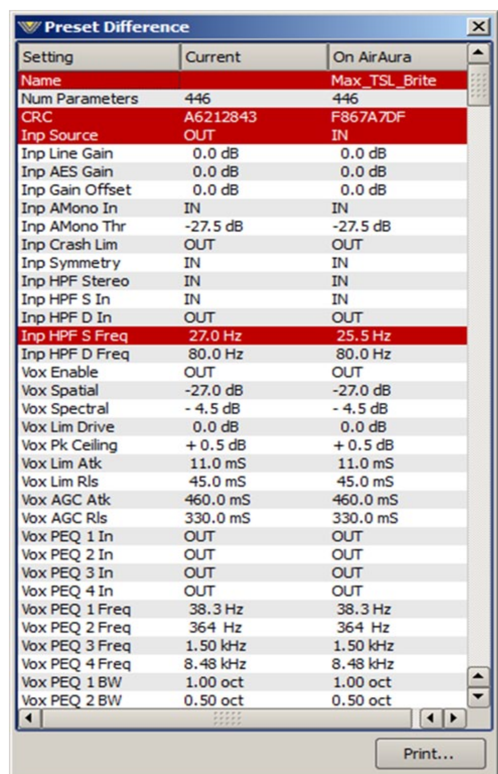
- The first method is by highlighting a preset (single left click) and then right clicking it to open a dialog box. Among the options are Lock Preset and Unlock Preset.
- The second method is by highlighting a preset as above, and then clicking the *Edit* option at the top of the *Preset Library* dialog 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 Changes

It is possible to visually compare two different presets on the AirAura 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.

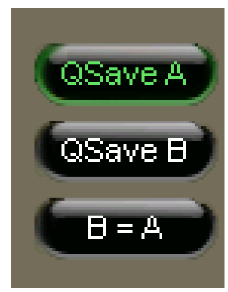


Devices

Clicking on the *Devices* button opens a list of AirAura devices known to the GUI (if any). Highlighting the desired AirAura's name and hitting *Select* causes the GUI to attempt to connect to it if you are Online.

In the event no AirAura's are listed yet, or a new one needs to be added, clicking the *Add* button brings up a small *Edit Device* dialog box which asks for the new prospect's name and IP address.

Quick Save (QSave)



The AirAura has a unique feature that allows instant comparison of work in progress to a known reference, such as another preset, or some midpoint while adjustments are being made. We call this feature **QSave** (for **Quick Save**).

In the upper right side of the AirAura Pro GUI you'll find three buttons like those shown on the right.

QSave A and *QSave B* represent two temporary memory buffers that hold all current AirAura processing settings as long as power is applied to the unit.

While *QSave A* is highlighted green any adjustments that you make to controls on the AirAura are being concurrently saved to temporary buffer *A*.

QSave B operates on buffer *B* and just like *QSave A* its being selected is shown by a green highlighting.

One way to use this feature is to compare the sound of a factory preset to changes you've made to it without having to first save it as a user preset. To do this:

- Recall the factory or user preset that you wish to adjust.
- Ensure that *QSave A* is highlighted. If it is not, press its button to highlight it.
- Press the *B=A* button. This will copy the contents of QSave buffer *A* to QSave buffer *B*. Now the contents of both *A* and *B* buffers are identical.
- Change some settings on the AirAura and they will automatically be stored in the *A* buffer.
- Compare your changed settings to the recalled factory preset by pressing the *QSave B* button.
- Compare those settings back to the factory preset by pressing *QSave A*.
- When you are happy with your changes you can commit them to a user preset using the preset *Save* dialog explained previously.

Likewise, *QSave A* and *QSave B* can be used to compare the sounds of two *different* sets of *user* settings. To do this:

- Load the preset that you want to change, make changes to it, and press *QSave A* to save those settings to buffer *A*.

- Make some additional changes and then press *QSave B* to save those additional changes to buffer *B*.
- Now buffers *A* and *B* have your two different settings you can compare them by toggling back and forth between *QSave A* and *QSave B*.
- When you are pleased with one set of settings and need more buffers for further tweaking, you can use the *A=B/B=A* button to make the two buffers the same and have one of them to use to start comparing from again.

Title Bar Region

Along the top edge of the AirAura Pro GUI screen (in line with the *VORSIS AirAura* product label to the left, and the Windows About, Minimize, and Exit icons to the right) are indicators and controls for the management of devices and presets. The AirAura Pro GUI is capable of controlling multiple AirAura units (*Devices*) and managing the *Presets* within them. Whether a connection is made (*Status*), and which device and which preset are presently under command are indicated. Double-clicking these indications will gain access to respective management screens.

Status

This indicator shows four different statuses:

Online – The GUI is in communication with and directly reflects an AirAura.

Offline – The GUI and AirAura are disconnected.

Trying – The GUI is attempting to find the AirAura on the LAN and connect to it.

Pending – The GUI has found the AirAura and is trying to complete the connection.

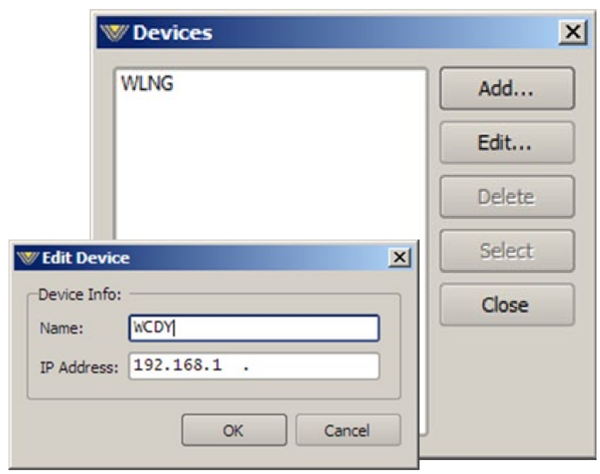
An adjacent and illuminated button toggles between *Online* and *Offline*.

Device

The name of the AirAura to which the Pro GUI is connected (or with which it is attempting to connect) shows in the *Device* window in amber text.

Double-clicking on the name area brings up the *Devices* dialog box. A list of AirAura devices that the GUI knows about is shown in the box. Highlighting the desired AirAura's name and hitting *Select* causes the GUI to attempt to connect to it if the *Online*/*Offline* button is green (Online).

In the event no AirAura units are listed, or one is adding a further device, or one for the first time, *Add* brings up the *Edit Device* dialog box, which asks for the new prospect's name and IP address.



Preset

The preset window shows the current preset that is active on the processor.

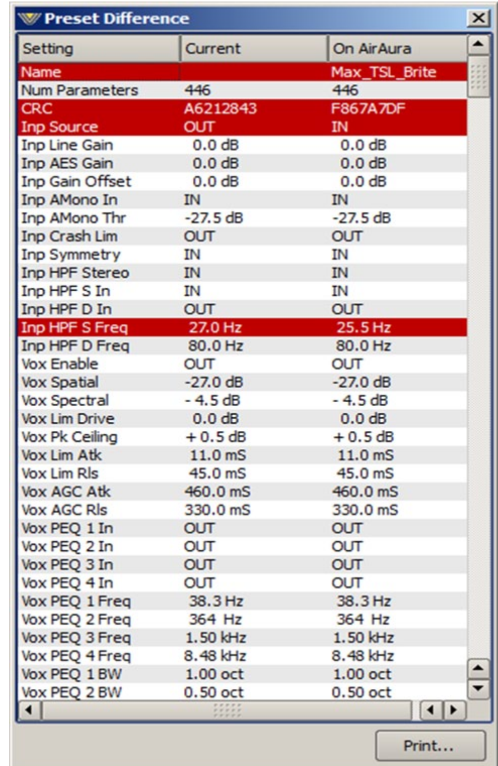
When preset name is displayed in GREEN text the currently running preset reflects exactly what has been last saved to that preset and that no changes have been made.

When the preset name is showing in RED text, the current preset has been modified and is not saved on the AirAura hardware. In order to return the preset name text to GREEN, the modified preset must be saved, or a new preset has to be retaken from the list of saved presets.

As covered earlier (page 4-67), it is possible to visually compare two different presets on the AirAura with a special feature called Preset Difference.

Double-clicking on the name brings up a *Preset Difference* – parameter-by-parameter, setting-by-setting – comparison list between the current AirAura settings and the recalled preset.

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 Difference

Setting	Current	On AirAura
Name		Max_TSL_Brite
Num Parameters	446	446
CRC	A6212843	F867A7DF
Inp Source	OUT	IN
Inp Line Gain	0.0 dB	0.0 dB
Inp AES Gain	0.0 dB	0.0 dB
Inp Gain Offset	0.0 dB	0.0 dB
Inp AMono In	IN	IN
Inp AMono Thr	-27.5 dB	-27.5 dB
Inp Crash Lim	OUT	OUT
Inp Symmetry	IN	IN
Inp HPF Stereo	IN	IN
Inp HPF S In	IN	IN
Inp HPF D In	OUT	OUT
Inp HPF S Freq	27.0 Hz	25.5 Hz
Inp HPF D Freq	80.0 Hz	80.0 Hz
Vox Enable	OUT	OUT
Vox Spatial	-27.0 dB	-27.0 dB
Vox Spectral	- 4.5 dB	- 4.5 dB
Vox Lim Drive	0.0 dB	0.0 dB
Vox Pk Ceiling	+ 0.5 dB	+ 0.5 dB
Vox Lim Atk	11.0 mS	11.0 mS
Vox Lim Rls	45.0 mS	45.0 mS
Vox AGC Atk	460.0 mS	460.0 mS
Vox AGC Rls	330.0 mS	330.0 mS
Vox PEQ 1 In	OUT	OUT
Vox PEQ 2 In	OUT	OUT
Vox PEQ 3 In	OUT	OUT
Vox PEQ 4 In	OUT	OUT
Vox PEQ 1 Freq	38.3 Hz	38.3 Hz
Vox PEQ 2 Freq	364 Hz	364 Hz
Vox PEQ 3 Freq	1.50 kHz	1.50 kHz
Vox PEQ 4 Freq	8.48 kHz	8.48 kHz
Vox PEQ 1 BW	1.00 oct	1.00 oct
Vox PEQ 2 BW	0.50 oct	0.50 oct

Print...

