FM-531HD

FM & HD DIGITAL AUDIO PROCESSOR

TECHNICAL MANUAL

Wheatstone Corporation

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

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VORSIS ULTRA-HIGH RESOLUTION PROCESSING

FM-531HD FM & HD Digital Augio Processor Technical Manual - *Software Version 2.0.0 and higher*

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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.

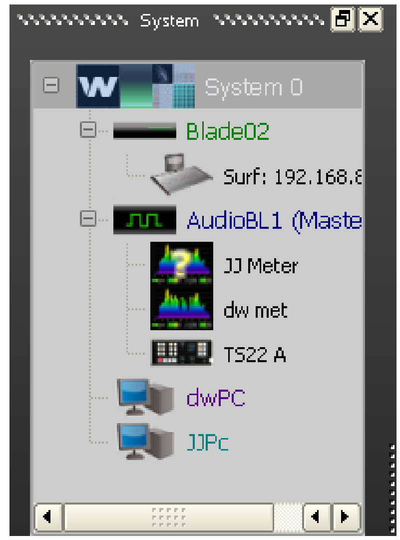
Caution! ***Any modifications not expressly approved in writing by Wheatstone could void the user's authority to operate this equipment.***

Adding Devices To The 3rd Party Tab

In order to utilize all the features of the FM-531HD Digital Audio Processor the device must be added to the *System 3rd Party Devices* tab in the Wheatstone WheatNet-IP Navigator program (aka the Navigator GUI). This sheet shows you the basics of that procedure. Refer to the *WheatNet-IP Audio Over IP Network Technical Manual* for additional details.

You will need to know the IP address of the device being added, so you will want to find that out before you start.

Launch the Navigator GUI and make sure that **System 0** is selected in the *System* pane. You will see something like this:



Now, to the right of the *System* pane, select the *System 3rd Party Devices* tab.

Click the *Add* button to bring up the *Add 3rd Party Device* dialog:

Type in a convenient *Name* and insert the *IP Address* of the device being added. Leave the *TCP Port* at the default setting of **60021**. From the *Host Blade* drop down select the Blade that you want to associate the 3rd Party device with. Click *Ok*.

This completes the process of adding the device to the *System 3rd Party Devices* tab. The added device should show up in the *System* pane under the Blade you added it to. If it does not show up, or if it shows up but has a yellow question mark on it, then there is either a network issue that needs attention, or the device is not connected to the network at all, or one or more steps have been omitted or done incorrectly in the configuration process.

FM-531HD Technical Manual
Table of Contents

Chapter 1 - General Information

Chapter 2 - PC Based Audio Processing Guru® GUI

Chapter 3 - FM-531HD Pro GUI

Appendices

Appendix A

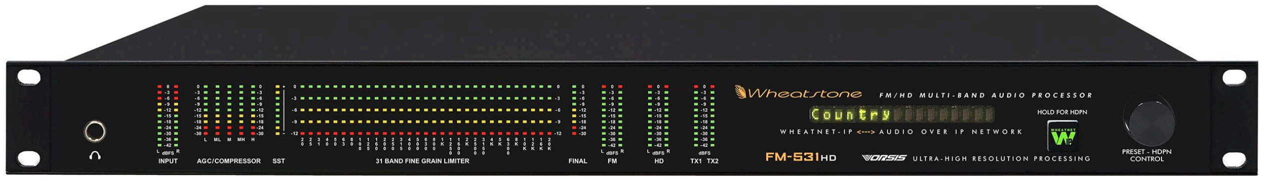
Appendix B

Appendix C

Appendix D

General Information

Chapter Contents



General Information

Introduction

Welcome to the FM-531HD Digital Audio Processor from Wheatstone. The FM-531HD has been carefully designed to deliver a clean and easy to listen to sound on the dial along with very competitive loudness if that is the goal. The FM-531HD expands on technologies previously invented by Wheatstone, incorporates several technologies not available in other processors, and delivers that “sought after” sound to traditional analog FM transmission. Like other Wheatstone processors, the FM-531HD has been carefully designed to not have a fixed sound of its own – the end user is free to use the vast pallet of tools to create any desired on air sound.

The processing algorithms of the FM-531HD have been carefully evaluated though thousands of hours of critical listening and with a very wide range of program material. The FM-531HD can deliver a 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 the FM-531HD is via Windows-based Graphical User Interface (GUI) software and can be accomplished via 100BaseT wired Ethernet. Up to four simultaneous remote control sessions are possible. Eight rear panel General Purpose Inputs (GPI) are provided to allow hardware-level interfacing of the FM-531HD to station systems.

The Wheatstone FM-531HD utilizes an internal base sampling rate of 192 kHz. Far higher sample rates are used where it is beneficial to do so, such as within the clipper. A 24-bit, 144dB internal dynamic range signal path allows audio to be processed with extreme precision.

For complete control over all of the available processing parameters a full-control GUI intended for processing experts can be obtained by registering for it at **http://www.wheatstone-processing.com** Registered users can download a full-control “Pro-Version” of the GUI which provides full access to 300-plus processing adjustments.

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

FM-531HD Feature Overview

The FM-531HD incorporates many features found in the Wheatstone's flagship audio process, the AirAura, but delivers them in an affordable package. The five band AGC incorporates the acclaimed Wheatstone SST (Sweet Spot Technology™) and operates through a proprietary technique called Density Compensation™ . The Density Compensation™ algorithm utilizes real-time analysis of the pre and post-processed audio, and data from the processing controls as the user has adjusted them, to invisibly manage the user set spectral balance and program density. The output of the five band AGC/Compressor is sent to two separate 31 band limiters; one for HD transmission and one for FM transmission.

Unlike processors with fewer limiter bands, human psychoacoustics renders the operation of individual limiters of the 31-band limiters virtually inaudible. At the same time, subtle audio details residing in adjacent bands are revealed, details not able to be heard in other processors. The audio frequency dividing networks within the two 31-band limiters share certain technical similarities with the high-resolution precision filter banks used in perceptual codecs. Like perceptual codecs, the 31-band limiters rely on a well-researched characteristic of human hearing known as "critical band masking" in order to hide the operation of individual limiters from the ear.

The optional HD signal path is equipped with its own 31-band limiter. This is followed by a look ahead limiter for any final peak correction before the audio is presented to the codec.

The FM signal path is equipped with a state of the art distortion cancelling clipper as well as a selectable look-ahead limiter that may be used in combination with the clipper. A digitally perfect stereo generator follows the clipper and provides two multiplex composite stereo outputs, two SCA inputs, and a digital multiplex baseband192 signal. The rear panel BNC input connector for SCA-2 may be switched internally to provide a 19 kHz stereo pilot square wave "Sync" output for RDS applications requiring a separate 19 kHz synchronization signal.

Dozens of factory presets are provided with the FM-531HD making a wide range of on air sounds available to the user. When used in combination with the supplied GUI GURU, these presets offer literally millions of different on-air sound combinations to the user.

The System Menu of the GUI provides access to a pair of flexible preset scheduling utilities. As well as the usual short-term scheduler, the FM-531HD is equipped with a "long-form" scheduler feature that can be programmed for automatic preset changes well into the future. This scheduler is useful for changing presets weeks or months after a preset is prepared, such as for a special event, and without having to edit the daily preset schedule in order to accommodate these "special" presets.

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

Rack Mounting

The FM-531HD is designed to fit into an industry standard 19" equipment rack, and requires one rack unit (1.75 inches) of vertical space. If using only two rack screws always use the bottom two screws to prevent twisting of the front panel and other undue forces from harming the processor chassis.

The FM-531HD 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.

The FM-531HD 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 FM-531HD and devices immediately above and/or below it.

WARNING!

With very few exceptions, the FM-531HD chassis DOES NOT need to be opened in the field.