Notes on Online and Offline Working

Often it will be required that the Pro GUI act directly and instantaneously upon AirAura when a preset is selected. This is the *Online* mode

Offline, however, is more of a management utility intended for creating presets or tweaking them without needing to listen to them on the host AirAura.

Changes can be made to presets and configurations without affecting the sound of the processing occurring within AirAura.

Importantly, any adjustments made on the GUI *Offline* will not take effect until the preset changes have been saved, AirAura is *Online* to the GUI again, and the preset is made active by selecting it.

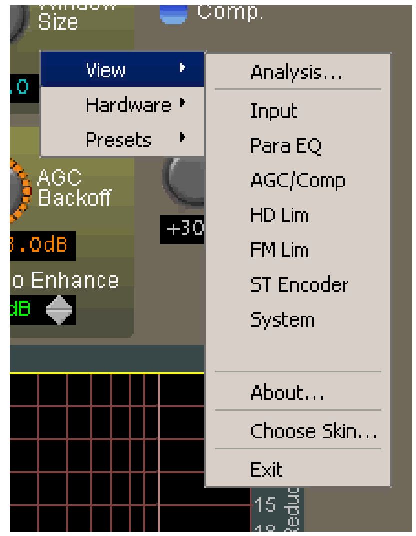
Invoking a preset – by double-clicking a preset from the box brought up by clicking the *Preset* button – does one of two things, depending on whether the Pro GUI is in *Online* or *Offline* mode:

- *Online* – the double-clicked preset is immediately sent to the AirAura and becomes active.
- *Offline* – the double-clicked preset brings that preset onto the main control and display screen for perusal/editing. Any data entry/control which involves activity within the main graph will still operate upon the preset, but not on the AirAura's audio.

Accessing Menu Options

Right clicking anywhere on the AirAura Control Panel will open a pop up menu tree with access to *View*, *Hardware*, and *Presets* choices. These choices lead to sub-menus and dialog boxes that may also be accessed by clicking on other dedicated buttons on the main AirAura control panel. As with many Windows programs, the Pro GUI has multiple ways to access menu trees – go ahead and explore!

View Menu Items



The *View* menu tree may be accessed by right clicking anywhere on the main AirAura Control Panel.

Analysis... – The AirAura Pro GUI offers the same analysis options as the Guru GUI. You can access these analysis features by clicking *Analysis*.

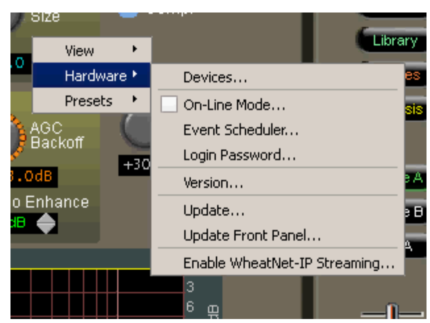
The next set of options in this window (*Input – System*) correspond to the tabs along the top of the GUI. This is just another place to access those options.

About – Brings up the *About* box to indicate the GUI's software version. Note this is not where the firmware versions running on the hardware are polled. That is done under the *Hardware* option covered in a moment.

Choose Skin... – Brings up the *Choose Skin* dialog box. In some GUIs there are multiple skins available which allow the user to pick different color schemes and such. Not all GUIs have multiple skins but many do.

Exit – Closes down (exits!) the entire Pro GUI (not just the dialog window).

Hardware Menu Items



The *Hardware* menu tree may be accessed by right clicking anywhere on the main AirAura Control Panel. Please note that most of the Hardware-related items require you to be connected (Online) to a AirAura because that is where the data must come from. Sub menu choices include:

Devices... – Opens the *Devices* dialog box. Allows the creation, editing, selection, and deleting of Vorsis processors connected to your system.

On-Line Mode...–Toggles between ONLINE and OFFLINE modes. This is the state the GUI will assume the next time it is started. This state is also remembered when the GUI is closed – if it was online to AirAura at the time it will attempt to go back online the moment the GUI starts up next.

Event Scheduler... – Opens the *Event Scheduler* dialog box for editing Events.

Login Password... – Opens the *Passwords* dialog box for editing login passwords.

Version... – Displays the current software and firmware versions running in the AirAura hardware.

Update... – Opens the *Choose a file to download* dialog box. Only valid files of type **.vbn* are visible. See below.

Update Front Panel... – Opens the *AirAura Front Panel GUI Loader* dialog box. Only valid Vorsis files of type *.lbn will be visible. See below.

New or updated AirAura software may be released at any time to implement new features, change specifications or correct known bugs. The Hardware Update/Update Front Panel choices will open separate dialog boxes which will prompt the user for the correct file type to be uploaded to AirAura. Upon completion of the update you will be asked to restart AirAura (cycle its power).

AirAura Hardware Update

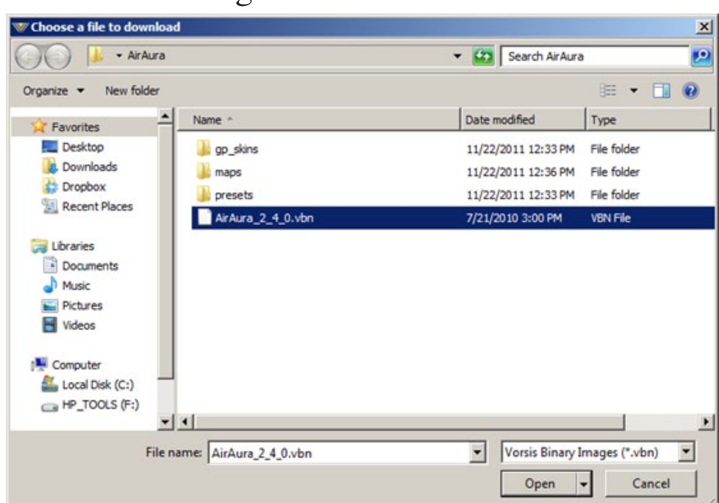
The AirAura is extremely easy to update when new firmware becomes available from Wheatstone. As mentioned above there are two different file types involved in updating AirAura. They are:

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.

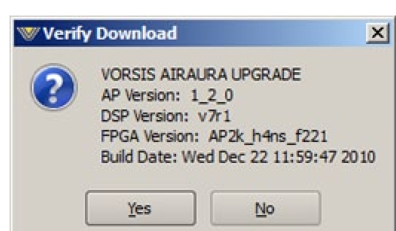
VBN and LBN file updates may be performed separately, or they may be performed together if updates for both sections of the AirAura have been made available.

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



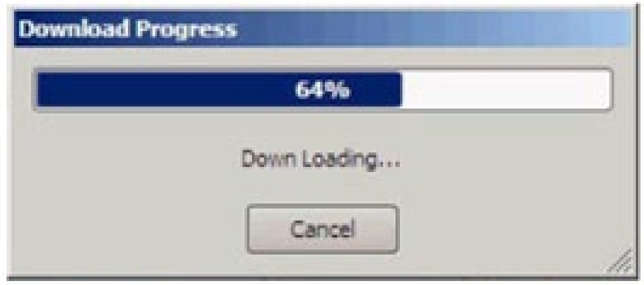
To update the AirAura, right click on any portion of the upper control area in any screen of the remote GUI to open pop up menu tree. For VBN updates of the motherboard, select the *Hardware Update...* option listed above the *Update Front Panel....* The screen at right will appear.

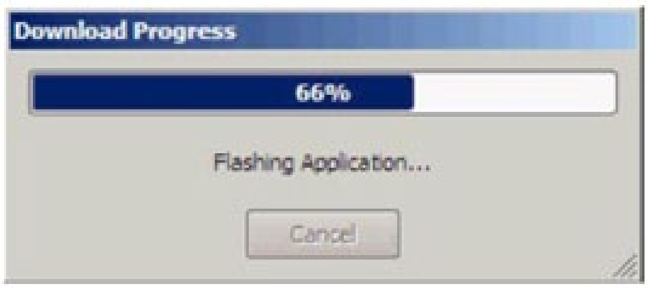
Navigate to the location of the VBN file that was downloaded from the Vorsis website or if a new GUI version was just installed, the VBN that was unpacked during its installation. Verify that the version number is correct in that it matches the version number listed in the documentation about the firmware 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.



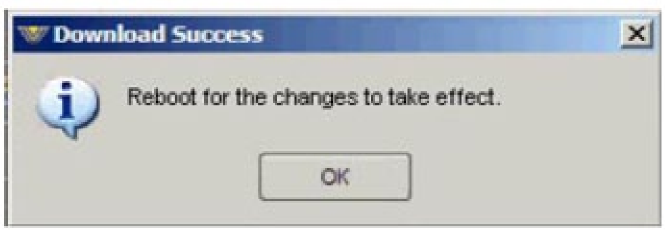
Recheck that the versions match the documentation! Then click *Yes*.

Progress boxes will pop up as the download progress occurs:





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



After AirAura reboots the new VBN code will be running on the motherboard.

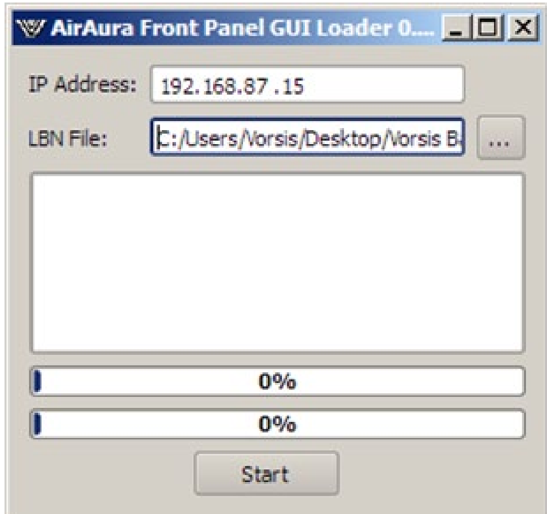
Front Panel LBN Updates

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. This information is available on the front panel of the unit itself. Use Right Click/ File/Settings on the front panel itself to locate this data.