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

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

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

FM-531HD Installation Tips

Grounding

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

Surge Protection

Always place surge protection circuits as close as possible to the device being protected. AC power line surges should be handled in a way that keeps instantaneous potential differences between the power line hot, neutral, AC grounding conductor, the station ground and the processor chassis as low as possible. Likewise, measures should also be taken to keep the instantaneous potential difference between the audio cable shields and the processor chassis as low as possible (this applies to all audio equipment, not just the FM-531HD), particularly when the equipment is located within the electrically hostile environment of a station's transmitter facility.

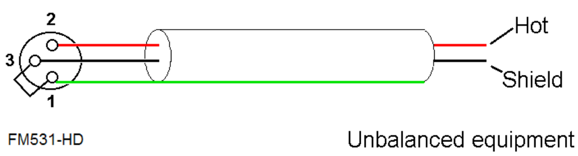
UPS/Power Conditioning

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

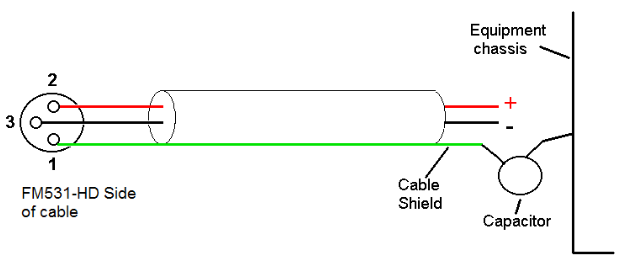
- How does the unit handle AC power that is not exactly 60Hz, such as when the facility is on its backup generator?
- If the unit has onboard surge protection, what kind of surge capability does it have and where are those surges directed to?
- Is the unit equipped with remote monitoring capability?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

Analog Audio Input Connections

Balanced audio *input* sources should be connected to the FM-531HD using standard two-conductor shielded audio cable such as Belden 8451 or 9451.



Unbalanced input audio connections should be made with shielded *two conductor* cable. At the unbalanced source's output connect its "+" output to processor's "Hi" (XLR pin 2) input and connect the shield wire to the "Lo" (XLR pin 3) input. If the cable's shield is used (recommended) connect it at the hardware end only (XLR pin 1) to prevent AC ground loops.



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

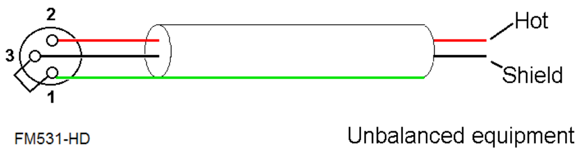
TDK	CS17-F2GA103MYGS
Murata	DE2F3KH103MA3B
Panasonic	ECK-ATS103MF
AVX	65N103MBLCP

Field experience indicates that the listed capacitors can handle the reasonably high RF currents that might exist on audio cable shields at AM and FM transmitter sites.

Analog Audio Output Connections

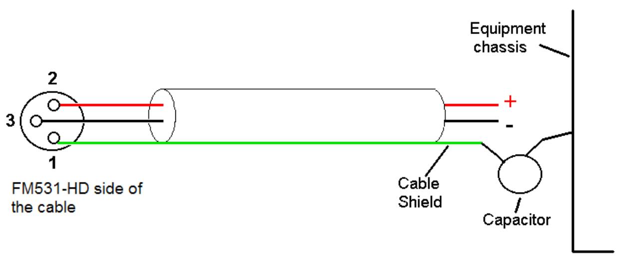
Balanced audio *loads* should be connected to processor's outputs using standard two-conductor shielded audio cable. *Unbalanced* audio loads should be avoided, but if they can't they should be connected using shielded *two conductor* cable such as Belden 8451 or 9451 (as if connecting a balanced source).

The FM-531HD is equipped with active balanced output stages that behave like a transformer. Because of this behavior the wiring may be different than expected.



The unbalanced load's "Hi" lead should be connected to XLR connector Pin 2 ("Hi"). The unbalanced load's <u>shield</u> should be connected to the processor's output XLR Pin 3 ("Lo"). Then, and to enable the processor's balanced output amplifier to operate correctly when driving the unbalanced load it is recommended that the FM-531HD XLR output Pin 1 (ground/ shield) also be connected to Pin 3 ("Lo"), noting that this MUST BE DONE at the processor's output connector (see the diagram above).

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



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

Digital Audio/baseband192 Connections

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

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

Where to Install the FM-531HD

The best location to install the FM-531HD is at the transmitter site. This requires that a discrete Left/Right STL, either analog or digital, be involved in the signal path. The major benefit of a transmitter site installation is that it enables the use of the FM-531HD's built-in lab-grade stereo encoder which allows tighter control of modulation peaks.

A transmitter site location for the processor has the additional benefit of allowing the use of the FM-531HD's highly oversampled composite processor for an additional loudness advantage. The composite clipper algorithm of the FM-531HD is much cleaner and more forgiving than those in other products, and in combination with the 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 the FM-531HD's intelligent oversampled composite clipper. Also, depending on factors including the sample rates being used the exciter's AES digital input can exhibit inferior peak control compared to the exciter's composite input.

When the processor 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:

Analog Left/Right STL

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

Composite Analog STL

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

Analog Phone Lines

Discrete left/right analog "phone line" STL's are not recommended because of the inability of most Telco service providers to meet the flat frequency response and phase matching requirements of the pair of audio circuits. Furthermore, in many countries wideband analog circuits have become unavailable or their cost 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 they may be acceptable.

Digital STL

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

When a digital STL employing codec-based audio compression is being used, the processor should be located at the transmitter site which places it after the codec. The reason for this is that most codecs will sound better when presented with *unprocessed* studio audio instead of highly processed and pre-emphasized audio from the processor's output. Further, the encoding schemes used in such STL's cannot accurately pass the well-defined peak levels created by the FM-531HD, which can create a modulation (loudness) disadvantage.

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

- The stereo generator and composite clipper inside the FM-531HD will not be available. Many digital exciters offer stereo generator and composite clipper functions, but their clippers have historically been quite crude compared to the exceptional clipper in the FM-531HD. Therefore "exciter hosted" composite clippers are *not* the optimum choice when the station's ultimate sound *quality* is important.
- Compressed STL's do not perform well when presented with competitively processed audio, especially when that audio is pre-emphasized. The reason for this is because codecs do their work by examining the audio for opportunities to *remove* content that *shouldn't* be audible to the average human ear. When densely processed audio is presented to a codec there are fewer "opportunities" for it to remove redundant audio information and most importantly, then *mask* that removal from our hearing so that we don't notice it. When handling heavily processed (limited dynamic range) material codec operation is usually much more obvious – even to the point of being objectionable – than when the processing is located *after* the codec where the "masked" artifacts are only occasionally and usually minimally unmasked by "processing gain."

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

TIP: If using the FM-531HD 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 FM-531HD. This will prevent gross distortion and potentially large modulation overshoots.

Where Should Pre-Emphasis Go?

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

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

Arbitron People Meter (PPM)

Field experience has been that Wheatstone audio processors 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, the FM-531HD has no power switch (switches usually become intermittent over time without regular use).

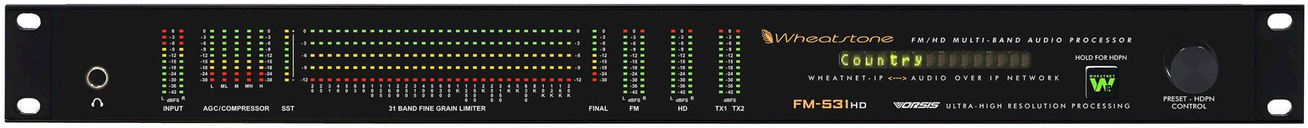
The FM-531HD accepts AC line input voltages between 90 and 260 VAC, 50 or 60Hz. Power consumption is under 100VA.

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. The FM-531HD is, after all, a highly specialized "computer."

The use of a UPS (uninterruptible power supply) is usually recommended and will protect the FM-531HD from short duration power interruptions and glitches which might otherwise signal it to reboot. When the FM-531HD reboots there will be a loss of audio for approximately 15 seconds.

Front Panel Interface



In addition to the comprehensive LED metering display, the front panel also hosts a dot matrix display that shows the current preset on the processor as well as a jog wheel, take button and headphone jack. The jog wheel allows the user to scroll and take presets as well as adjust input and output levels and composite level, and access other menu options like headphone volume control for the front panel jack.

FM-531HD Front Panel Navigation

- PRESETS
  - PRESET LIST
- INPUT
  - SOURCE
    - ANALOG
    - DIGITAL
      - AES
      - WheatNet
  - ANALOG GAIN, dB
  - DIGITAL GAIN, dB
- OUTPUT
  - HEADPHONE SOURCE
    - INPUT
    - ANALOG INPUT
    - DIGITAL INPUT
    - FM PRE_DELAY
  - PRE-EMPH
    - FLAT
    - 50uS
    - 75uS
  - TX-1 OUT
    - TX-1 LEVEL dB
  - TX-2 OUT
    - TX-2 LEVEL dB
  - PILOT
    - PILOT INJ %
  - SCA-1 IN
    - SCA LEVEL dB
  - SCA-2 IN
    - SCA LEVEL dB
  - DIG OUT
    - AES3
      - DIGITAL LEVEL dBFS
    - BASEBAND192
- NETWORK
  - IP ADDRESS
  - SUBNET MASK
  - GATEWAY
  - MAC ADDRESS
  - EDIT
- VERSION
  - SW: xx.xx.xx
  - HW: xx.xx.xx
  - DSP: xx.xx.xx
- ACCESS
  - LOCK
  - UNLOCK
  - EDIT
    - CHANGE?
      - N
      - Y
        - OLD PW:
        - NEW PW:
- EXIT

Basic Setup

Once the processor 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 and adjustments to the input and output and preset takes may be made. To tailor the sound of a preset, the GUI interface running on a Windows PC will be needed. The setup procedure that follows uses the front panel controls.

- Connect the Analog or Digital audio inputs as appropriate for your installation.
- Select the main input source (Analog or Digital). Digital may be via XLR AES or optional WheatNet-IP.
- While in the <INPUT> menu and observing the Input meters, adjust the Input Gain control (Digital or Analog 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 the FM-531HD audio 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) when feeding the processor with unprocessed audio.
- For best overall audio chain performance, the Input of the FM-531HD should be calibrated so that it reaches 0dBFS when the signal chain prior to the processor reaches its 0dBFS, or digital full scale level. This aligns the overload point of the audio chain prior to the processor for maximum headroom and best signal to noise.
- Set the Pre-Emphasis to the value that is appropriate for your region. *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.*
- Now make the audio output connections appropriate for your installation, noting that the Analog and Digital outputs may be used simultaneously.
- The FM-531HD is shipped with the analog output set for MPX. This means that the two BNC female connectors (TX-1 and TX-2) plus the analog XLR connectors (Left or Right) may be used for composite operation. To use these outputs in other modes (L/R or L/R de-emphasized) you must connect to the processor using the supplied GUI on a Windows PC. Adjust the TX outputs as needed for 100% modulation. This should be done with normal programming and not test tones. The RMS level of alignment tones does not crest at 100%. It may take a minute or so with minor adjustments to find a good starting point based on the music/ speech programming being fed to the processor. The default setting is +2.40dB and is calibrated this way to drive the default levels on most exciters.
- If using the AES Digital Output, connect it to the equipment following the FM-531HD and adjust the AES Output level control in the Output Menu as appropriate for your installation.
- If using the digital baseband192 output, modulation levels into the exciter are controlled by the TX-1 and TX-2 level controls. Currently only the TX-1 signal is supported by transmitter manufacturers.
- The FM-531HD Audio Processor factory default preset is called Quick Start. We recommend using this preset as a starting point for aligning and putting the processor on the air. Once the processor is actually on the air, you may scroll thru the list of presets to find one that best represents the format of the station. To do

this, scroll to <PRESET> and press the enter key. To activate a preset, simply press the enter key or the jog wheel once the preset you wish to listen to is displayed. NOTE: *Preset names don't necessarily mean that the preset can or should ONLY be utilized for a program format with a similar name. It is advisable to listen to each preset to see how it sounds on your station, in your market, and with your program material.*

- If you are in a region that requires the use of the BS412 limiter (some European countries), you may control the behavior of the limiter from the front panel. However, you cannot activate the limiter (or deactivate it) from the front panel. You will need to connect with the supplied GUI to do this. The BS412 limiter control can be found under the <OUTPUT> part of the menu.

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 chapter which also cover the operation of the PC-based Guru GUI application.

Rear Panel Connections

The image below shows the rear panel of the FM-531HD and the location of various connectors associated with an installation:

All audio input and output, control, Ethernet, and power supply connections are made via connectors on the processor's rear panel.

The FM-531HD's rear panel connectors from left to right are:

Analog Left Channel In

Analog Right Channel In

Analog Left Channel Out*

Analog Right Channel Out*

SCA-1 Input

SCA-2 Input

TX-1 Out

TX-2 Out

AES Digital Input

AES Digital/baseband192 Out

HD AES Digital Out

GPI Input

Ethernet Control/WheatNet-IP Interface

IEC Standard male AC Power Input

** When FM-531HD's output mode is set to Composite Multiplex, the 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.*

The listed connections are made via various connectors mounted on the FM-531HD's rear panel.

- Seven 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.
- One RJ-45 connector is provided for Ethernet connection and WheatNet IP audio streaming.
- One DB-9 connector provides 8 GPI inputs (for selecting presets in the first eight storage slots). The pinout drawings in the Appendix summarize wiring connections for these features.

Audio Inputs

The FM-531HD accepts 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;

• Optional WheatNet-IP via 100BaseT Ethernet connection to a WheatNet-IP audio network.

Input audio can be applied to all inputs simultaneously.

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 or WheatNet-IP stream, or after 30 seconds of "silence" (level below -48dBFS).

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

Failover from AES3 to WheatNet-IP (or vice versa) is not available.

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

FM Audio Outputs

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.

Analog Out – XLR-M

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

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

AES / baseband192 Out – XLR-M

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

HD Audio Outputs (optional)

The optional HD audio output is available as AES on a male XLR connector on the rear of the FM-531HD.

HD AES Out – XLR-M

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

Headphone Monitoring

An overload protected stereo headphone amplifier drives the front panel ¼" stereo headphone output located on the left side of the processor's 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 Connection

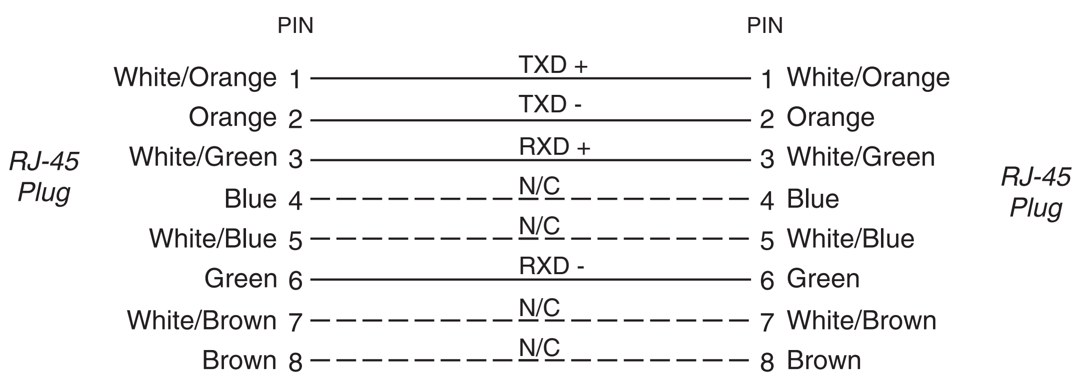
The FM-531HD audio processor has a 100Base-T Ethernet port on the rear panel. The port is Auto-MDIX, allowing it to support straight through and cross-over cables. The Ethernet interface can support up to four simultaneous connections to remote GUIs.