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.

If the information is correct, select the *Open* option in the *Choose a file to download* window. The following dialog box will appear:



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 AirAura 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 completed.

You may also delay rebooting AirAura 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.

Presets Menu Items

The *Presets* menu tree may be accessed by right clicking anywhere on the main AirAura Control Panel.

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.

These dialogs replicate those accessible using the so-labeled buttons on the GUI itself. The right click menu structure is simply another way to get there from here...

Appendices

Appendix A

Appendix B

Appendix C

Appendix D

Appendix E

Appendix F

Appendix G

Appendix H

Appendix A

Contents

Wired Ethernet Configuration

Factory Default Configuration

AirAura requires *two* IP addresses – one serves the main DSP board and the other serves the microcontroller that operates the front panel. Ethernet was used for the DSP to front panel communications interface because of its speed over, for instance, a serial connection, a tremendous benefit when calculating and displaying the high resolution graphics required by the AirAura's front panel.

AirAura hardware is factory preconfigured to the following network settings:

Main DSP Board IP Address	192.168.1.200
Front Panel Controller IP Address	192.168.1.201
Gateway	192.168.1.1
Subnet Mask	255.255.255.0

Changing the AirAura's wired Ethernet interface is a simple process; however two considerations must be kept in mind when doing so.

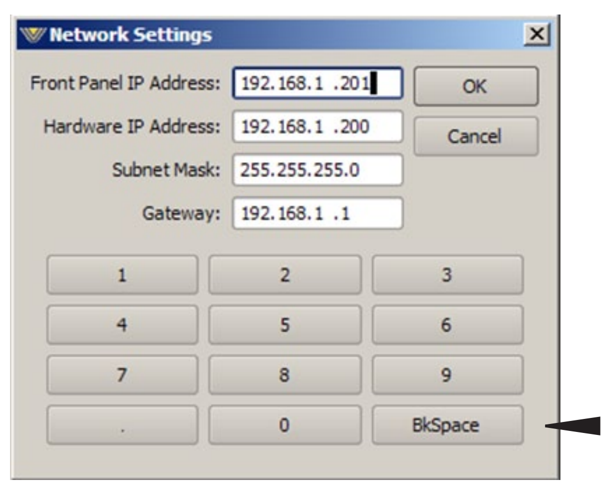
1. The Main DSP board and the front panel *must* be on the same network subnet. In other words, if the Main DSP board has been assigned an IP address of 109.16.77.200, then the front panel must also have an IP address assigned to it that is in the range <u>between</u> 109.16.77.1 and 109.16.77.254 in order to enable communications between them.

NOTE: We do not recommend using a **.0** or **.255** address!

2. The two rear panel RJ-45 connectors share the wired side of the wireless interface which is connected to the same internal Ethernet switch IC as the Main DSP board and the front panel. Therefore, if the Main DSP board and front panel IP addresses are changed, the <u>wired side</u> of the <u>wireless interface</u> must also be reconfigured in order for wireless to work. Please see **Appendix D** for the procedure for reconfiguring the wireless interface.

Changing the Configuration

To change the IP address and network configuration of the AirAura's Main DSP board and front panel interface the features built into the front panel may be used. Using the front panel trackpad place the cursor on the left display. Then using the rightmost illuminated front panel "V" button click the button to bring up the View/Hardware dialog.



Within that dialog, select the Hardware/Network Settings option and the dialog box shown on the right will appear.

Next, using the *BkSpace* (backspace) button of the numeric keypad, backspace over any unneeded entries in the *Front Panel IP Address* field.

Next, using the numeric keys, enter the new IP address for the front panel, noting that we usually assign it the next highest IP address above the hardware address though this isn't really necessary as long as the two addresses used are in the same network addressing range.

Next, use the backspace key to erase any undesired characters in the *Hardware IP Address* field. Then, using the numeric keys, enter the new Hardware IP Address.

Use the same procedure to reset the Subnet Mask and Gateway settings to those appropriate for your network.

Once this has been done, click on the *OK* button and the new settings will be stored by the AirAura. When the "You must reboot for the changes to take effect" message appears, wait 15 seconds and then reboot the AirAura by removing its AC power for a few seconds and then reconnecting it. Once AirAura finishes booting up it will be operating on the new network settings.

If, after the AirAura boots up the front panel is not showing activity, recheck the network settings to ensure that they are what you expect them to be. If the Main DSP board and front panel are on different network address ranges they will *not* be able to communicate.

Appendix B

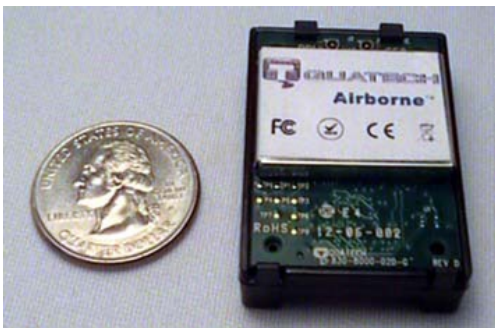
Contents

Wireless Ethernet Configuration

Introduction

AirAura utilizes an industrial grade off-the-shelf hardware solution for its wireless interface. The benefits of using a ready-made wireless device instead of 'rolling our own' are many, the most salient of them is us being released from the tedious task (and high cost) of having an RF-based design tested and approved by myriad various worldwide regulatory agencies.

The device we chose for AirAura is manufactured by Quatech of Hudson, Ohio USA, model number WLNG-ET-DP501 and is shown in the photo below.



This device is a wired to wireless Ethernet bridge. Ethernet bridges incorporate Network Address Translation, or NAT, into their router schemes in order to be able to sort out traffic to/from the two sides of the bridge. The module is compliant with NAT3 standards.

NAT *requires* that the IP addresses on each side of the interface be *different* so that it knows what data to forward to which port. Note that the factory default IP addresses for the remote GUI are *different* for AirAura's wired and wireless interfaces. According to NAT rules, this is a *must*!

The wireless interface is equipped with two diversity antennas in order to provide maximum operational range. The antennas supplied with the AirAura product are type approved for use with the module.

The module supports worldwide WiFi frequency bands and channel assignments. The frequency band to use can be configured within the web-based configuration dialog by selecting the region where the module will be used.

Selecting a region automatically brings up the correct frequencies and channels approved for use in that region.

As shipped from our factory the module is configured to use the United States approved frequency bands but may be factory configured to utilize other bands upon request.

Factory Default Configuration

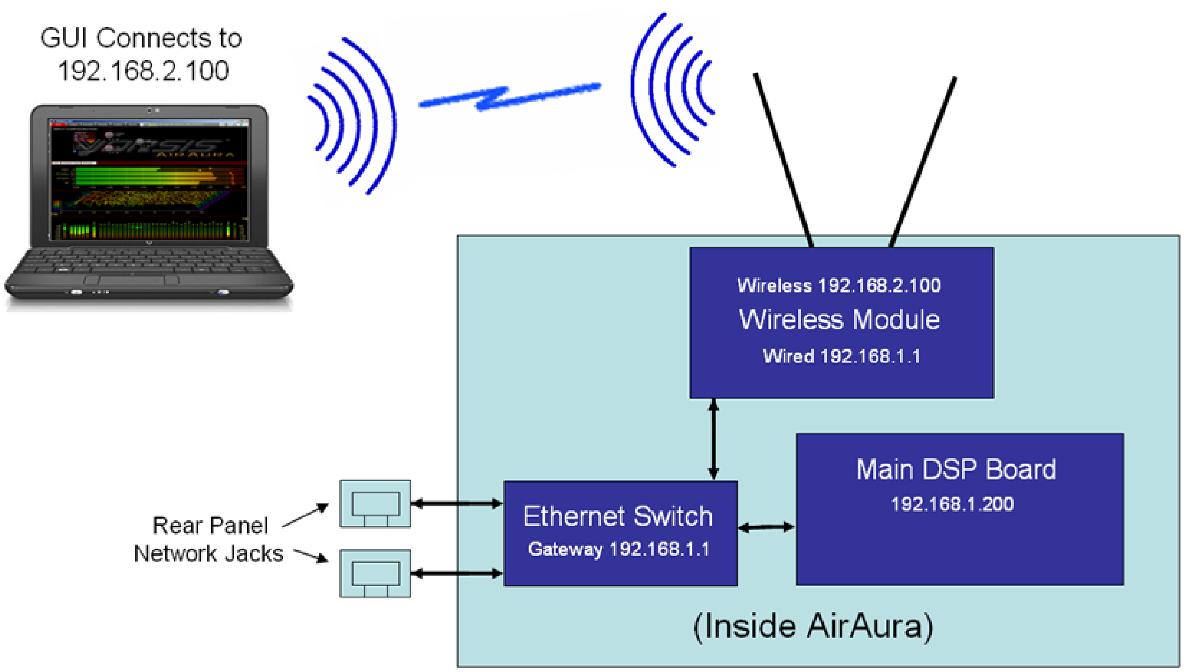
The following are the Vorsis factory defaults for the most important parameters of the module's basic configuration.

Wireless LAN Connection Type:	AdHoc
Wireless LAN Channel:	1
SSID:	AIRAURA
Wireless Data Rate:	Auto
Wireless LAN Region:	United States
WLAN Security:	Disabled
WLAN DHCP:	Disabled
WLAN Static IP Address:	192.168.2.100

WLAN Subnet Mask:	255.255.255.0
WLAN Gateway Address:	192.168.2.110
Ethernet DHCP:	Disabled
Ethernet Static IP Address:	192.168.1.200
Ethernet Subnet Mask:	255.255.255.0
Ethernet Gateway IP Address:	192.168.1.1 ** (see notes)
Ethernet Role:	Router
MAC Cloning:	Disable
Ethernet Port Speed/Duplex:	Auto-negotiate

The block diagram shown below illustrates how the remote GUI and AirAura communicate with each other via the wireless interface.