Networked systems are normally connected to the FM-531HD via “straight” (pin to pin) CAT5 cable. For typical CAT5 cable pinouts see below. These connections are for communicating with the configuration computer via a network. If you are connecting directly between the computer and the FM-531HD with no network in between, it is normal to use a crossover cable.

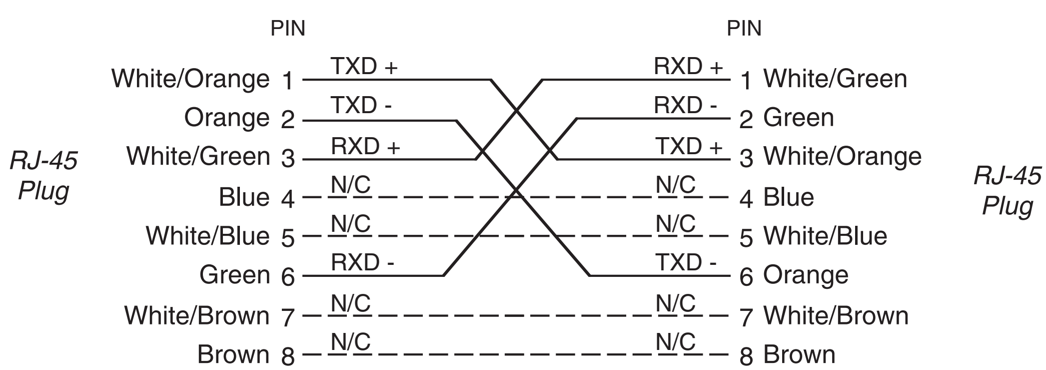
Ethernet – RJ-45

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)

The FM-531HD is equipped with eight General Purpose Input (GPI) control input ports on the rear female DB-9 connector. All GPI connections are optically-isolated to prevent external ground loops and to prevent dangerous voltages from being introduced into the processor. 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.

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 FM-531HD 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 (+)

Processing Presets

The FM-531HD audio processor comes equipped with several dozen factory presets and can hold a total of 80 presets in its onboard memory. Additional customer-created presets may be saved within the processor'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 clicks and pops when presets are changed, preset parameters are slewed between the current parameter values and those of the new preset whenever a new preset is taken. Because of this, when changing presets, especially between presets that are tuned quite differently, it may take several seconds for the new settings to completely settle in. It is important to remember this concept; the FM-531HD's preset switching is designed to make it as unobtrusive as possible and therefore parameter changes are not instantaneous. This factor must be taken into consideration whenever switching between presets in order to compare them!

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

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

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

NOTE:

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

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

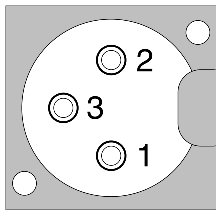
Preset storage is as follows:

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

Analog XLR Connections

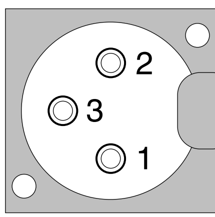
Analog In - XLR-F

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

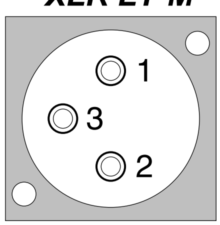
XLR RT-F



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

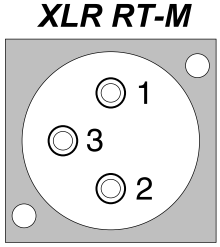
Analog Out - XLR-M

XLR LT-M



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

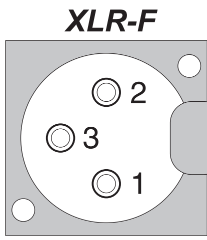
XLR RT-M



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

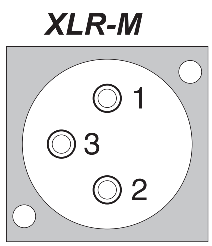
Digital XLR Connections

AES In - XLR-F



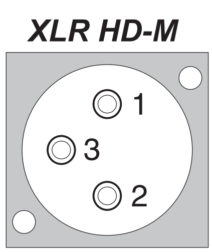
PIN 1 XLR SH - AES IN SH
PIN 2 XLR HI - AES IN HI
PIN 3 XLR LO - AES IN LO

AES Out / baseband192 - XLR-M



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

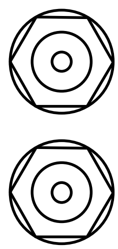
AES HD Out - XLR-M



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

BNC Connections

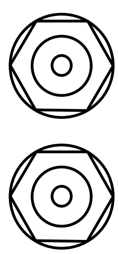
BNC - SCA



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

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

BNC - TX

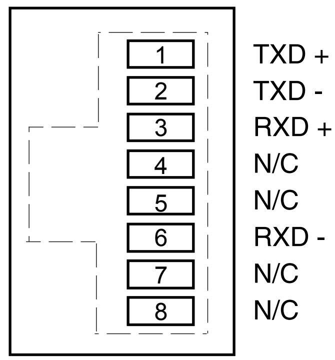


1
PIN 1 BNC 1 HI - TX 1 OUT HI - CENTER PIN
PIN 2 BNC 1 SH - TX 1 OUT SH - SHELL

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

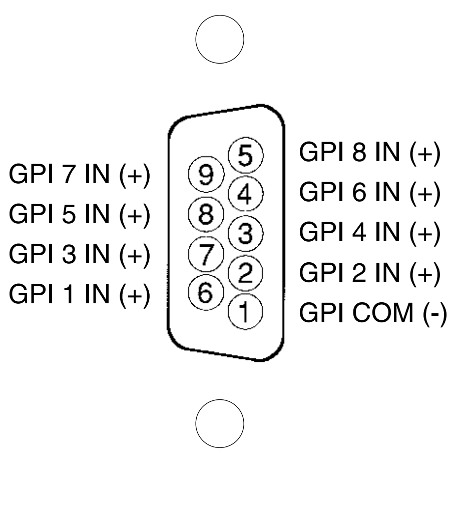
Ethernet - RJ-45

RJ-45 ETH



General Purpose Interface - DB-9

GPI - DB-9



PC Based Audio Processing Guru® GUI

Chapter Contents

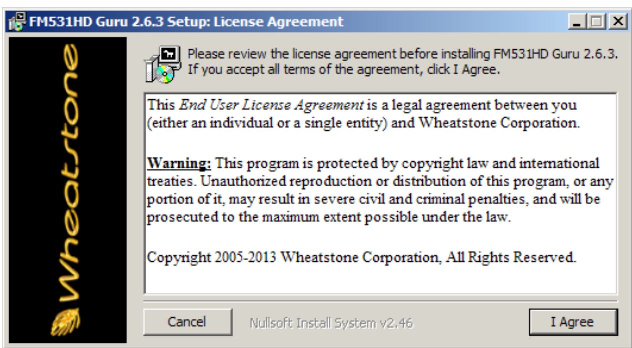
PC Based
Audio Processing Guru® GUI

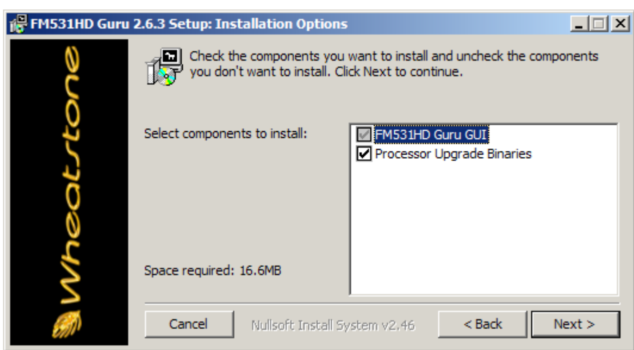
Installation

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 FM-531HD 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 "FM531HD_Guru_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.



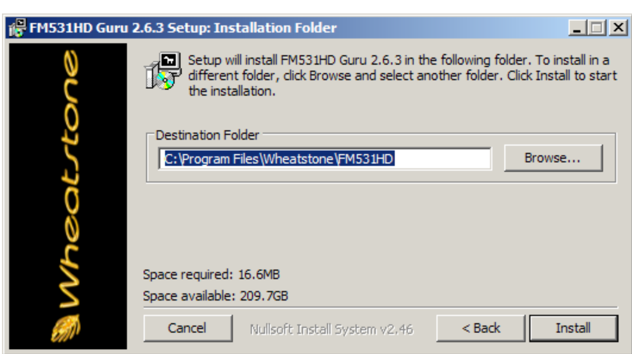


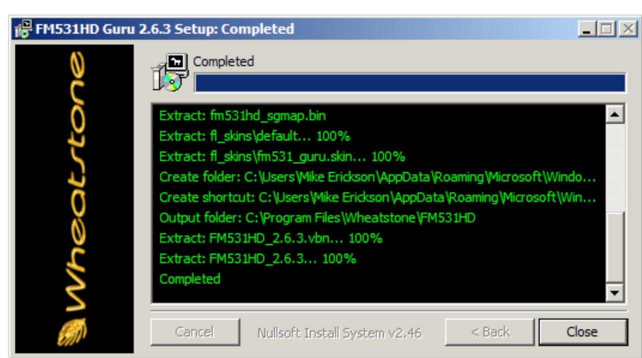
Click on "I Agree" and the installer will present a new prompt asking for what files to install. Leave all check marks selected and press the *Next* button to proceed.

Next the installer will ask where to install the GUI. The default is: *C:\Program Files\Wheatstone\FM531HD*.

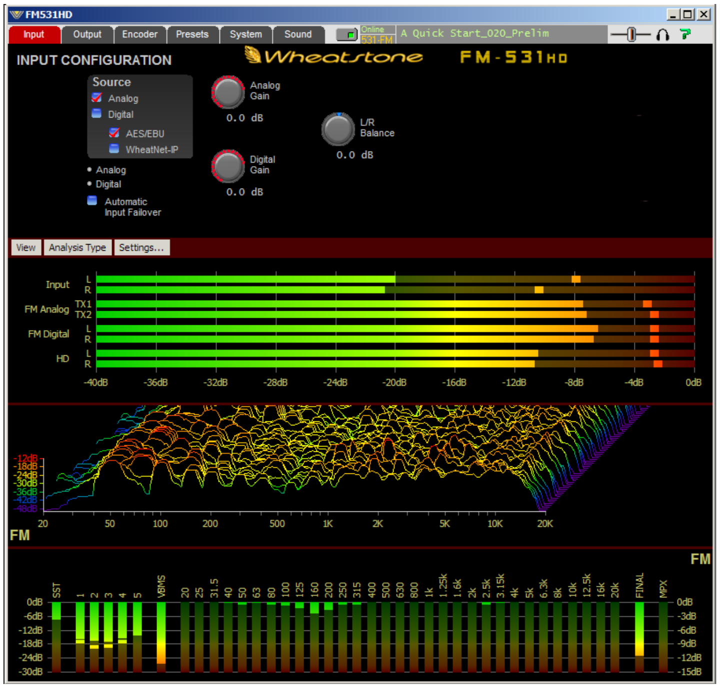
The GUI does not modify the registry and does not require external drivers. Therefore it can be run from any folder, including directly from a USB thumb drive!

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 the FM-531HD!



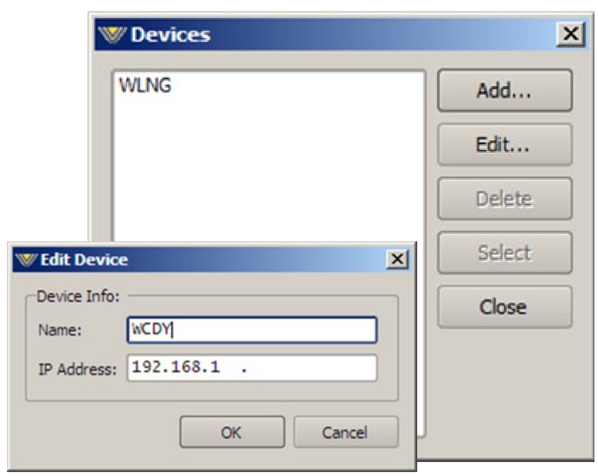
In order to communicate with FM-531HD the following conditions must be met.

1. The PC has an Ethernet interface and a cable is connected between it and FM-531HD (a crossover cable is NOT required).
2. The PC's network interface is enabled and is configured to be on the same IP address <u>subnet</u> as FM-531HD. The default IP addresses for FM-531HD 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. The FM-531HD 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 FM-531HD.

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 FM-531HD 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 FM-531HD 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 FM-531HD's name should appear inside the Device window underneath the Online status.

If the FM-531HD 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 FM-531HD's IP Address Configuration

The FM-531HD's pre-configured factory default Ethernet Configuration is as follows:

Hardware IP Address:	192.168.1.200
Subnet Mask:	255.255.255.0
Gateway:	192.168.1.1

The FM-531HD's network configuration can be viewed or changed using the front panel and scrolling to <NETWORK> and hitting the enter key.

You can then view the four setting configurations:

Hardware IP Address – This is the address of the FM-531HD's DSP farm and is the address that the remote GUI running on a host computer uses to connect to in order to control the FM-531HD. Clear this dialog box by backspacing over the existing address and then enter the new address using the decimal point button to jump from octet to octet within the IP address box.

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 the FM-531HD will use to reach connections *outside* of its own internal network. If you are on a local LAN segment or connected to the FM-531HD by a direct connection, the Gateway address is not usually necessary and can be set to either 0.0.0.0 or 255.255.255.255. Note that this choice appears to be a bit "Windows-dependent" and therefore dependent on the host PC being used.

After determining the IP address that you wish the FM-531HD to use, enter that address in the FM-531HD's Hardware IP address field using the 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 hard to see leading or training blanks in the address. Always use the decimal point button to navigate to the position of the next network address octet.

MAC Address – The MAC Address is a unique identifier for the processor hardware on your network. It is specific to your device and different from any other device on the network. The MAC address of the FM-531HD audio processor cannot be changed.

After viewing the settings above, there will be an option to <Edit..> the settings if changes are needed. If you need to edit the settings:

- Scroll to the <Edit..> option on the dot matrix display
- The first field of numbers will blink on the <IP> field. Using the jog wheel, increase or decrease the numbers until they are at the correct value. Pressing the enter key will move you along to the next set of numbers. Repeat until the IP address is correct.

When the IP address is complete, the next set of addresses (Subnet) will appear for editing. If you need to edit this field, repeat the steps above. If not, simply press the enter key to move thru the fields until you get to the final set of addresses (Gateway). Again, you have the option of editing or just pressing the enter key to bypass each field.

At the end of the menu, the box will confirm any changes by displaying <Okay>.

Reconfiguring The Remote GUI

If the IP addresses of the FM-531HD have been changed from those configured in the remote GUI a few pages ago it will be necessary to go back into the GUI configuration and change the IP address for the FM-531HD device that was originally configured for the remote GUI. 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 FM-531HD 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 FM-531HD 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 the FM-531HD by clicking on the *Online* button at the top of the GUI. When the status indicates Online the GUI is communicating with the FM-531HD.

Connecting Directly Without a LAN

You can work without a LAN by connecting the FM-531HD and the PC Ethernet ports together with a standard Ethernet cable – either a straight through or crossover cable may be used – the FM-531HD's Ethernet interface ports are Auto MDIX.