AirAura Wireless Network
Showing Factory Default Configuration



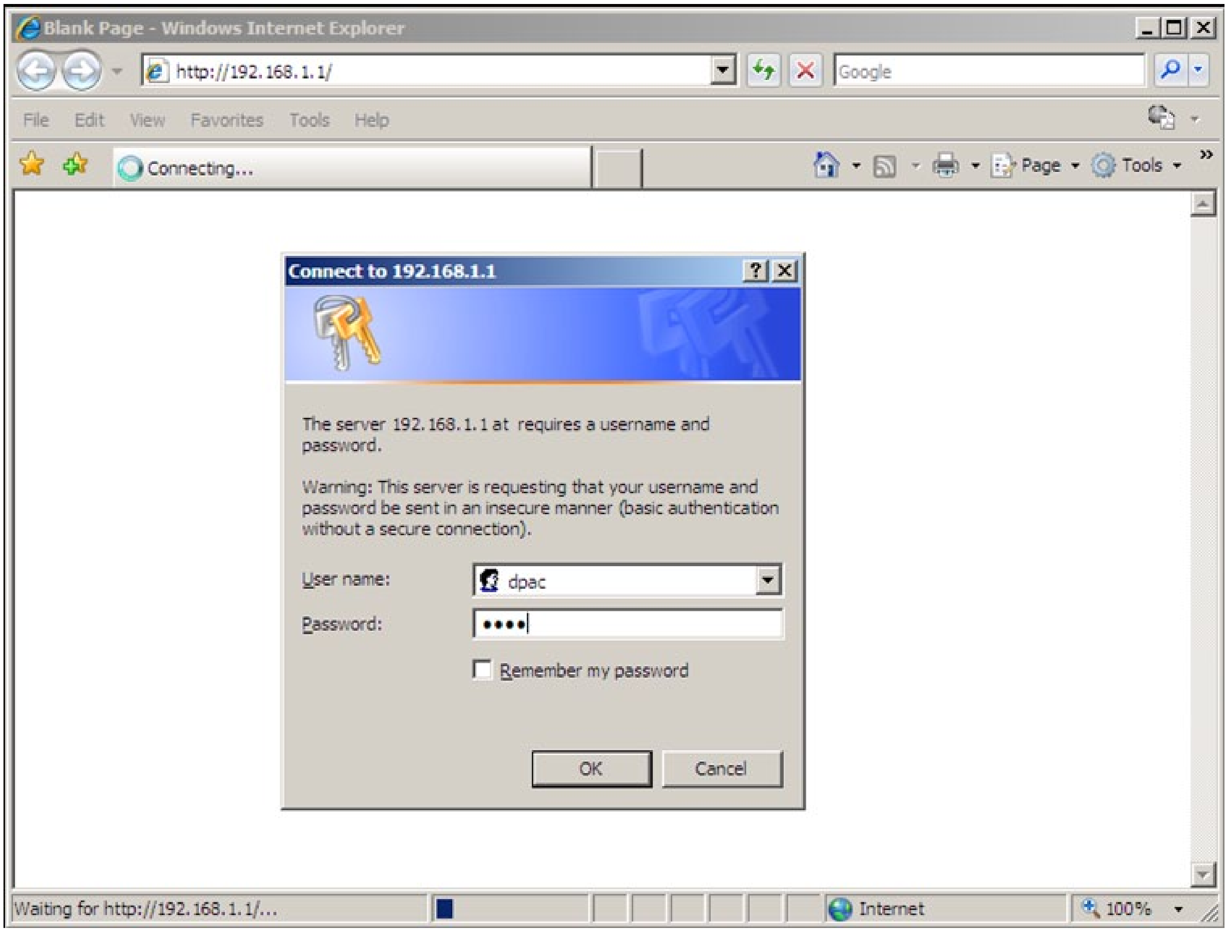
In most circumstances it is not necessary to reconfigure the wireless side address (the 192.168.2.100 address that the remote GUI uses to connect via wireless) because it is the wired side of the interface that is normally attached to a station's local area network.

An important concept to remember about how the wireless module functions is that the Ethernet IP address of the module isn't its IP address – it's the IP address to which packets received on the wireless side are forwarded to.

Also, the Gateway address of the module's wired side is the IP address that a web browser should be pointed in order to log in and modify the module's networking configuration.** (see notes)

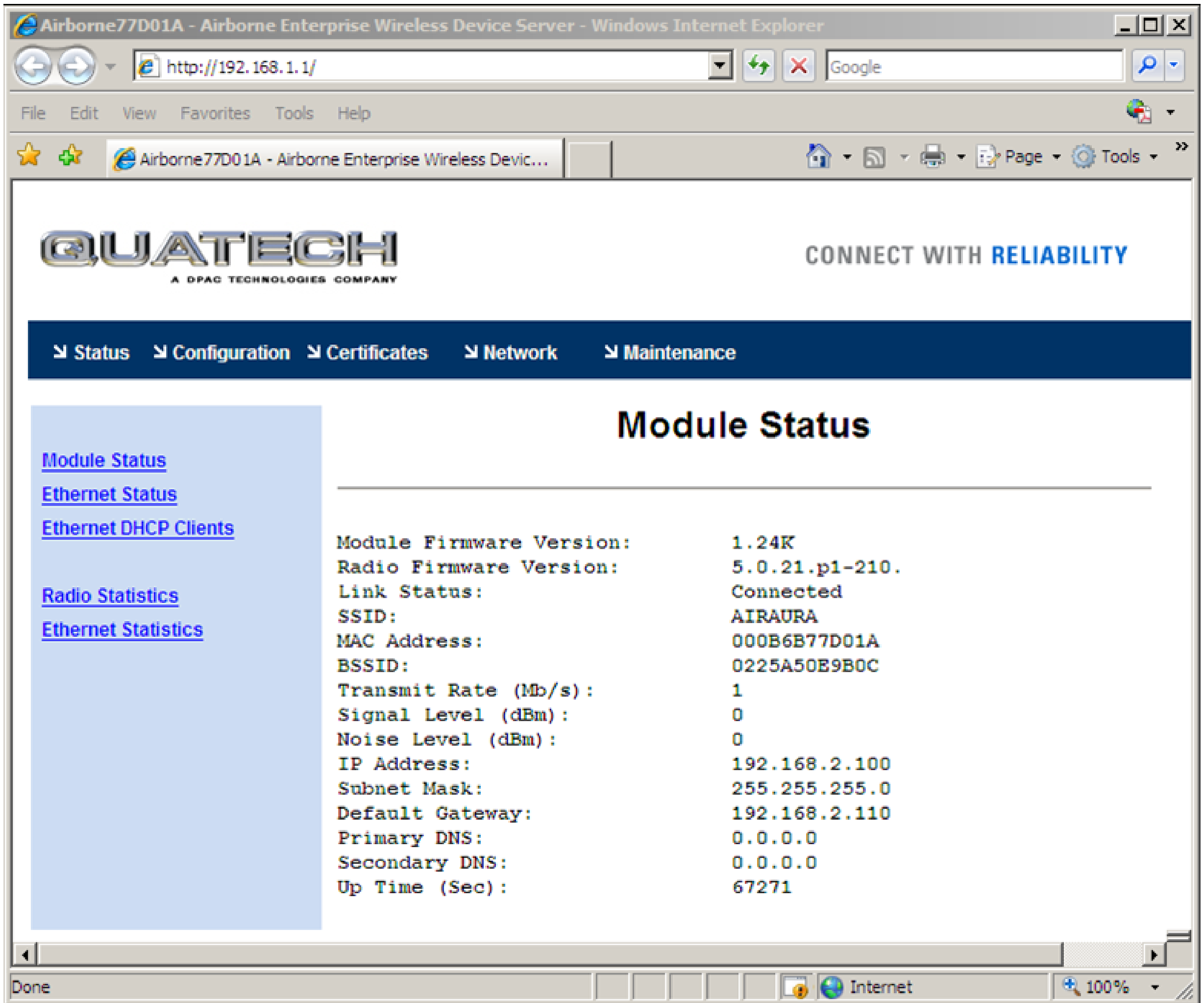
Changing the Configuration

With the AirAura directly connected to a PC that is not connected to any external wired networks, open a web browser and enter the IP address of 192.168.1.1 in the URL address field (see below).



At the login prompt, enter the credentials "dpac" for the User name and "dpac" for the Password – both are lowercase.

Once valid credentials are entered and the *OK* button is pressed, a module status page should appear:



In the selection bar above the words *Module Status* are options for investigating or changing the characteristics of the module. We are concerned only with the option called *Configuration* at this time. Click this item to select it.

When the configuration page loads, select the item called *Network Settings* in the left column. Also, unless security or the operational region needs to be changed we are only concerned with the following entries on the *Network Settings* page – the Wheatstone factory defaults are shown here.

WLAN Static IP Address:	192.168.2.100
WLAN Subnet Mask:	255.255.255.0
WLAN Gateway Address:	192.168.2.110
Ethernet Static IP Address:	192.168.1.200
Ethernet Subnet Mask:	255.255.255.0
Ethernet Gateway IP Address:	192.168.1.1 ** (see notes)

If the IP addresses of the AirAura are modified to be compatible with the local network and that IP address scheme is different from our factory defaults, the IP address of the wired side of the wireless module must also be changed.

For instance, if the AirAura's new Hardware IP Address and Gateway IP Address are 66.23.77.40 and 66.23.77.1, respectively, then the entry for the Ethernet Static IP Address should be changed to 66.23.77.40 and the Gateway IP Address changed to 66.23.77.1** (see notes). Remember, the module's Ethernet Static IP Address isn't really the module's static IP address; it is the IP address where packets received on the wireless side of the module will be forwarded.

*Please keep in mind that whatever the new Ethernet Gateway Address is, that is the address a web browser must now be pointed to in order to reach the module's configuration interface once it is restarted and the new configuration is in place.** (see notes)*

Once changes have been made, scroll down to the bottom of the Network Settings page and click on the *Commit* button. Then, in order for the new data to be active the module requires a reboot. You can either reboot the AirAura itself, or if that is inconvenient, click on the *Maintenance* tab of the configuration screen (at the top center of the page) and when that page loads, click on the *Restart Module* option in the left column of choices.