NOTE: The controlling PC and FM-531HD's network settings must be configured to place them both on the same subnet! 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 the FM-531HD 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 FM-531HD. The laptop will need an IP address configuration that puts it on the same 192.168.1.x network subnet as the FM-531HD.

About DHCP and FM-531HD

The FM-531HD does not utilize nor support Dynamic Host Configuration Protocol (DHCP, or automatically assigned network addressing) – and needs to be assigned "STATIC" network-unique IP addresses.

FM-531HD's Ethernet Ports and Network Protocols

The FM-531HD uses both TCP and UDP protocols to communicate with the remote GUI. The TCP protocol is used between the GUI to the FM-531HD for control functions. TCP's high reliability (due to handshaking) ensures that control changes sent to the The FM-531HD by the GUI will be received *without* errors.

Metering data is not at all critical, and only the differences between past and next meter values are sent. Therefore a "no-handshaking" protocol, UDP, is used to send metering and other real time data back to the GUI. Dropped or corrupted packets will 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 that is required in order to run the GUI's remote meters and analysis displays.

The FM-531HD's TCP protocol utilizes a fixed port, port **55900** for control. The FM-531HD uses UDP for metering and tries to open port **60000** first. If port **60000** is not busy, it uses it. If UDP port **60000** happens to be busy, UDP port **60001** is tried next. This process continues until either an idle UDP port is found or the search stops at UDP port **60010**.

NOTE: The above reference ports must be open for any through-the-firewall connectivity. This also includes the Windows firewall on the host PC that is running the GUI software – it must have these ports open too.

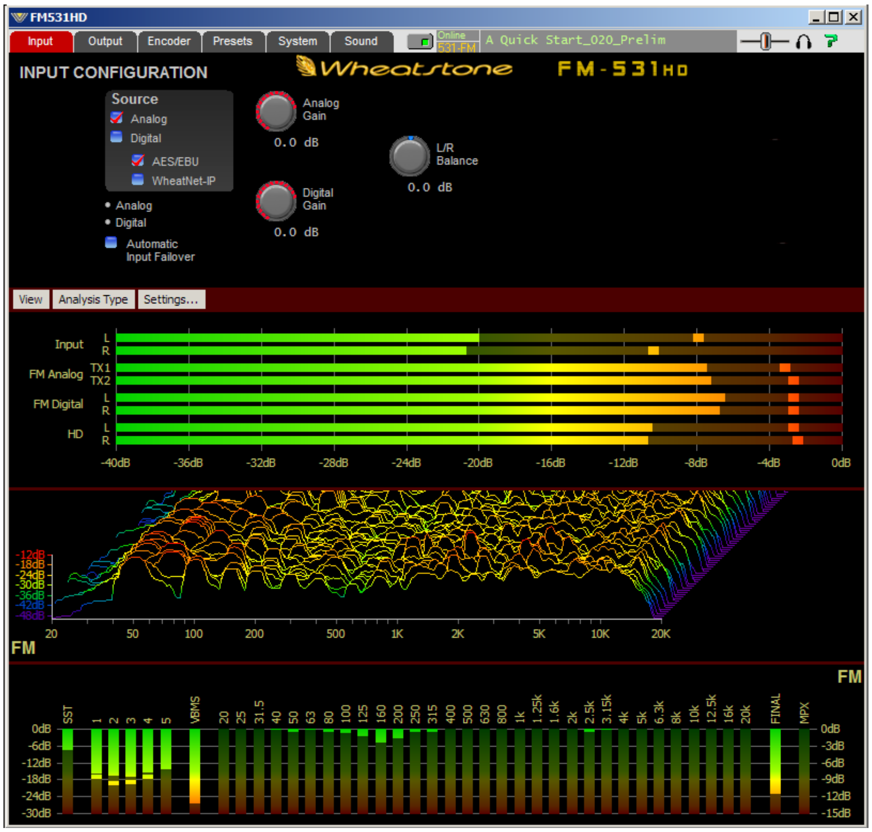
If the GUI Is "Connected" But the Meters and Analysis Displays Don't Work

Occasionally a network or internet service provider will block UDP traffic. When that happens the GUI will connect to the FM-531HD and the FM-531HD can be controlled, but no metering or analysis graphics will be displayed. Fortunately we have provided an easy workaround:

Navigate to the GUI's System screen and check the box labeled *Metering data over TCP/IP* and the issue should be solved. Note that choosing the TCP protocol for metering will increase the amount of network traffic because now every metering and analysis packet requires a handshake to ensure its integrity. The poorer the network, the more bandwidth the GUI will try to use (a poor network implies the need for more handshaking to ensure good data).

Using Guru GUI

The GUI may be repositioned on the host computer's screen by left-clicking and holding its title bar and dragging the GUI to the desired 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. Guru GUI may also be resized on the desktop as desired by dragging its borders.



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

- It will begin with the Control Area at the top;
- Followed by the Input/output Level displays;
- Followed by the Gain Reduction Displays;
- Ending with the Dynamic Analysis displays.

Input Menu

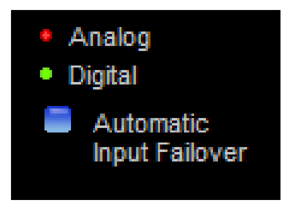
This is the first tab on the top far left of the GUI, noting that the screen that is currently being displayed is indicated by a red background on the selected tab.

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.

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 and the analog input exceeds -48dBFS. The *Digital* indicator lights if there is a valid AES signal present at the digital input above -48dBFS. In this example, Digital is the active input (green indicator).

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 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 range of +12dB to -36dB.

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 range of +12dB to -36dB.

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*.

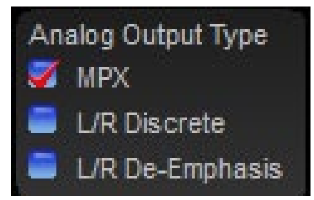
FM-531HD is also equipped with our Wheatstone® baseband192 digital composite interface which provides a direct, pure digital signal path between the components of the stereo multiplex spectrum and the associated transmitter's digital FM modulator. With this interface the usual A to D and D to A and/or sample rate conversions are absent resulting in virtually perfect stereo baseband and modulation performance.

The FM Digital output available as AES3 digital signal or baseband192 digital multiplex signal.

The FM AES3 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. To select the type of pre-emphasis, please see the ENCODER section below. Besides de-emphasizing the output, the signal can be routed pre or post 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.

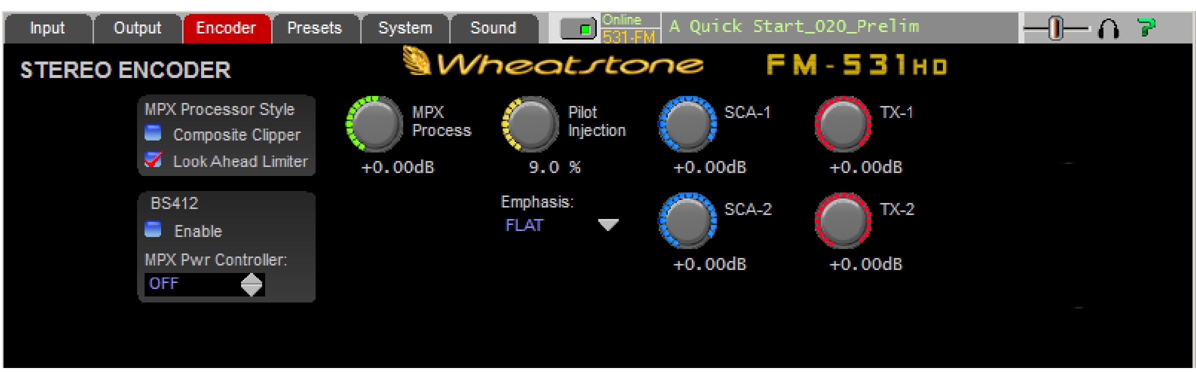
NOTE: The Analog Output Type box needs special mention!