Note that once the module has been instructed to restart, the browser will lose contact with the module until it is up and running again. Also, if the Gateway address has been changed the browser will need to be pointed to that new address in order to reach the configuration page.

This is important! If changing the module's configuration moves it off of the network address range that the host PC has been using to configure the module, the host PC's IP address must *also* be modified in order to put them both on the same network address range again and re-enable communications.

If difficulties arise with the module configuration, please contact Wheatstone Customer Support at (252) 638-7000 or email them at techsupport@wheatstone.com.

NOTES:

** Using a Gateway address on the wired side of the WLAN module can cause a local connected network to become confused if it and the gateway addresses of the module are the same. It is sometimes best to program the WLAN module's Gateway address to an unused address on the locally connected network. If an invalid entry is used for the WLAN module Gateway that address can no longer be used for the module's web-based configuration and the wireless address will need to be used instead.

Appendix C

Contents

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 AirAura'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^{th} Order All-pass

Auto Mono

Auto Mono Modes:	In/Bypass
Auto Mono Threshold:	User Adjustable from -40.0dBFS to 0.0dBFS
Timing:	Fully Program Dependent

High-Pass Filter

HPF Filter Class:	24dB/octave Butterworth
Frequency range:	20Hz – 300Hz
HPF Modes:	Bypass / Stereo / L/R (Sum/Difference)

Parametric Equalizer

Sections:	Four - Identical
Frequency Range:	20Hz – 20kHz
Bandwidth Range:	0.2 – 3.0 octaves
Boost/Cut Range:	+/- 14.0dB
Options:	Bypass (individual bands or all bands)

5 Band AGC/Compressor with SST

AirAura's 5 Band AGC/Compressor operates in sum and difference (M/S) mode.

Drive Gain Range:	-12.0dB to +6.0dB in 0.5dB steps
Makeup Gain Range:	+24.0dB to +36.0dB, 0.5dB steps
AGC/Compressor Thresholds:	- 50dBFS to -70dBFS
AGC Threshold Backoff:	0dB to -6dB
AGC Attack:	50mS to 500ms
AGC Release:	1 second to 7 seconds
AGC Band Coupling:	0dB to -6.0dB in 0.5dB steps, Mid to High & Mid to Low
Compressor Attack:	3.0mS to 100mS
Compressor Release:	50mS to 1000mS (1Sec)
Compression Ratio:	1:1 to 20:1
L+R Mixer:	Boost/Cut +/-6dB
L-R Mixer:	Boost/Cut +/-6dB
Stereo Enhance:	0dB to +6.0dB (also offset by L-R mixer settings)

Gating Mode:	Ooze/Freeze
Gate Threshold Range:	-20.0dB to -79.5dB, plus OFF
SST Drive:	+30.0dB to +40.0dB in 0.5dB steps
SST Ramp Rate:	2 to 7.0 seconds
SST Gate Threshold:	-6.0dB to -70.0dB in 0.5dB steps
SST Window Size:	3dB to 12.0dB
SST Operating Style:	Automatic, Linear or Adaptive

Crossover Frequencies

Super Low to Low Band:	20.0 Hz to 149 Hz
Low to Low Mid Band:	162 Hz to 687 Hz
Low Mid Band to High Mid Band:	728 Hz to 2.18 kHz
High Mid Band to High Band:	2.59 kHz to 20.0 kHz
Band Output Trims, Sum:	+/- 6dB
Band Output Trims, Difference:	+/- 6dB

31-Band HD Peak Limiter

A four band parametric EQ follows the 31Band limiter.

Multiband (31Band) Drive:	80 – 100%
VBMS Drive:	- 6.0dB to +6.0dB
VBMS Style:	Off/ Soft/ Hard
VBMS Output:	- 6.0dB to +6.0dB
VBMS Frequency:	60 - 300Hz
VBMS Sub:	Bass Sub-harmonic Support to 20Hz
Multiband Limiter Knee Shape:	Soft/ Hard
Multiband Threshold Range:	+/-6dB relative to Multiband Drive setting
Multiband Attack Time Range:	0.3ms to 100ms
Multiband Release Time Range:	3.0ms to 100ms
Lookahead Limiter Threshold Trim:	+/- 6.0dB
Lookahead Limiter Attack Time Range:	0.2ms – 30ms
Lookahead Limiter Release Time Range:	33ms – 300ms
Lookahead Limiter Delayed Release Range:	100ms – 1000ms (1Sec)
Clip Style:	Off/Round/Deep
HD Output Level Trim:	-79.95dBFS to 0.0dBFS, or OFF
Phase and Polarity:	Left and Right channels may be phase-reversed or swapped

31 Band FM Peak Limiter

A four band parametric EQ follows the 31-band limiter.

Multiband/Drive:	80 – 100%
VBMS Drive:	- 6.0dB to +6.0dB
VBMS Style:	Off/ Soft/ Hard
VBMS Output:	- 6.0dB to +6.0dB
VBMS Frequency:	60Hz - 300Hz
VBMS Sub:	Bass Sub-harmonic Support to 20Hz
Multiband Knee:	Soft/ Hard
Multiband Mode:	Discrete: 31-band “Fine Grain” Limiter™ Wide: 10-band “Triplet Coupled” Limiter™ Timbral: 10-band “Harmonic Coupled” Limiter™
Multiband Threshold:	+/-6dB relative to Drive
Multiband Attack:	10ms to 100ms (this is not a misprint)
Multiband Release:	3ms to 100ms (this is not a misprint)
FM Pre-emphasis Modes:	Flat/50μS/75μS
FM Pre-emphasis Location:	Pre 31 Band Limiter/Post 31 Band Limiter
FM Clipper Styles:	Hard/Firm/Round
Lookahead Limiter Threshold Trim:	+/- 6.0dB
Lookahead Limiter Attack:	0.2ms – 30ms
Lookahead Limiter Release:	33ms – 300ms
Lookahead Limiter Delayed Release:	100ms – 1000ms (1Sec)
FM Diversity Delay:	OFF to 10 seconds, 100μS increments
FM Digital Output Trim Level:	-79.95dB to +12.00dB, or OFF
FM Digital Output Options:	Pre-Delay and /or De-Emphasized

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 10% 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)

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 – 20 kHz *
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 75μS pre and de-emphasis

All specifications subject to improvement or change without notice.

Preset Management

AirAura's advanced preset management system allows the creation, storing, and recall of 80 presets within AirAura itself and an unlimited number may be stored on the Windows PC hosting the Guru or Professional 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 AirAura'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 AirAura contains a five-port, auto-sensing 10/100BaseT Ethernet switch with two of the ports 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 AirAura.

802.11g/b Wireless

AirAura has its own built-in wireless interface which allows easy control via the GUI software and a laptop equipped with a wireless interface.

Software Remote Control

The supplied Windows®-based AirAura Guru GUI software affords simplified access to AirAura'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 D

Contents

Wireless Module Approvals and Miscellaneous Data

AirAura's 802.11/a/g wireless interface is a high quality, off-the-shelf product manufactured by B&B Electronics, formerly Quatech of Hudson, Ohio USA.

Complete documentation for the module including very detailed configuration options can be found in the Airborne Wireless Ethernet Bridge Modules section of the Quatech website:

http://www.quatech.com/catalog/airborne_ethernet_modules.php

The wireless module is type approved in the US, Canada, and Europe, with certification pending in Japan (as of this writing). To comply with the regulations dealing with embedded wireless devices, the rear panel of the AirAura is so marked as required by law:

Using FFC grant: F4AWLNG1

Using IC grant: 3913A-WLNG1

B&B/Quatech contact information is below:

Hudson, OH Office:
B&B Electronics Mfg. Co.
5675 Hudson Industrial Pkwy
Hudson, OH 44236
Phone: +1 815.433.5100
Fax: +1 815.433.5109
Email: wirelesssupport@quatech.com
Web Site: www.quatech.com

Typical Usable Wireless Range

The following table illustrates the typical data rates, performance and range of the wireless module that can be expected when using the Omni-directional diversity antennas supplied with AirAura.

Data Rate	Typical Outdoor Range Unity Gain Antennas	Typical Outdoor Range 2dBi Antennas Each End
1.0 Mb/s	240m / 787 ft	380m / 1247 ft
11.0 Mb/s	135m / 443 ft	215m / 705 ft
6 Mb/s 802.11/g	135m / 443 ft	215m / 705 ft
6 Mb/s 802.11/a	49m / 160 ft	155m / 508 ft
54 Mb/s 802.11/g	12m / 39 ft	19m / 62 ft
54 Mb/s 802.11/a	4.5m / 15 ft	14m / 46 ft

Ranges are based on a combination of receiver sensitivity, transmitter power, free-space path loss, antenna gain factors, and link margin. Actual range will almost always vary from those stated and be less. Non-line-of-site applications will result in values even less than shown in the above table.

The listed "Data Rate" is the supported *connection rate* for the wireless link including handshaking. Because handshaking (to ensure reliable communications) utilizes a small portion of the stated data rate, the actual data throughput for the link to/from the AirAura and its GUI will be somewhat less than that stated in the first column.

IMPORTANT NOTE

AirAura utilizes an efficient scheme to send data to and from the remote GUI. This data rate is *far less* than the 1Mb/s rate that we've configured in the module as the factory default.

There is NO advantage to forcing the module to communicate using higher data rates. In fact, the above table should make it quite clear that exactly the opposite is true – "forced" high speeds can result in an unreliable connection, even at very modest distances.

Several years of field experience with AirAura's wireless interface operating in all kinds of situations has confirmed that the 1Mb/s data rate chosen as our factory default data rate provides *very good* coverage and *excellent* usability.

Transmitter Power

Transmit power is automatically managed by the device for best range and minimum power consumption. The maximum transmit power available at the module's RF connectors is typically +15dBm ± 2dB for B-Mode (all rates) and +12dBm +/-2dB for G-Mode (all rates). The RF power at AirAura's rear panel WiFi connectors will be somewhat less than this due to losses in the short but nonetheless not lossless interconnecting coaxial cables.

Antennas

Any customer supplied antenna used with AirAura must be designed for operation within the 2.4GHz ISM band and more specifically, must support the 2.412GHz to 2.482GHz bands for 802.11b/g operation. Antennas are required to have a VSWR of 2:1 maximum, referenced to the 50Ω system impedance. The antennas supplied with AirAura have been certified for use with the wireless module.

Performance

Wireless "performance" in an actual application is a very difficult parameter to define, mainly because the appropriate metric changes with each operational situation. Performance is always a combination of parameters, some of which (such as natural obstructions) are not usually under the control of the AirAura customer.

The underlying characteristic that needs to be observed for best performance is the link quality, not link speed. Link quality can be defined as "the bandwidth available over which communication between the AirAura and the host GUI can efficiently occur." The lower the link quality the less likely the two can communicate.

Appendix E

Contents

AirAura GPIO Interface

AirAura is equipped 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 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.

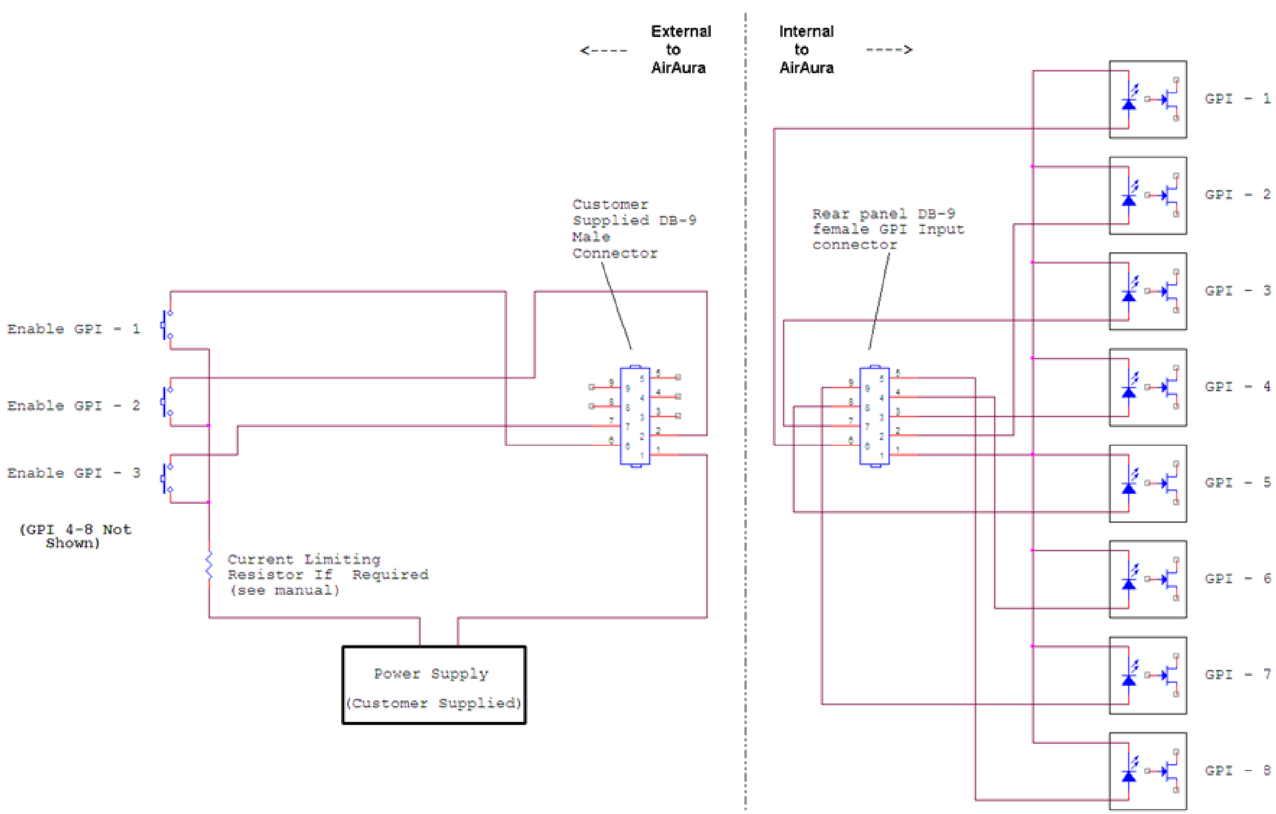
Each of the GPI inputs is current limited by a 475 ohm internal resistance. The LED device inside the optoisolators is rated at a maximum forward current of 50mA and a normal operating current of 5mA.

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

Note: GPO circuitry is enabled on Version 1.2.x software and above.

AirAura 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 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 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).

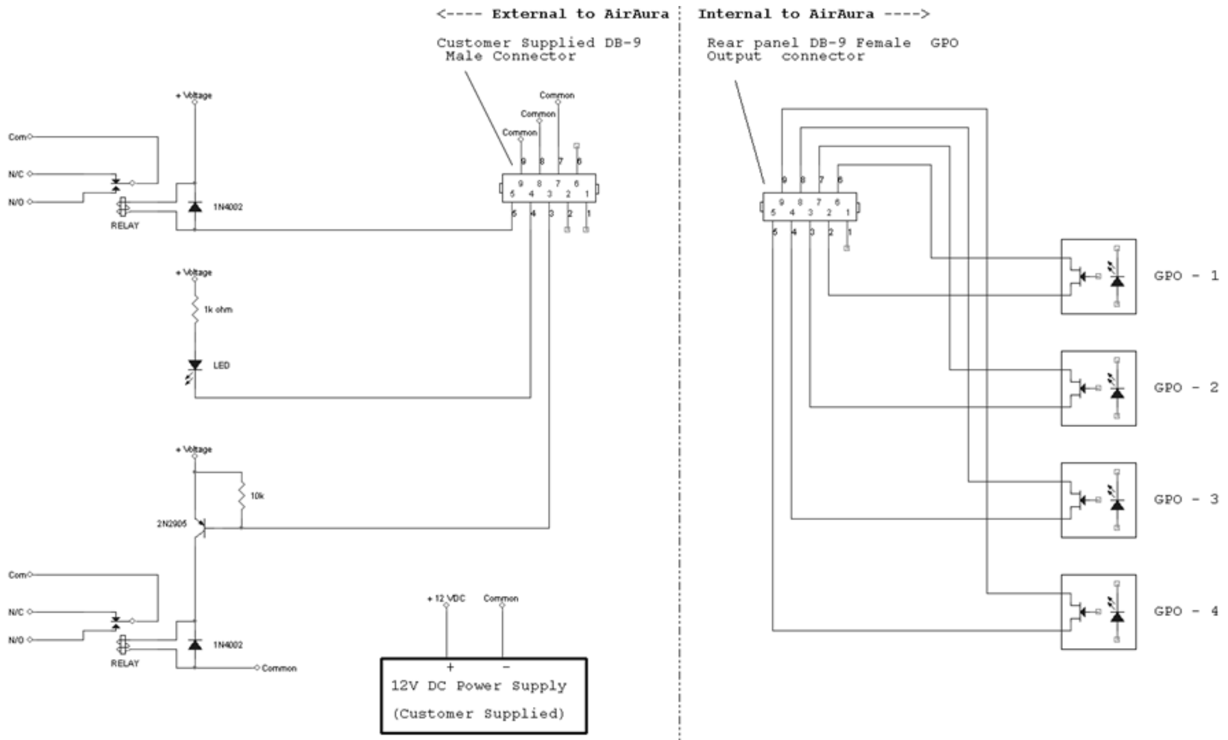
GP Output #3 – Becomes enabled upon a General System Failure.

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 AirAura'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 AirAura, 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 F

Contents

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?

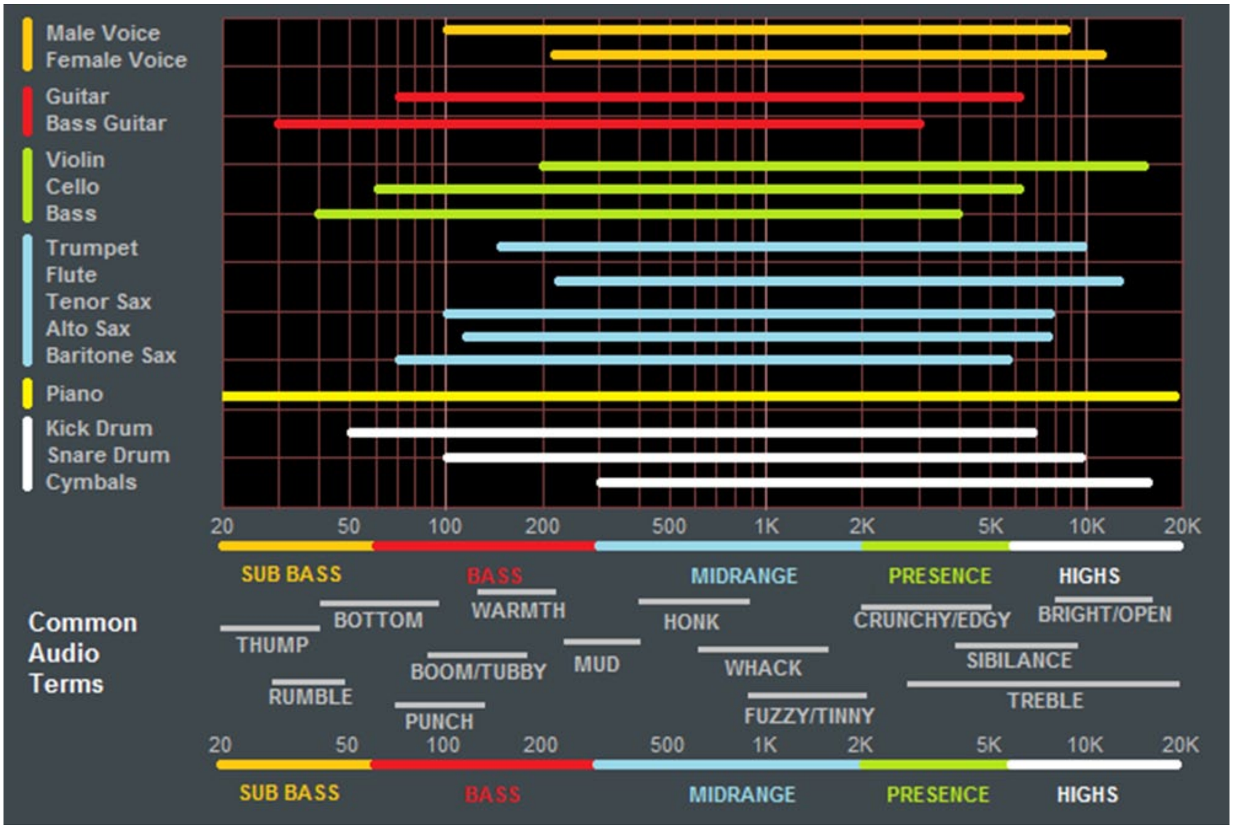
Lots of "audio" terms regarding audio processing have been invented and bandied about over the years. Ever since radio stations started employing circuits instead of humans to control the modulation of transmitters we've tried to come up with words to describe the effects of processing and what we hear. 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