Unlike other audio processors FM-531HD'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 <u>composite stereo</u>. 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



The FM-531HD 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 >60dB between 20Hz and 15kHz. Clicking on the *Encoder* tab opens the *Stereo Encoder* screen hosting controls for this section.

MPX Processor Style

The FM-531HD 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 look ahead 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 look ahead 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* 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.

FM-531HD 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* check box is not checked. Therefore if the MPX Power Controller setting is set to any value except off, when the *Enable* check box is checked MPX Power will immediately fall to the value set by the *MPX Power Controller* control.

Enable – When the *Enable* check box is checked the BS412 algorithm is controlling the multiplex power. Likewise, when the check box 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* check box 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!

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 the FM-531HD is being used. The pre-emphasis may also be switched off (FLAT) for testing or other purposes.

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. The TX-1 and TX-2 controls also adjust the modulation levels of the Baseband192 output. Currently only the TX-1 signal is supported by transmitter manufacturers.

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 the FM-531HD's comprehensive Preset Library Management utility.

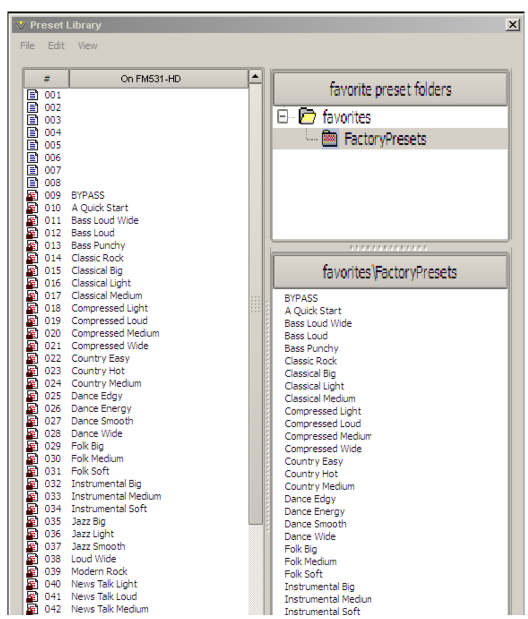
Selecting a Preset

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 the FM-531HD's next available open preset slot.



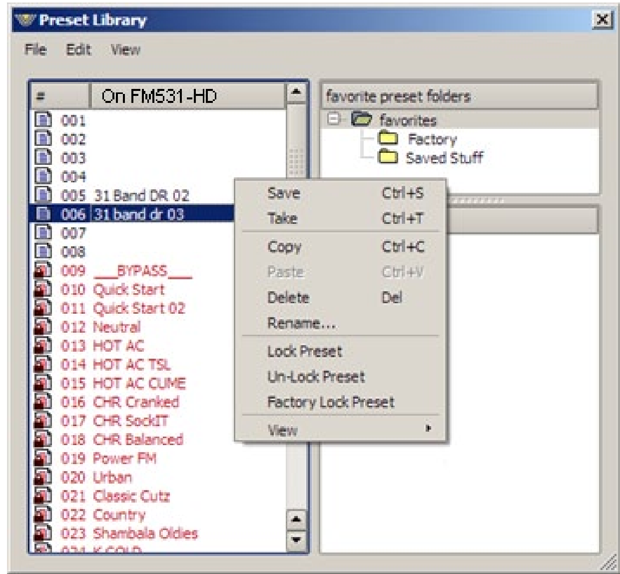
Preset Library Functions

Clicking on the *Library* button opens up the preset library management feature. When this window is open both the presets on the FM-531HD 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 the FM-531HD 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 safekeeping.

Locking Presets



Presets stored on the FM-531HD 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:

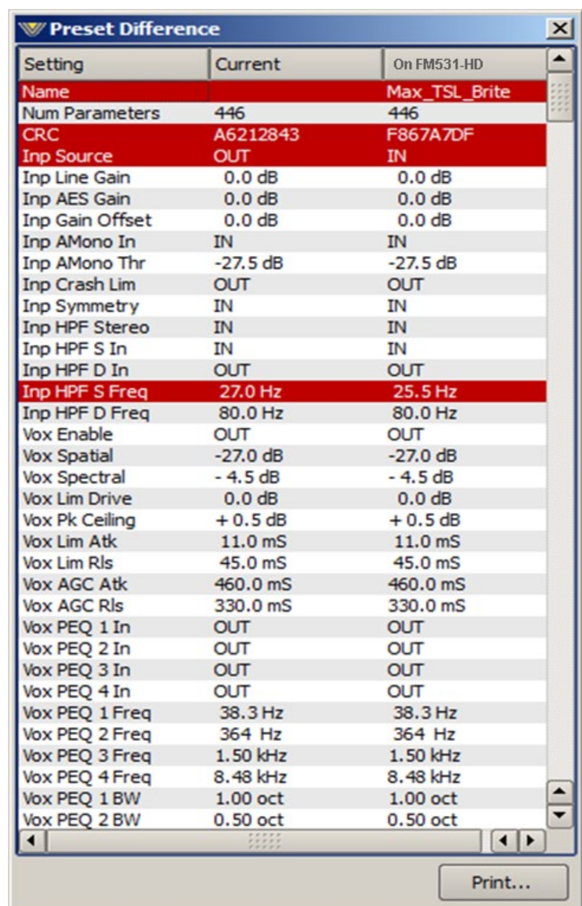
- The first method is by highlighting a preset (single left click) and then right clicking it to open a dialog box. Among the options are *Lock Preset* and *Unlock Preset*.
- The second method is by highlighting a preset as above, and then clicking the *Edit* option at the top of the *Preset Library* 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 FM-531HD with a special feature called Preset Difference.

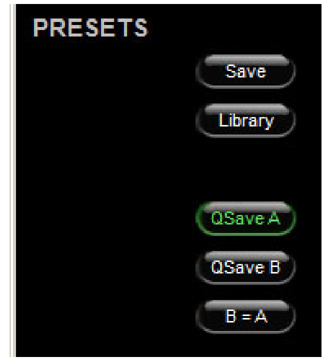
When the *Library* tab is opened, highlight any preset you'd like to compare with the current preset running on the processor. Then right click and select *View* and *Diff* from the sub-menu. You will see the window on the right open.

Any parameter that is listed with a RED background indicates a parameter that is different 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.



Setting	Current	On FM531-HD
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

Quick Save (QSave)



The FM-531HD 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 FM-531HD 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 FM-531HD 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 FM-531HD 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

Wheatstone 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 Wheatstone preset is saved all currently active parameters are stored with that preset, including system settings. 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 FM-531HD 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.

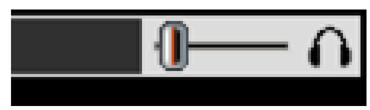
Headphone Source

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 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: The FM-531HD'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 FM-531HD 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 *Digital*.

Temp

This indicator is green when the GUI is online to the FM-531HD and its internal temperature is below 50°C/122°F. The numbers displayed represent the current temperature of the hardware.

Analog

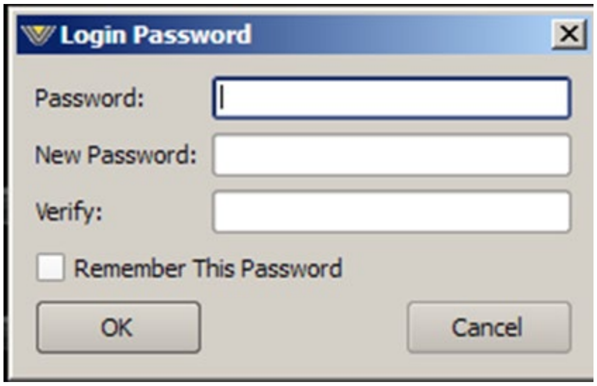
This indicator is green whenever there is analog audio present on *both* left *and* right channels and it is above -24dBu.

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 FM-531HD processor over the Ethernet connection. When a remote GUI session is initiated, the FM-531HD 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 the FM-531HD 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.