Much of the credit for the vast sound-tailoring flexibility and incredibly clean sound of Wheatstone's AirAura processor goes to its 31-band limiter. A 31-band limiter can accomplish two goals that no other limiter with fewer bands can. The first is that it allows the end user to achieve a great deal of loudness without hearing individual limiter bands work. This allows the sound to be very "open" (there's a term!) and loud, and clean, and without the harsh, unnatural re-equalization generated by limiters with fewer bands – the ones found in conventional processors.

The second goal was to create a unique limiter technology that didn't get in the way of end users using the vast pallet of powerful sound shaping tools to tailor their station's sound. We didn't want the limiters getting in the way of that enhancement.

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 "VBMS™," which stands for "Vorsis Bass Management System™." In general terms VBMS gets real bass on the air without it causing the problems commonly associated with "bass clippers." Users of AirAura version 2.4.x and higher have an even more powerful set of features called "Vorsis Bass Tools™," the natural extension of our acclaimed "VBMS™" technology.

Using VBMS™ or Vorsis Bass Tools™ to add more Thump & Rumble is quite easy. By setting the VBMS 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.

Vorsis Bass Tools™ in software versions above 2.4.x has a new feature called "Sub Bass." This new algorithm places targeted enhancement on the frequencies between 20Hz and 80Hz, and, unlike the VBMS algorithm, it 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 31-band limiter and VBMS/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 of 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'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 <u>L-R mixer</u> 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 is much more forgiving in this regard and most users have an easier time being loud with AirAura 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 or give is 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 G

Contents

Using The Factory Lock Utility

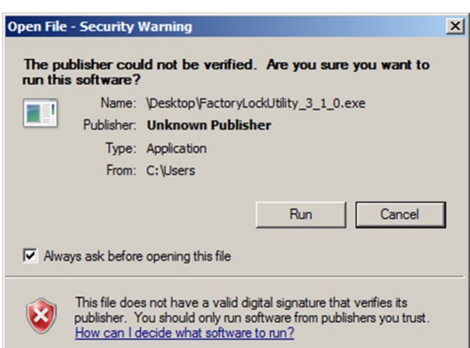
Background

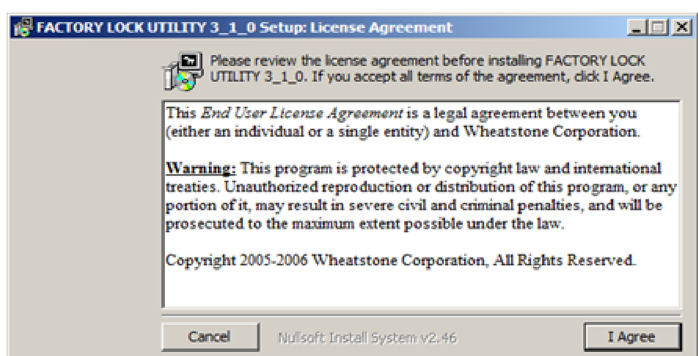
The Factory Lock utility tool was originally designed for audio processing consultants who use and recommend our processors to their clients. The utility allows them to unlock our factory presets, create and "factory lock" their own presets or even remove all but their own custom presets from a Audio Processor. Those consultants often make custom presets for their clients and after saving them to the processor hardware they then Factory Lock them. Sometimes they even remove all other presets from the processor but theirs which is their way of ensuring that no one tampers with the work they've put into giving the station the sound desired by management – the sound that management has paid those consultants to create!

As installed, Wheatstone GUI's offer medium-level preset security, meaning that any preset locked by one user can be unlocked by another. On the other hand, when presets are Factory Locked the GUI is not normally allowed to overwrite or delete those presets from the processor's hardware because the GUI's preset library utility does not have *permission* to remove or modify those presets. However, by giving the GUI some special "permissions" it is able to factory lock *and* factory unlock *any* preset regardless of how it was locked or who locked it. This is precisely what the Factory Lock utility does – it grants the user's GUI the special permissions that it needs to enable it to create, save and lock presets *at the Factory level*. Because the utility is also useful when updating factory presets in the hardware, we've make the utility also available to customers.

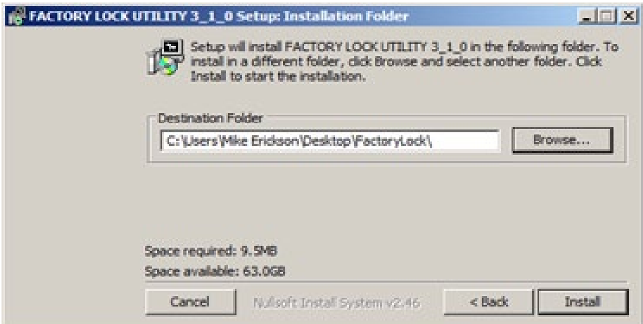
Installation

To install the Factory Lock utility double click on the *FactoryLockUtility_3_1_0.exe* icon to start the program. If you are a Windows Vista or Windows 7 user you may see a message that looks like the following, noting that it is acceptable to uncheck the *Always ask before opening this file* and clicking on the *Run* option.

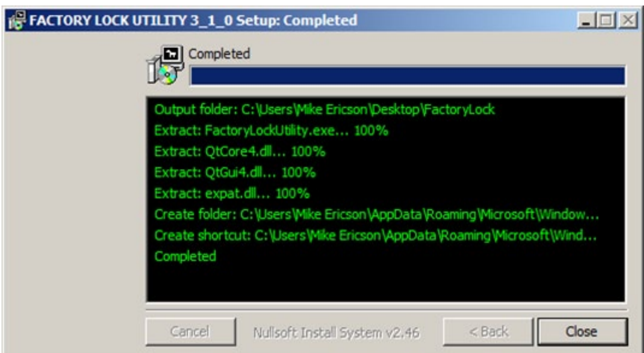




Next up is the License Agreement prompt. Please select the *I Agree* option to begin the installation process.



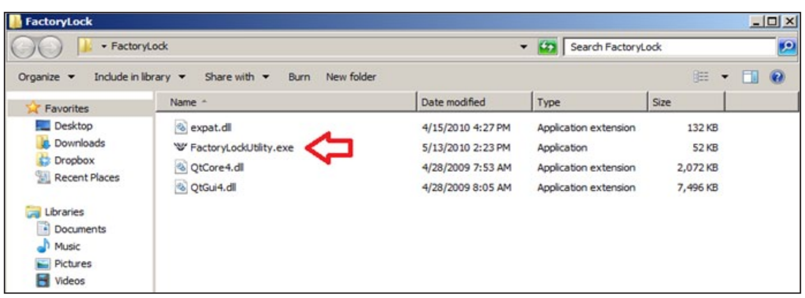
The next prompt will ask where you would like to install the utility on your computer. Please either accept the installation default or navigate to the location where you would prefer it to be installed by using the Browse button to open the Browse for Folder Windows dialog.



Next, an installation progress screen will appear. The entire installation process should be completed within a few seconds depending on the speed of the host computer.

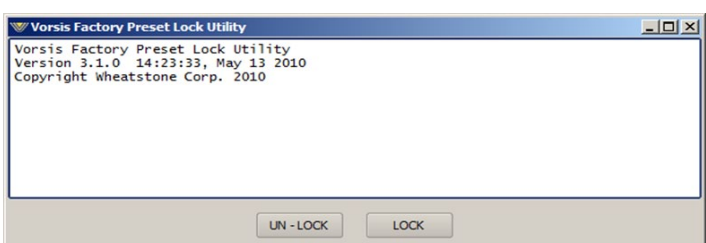
Using The Utility

STOP! ***<u>ALL WHEATSTONE PROCESSING GUI'S MUST BE CLOSED BEFORE RUNNING THIS UTILITY!</u>***

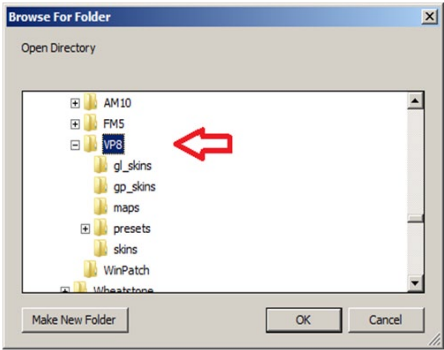


There are four files installed into the *Factory Lock* utility's folder. The only one you should be concerned with is the actual "FactoryLockUtility.exe" file.

Double click on this file to start the utility and the Lock Utility choice menu will appear.



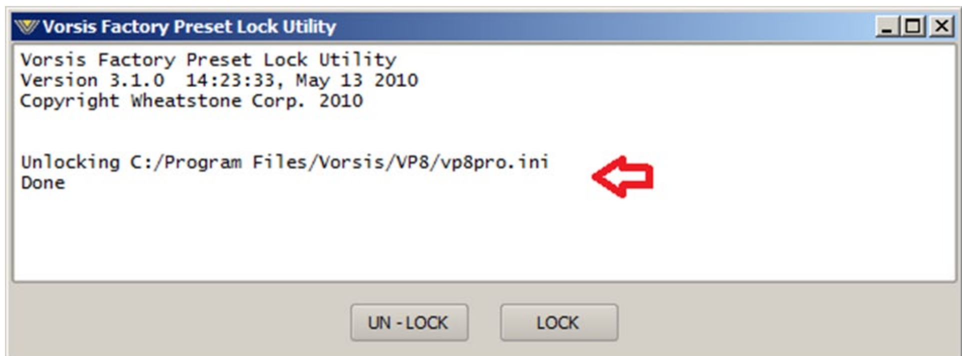
In order to grant your GUI the Factory Lock privileges click on the *UN–LOCK* button. A new navigation box will open and if the Wheatstone processor GUI has been installed in its default location the path you want to point the utility to is: "...Program Files\Vorsis\Processor Name".



Once you have reached the folder with the name of the processor, STOP!

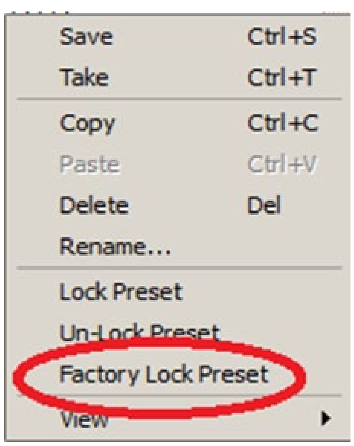
Click *OK* to proceed.

The utility's progress window should now "Done" and the Factory Lock options are now enabled.



Confirming Factory Lock/Unlock Is Operational

Now it is safe to restart the GUI, so please open it. When the Library button is clicked the Library dialog will open as before. Highlighting a preset in the list and right clicking on it should reveal that there is a new option in the list; Factory Lock Preset. Whenever Factory Lock Preset is visible, the GUI also has permission to unlock a Factory preset.



The three "Lock" options now available are:

Lock Preset – User locks the preset (BLUE);

Un-Lock Preset – Unlocks user (BLUE) and factory locked presets (RED);

Factory Lock Preset – Factory locks or unlocks any preset.

Note that Factory locked presets are shown in RED, user locked presets are shown in BLUE, and unlocked presets are shown in BLACK.

Removing The Factory Lock/Unlock Options

It is sometimes useful to undo the Factory Lock options in order to prevent inadvertent access or preset permissions changes. The Factory Lock utility is used once again, but this time we tell it to remove the Factory preset permissions from the GUI. To do this, run the utility again, pointing it to the same GUI installation folder used previously, and instead of clicking on *UN-LOCK*, click on *LOCK* instead. Remember again that the GUI must not be running or the utility will not be able to modify its permissions!

If Factory Lock permissions have been removed, opening the Library dialog and right clicking on a preset should reveal that the Factory Lock Preset option is now gone, as shown at left.

Appendix H

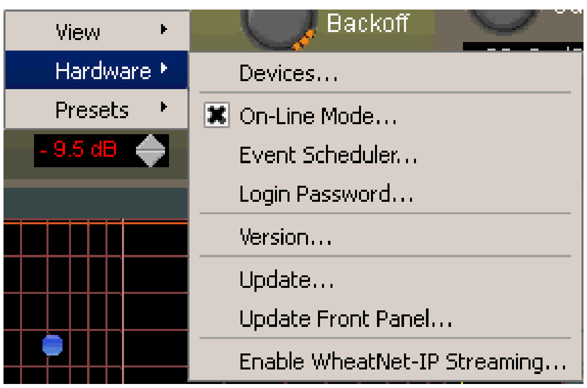
Contents

WheatNet-IP Configuration

Along with Digital AES3 and Analog inputs AirAura 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 AirAura'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 AirAura.

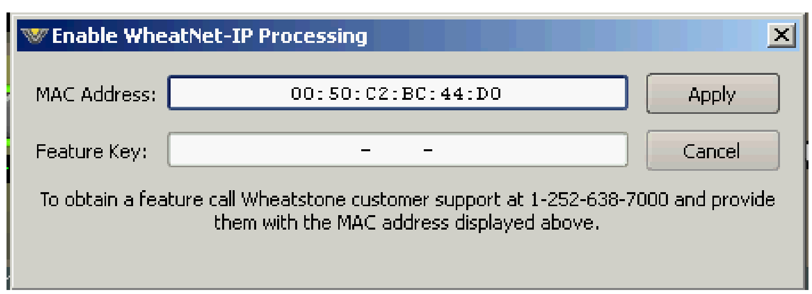
This procedure assumes that your facility is equipped with at least one Wheatstone Blade and the associated Navigator software. The Blade used may even be the multiband, eight processor Aura8ip Processing Blade whose physical inputs and outputs (half analog, half digital) may even be used as a substitute for AirAura's physical I/O.

Out of the box, WheatNet-IP is not factory enabled. A special passcode must be obtained from Wheatstone Technical Support to turn it on. When calling to get WheatNet-IP turned on, please have the MAC address of the unit handy because that address is required in order to generate the correct WheatNet-IP enable code for your hardware. Each AirAura has its own unique WheatNet-IP enable code, so providing us with the correct MAC address is crucial.

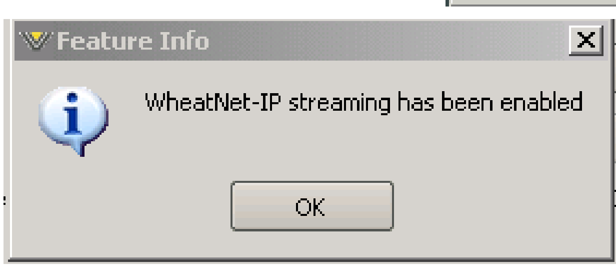


With the GURU or Professional GUI configured to connect to the AirAura device and online to it, right clicking on the controls area will reveal the dropdown menu at left. We will be navigating the Hardware menu option.

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 AirAura 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.

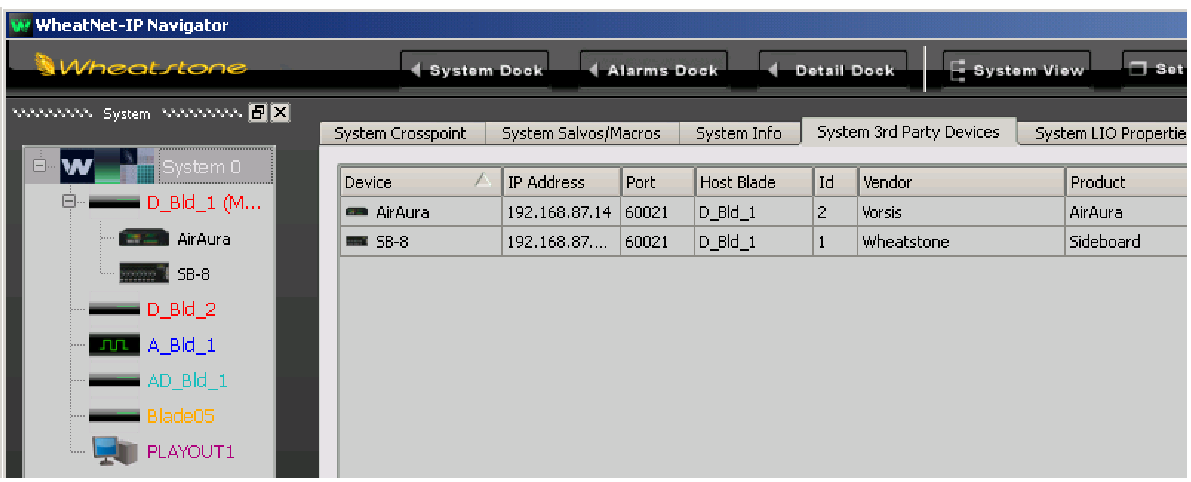


Now it's time to open up WheatNet-IP Navigator to gain access to AirAura's WheatNet-IP-associated input and output signals.

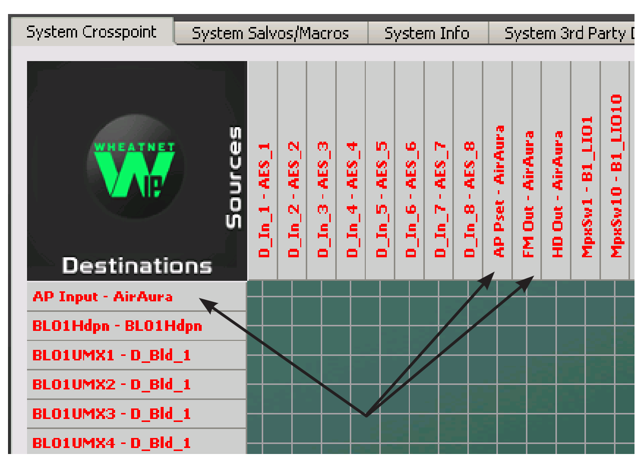
Once the WheatNet-IP Navigator software is open and you are logged into the system we must create a new device for AirAura. Because AirAura is not a "Blade" it appears and operates as a 'third party device' within the WheatNet-IP system. Our Sideboard, an outboard mixer/controller for use with any Blade's pair of 8-channel utility mixers is one example of a third party device. Another example of a third party 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 AirAura to The WheatNet-IP System

The first thing to do is click on Navigator's *System Third Party Devices* tab which will open a new window listing any third party devices currently present. In the image below an AirAura 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 AirAura may be added to the system. Here we will add the new AirAura'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 AirAura's Input and FM and HD output signals are now present and available for routing.

Note that as shown in the example at left we've used Navigator's signal name editing capability to assign AirAura'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 System product documentation.

Assistance in configuring AirAura for WheatNet-IP may be obtained by giving us a call at (252) 638-7000 or by emailing techsupport@wheatstone.com.