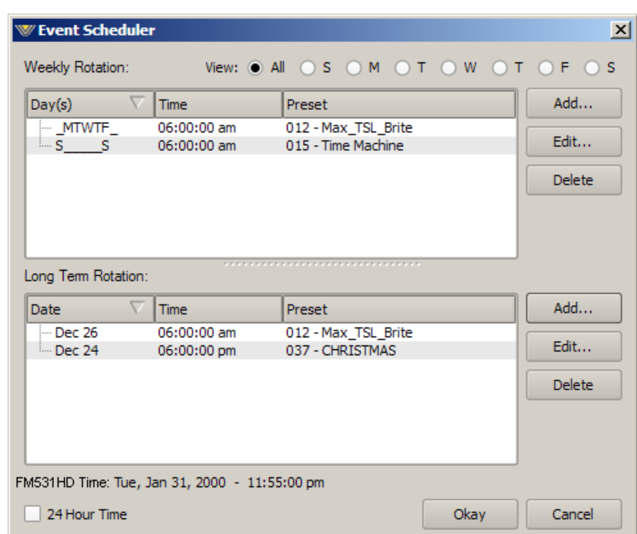
If the *Remember This Password* check box in the dialog is checked, the GUI stores the password on the local PC. If you do not check the check box you will be asked for the password *each time you connect remotely* (since the first round of verification will fail due to no locally stored password).

Occasionally you may see the word "pending" temporarily appear in the connection status box of the remote GUI while the FM-531HD and Remote GUI are handshaking on the password string being sent.

Events

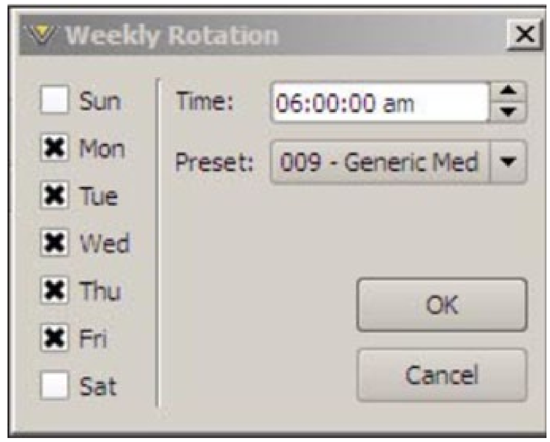
The FM-531HD contains a comprehensive scheduling utility that allows presets to be automatically changed on desired dates and times. There is also a *Long Term Rotation* scheduler that allows preset changes to be scheduled for any time in the future. To create, edit, or view entries in the *Event Scheduler* 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.

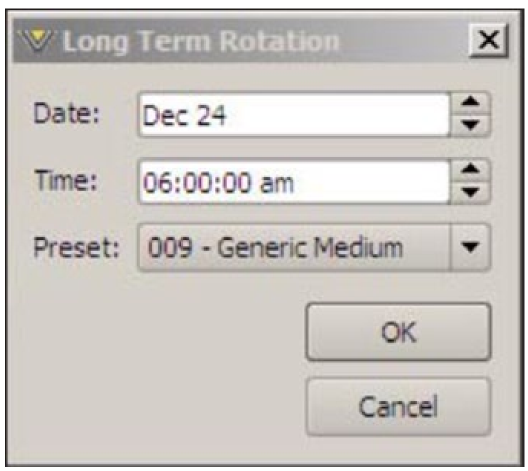
2. On the left column, select the days that the preset should be selected. Any combination of check boxes may be chosen.
3. Next select the time that the preset should change.
4. Next choose the preset that should be selected at the time programmed in the previous step. When finished editing, click *OK*.

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.

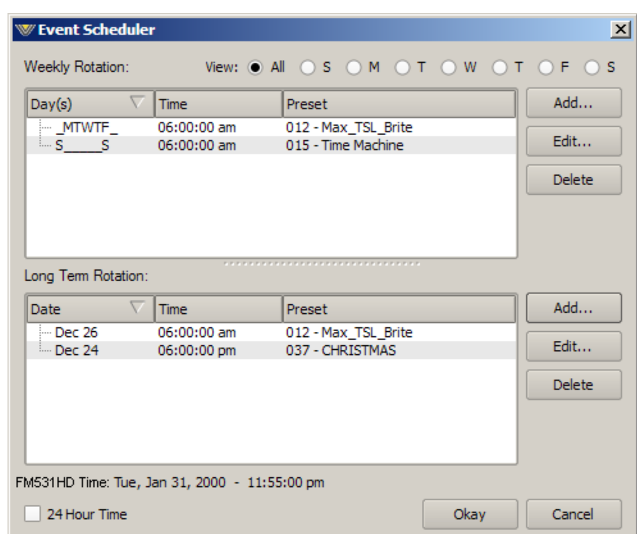
Creating a Long Term Rotation Schedule

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

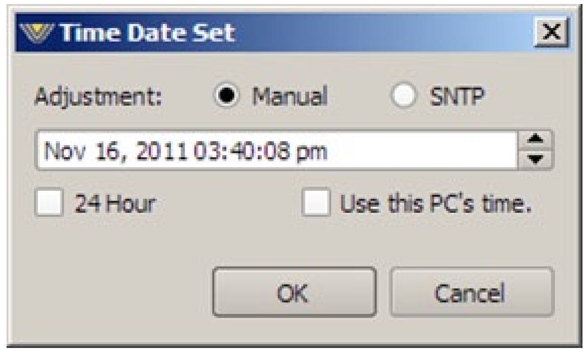
Deleting a scheduled preset change is as simple as adding one. Open the *Event Scheduler* window by navigating to the System menu and clicking *Events*. The *Event Scheduler* window will open.



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 the FM-531HD's onboard computer to be set or changed. FM-531HD'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)

The FM-531HD supports automatic and unattended system time updates via SNTP. Using SNTP can keep the FM-531HD'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 FM-531HD's NTP function is enabled the network that the FM-531HD 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 the FM-531HD 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 <u>shared</u> between the FM and HD processing chains.

FM Sound Adjust

AGC Depth

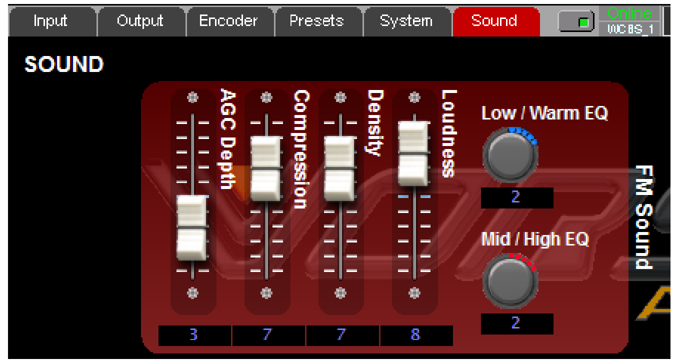
The *AGC Depth* control adjusts how deep the AGC’s *long term* gain reduction will be with typical program material and with appropriate input levels. Higher settings cause deeper long-term gain reduction and the additional “reserve” gain caused by raising the setting of this control will tend to hold 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.

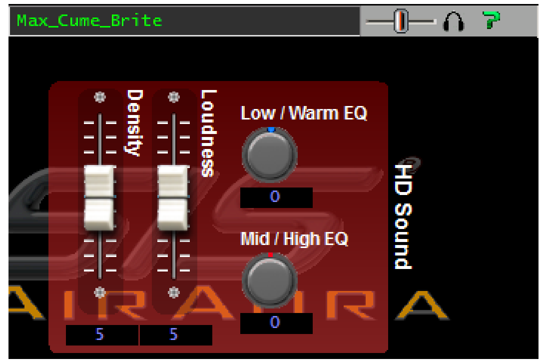
Loudness (finally!)

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 the FM-531HD'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 look ahead 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

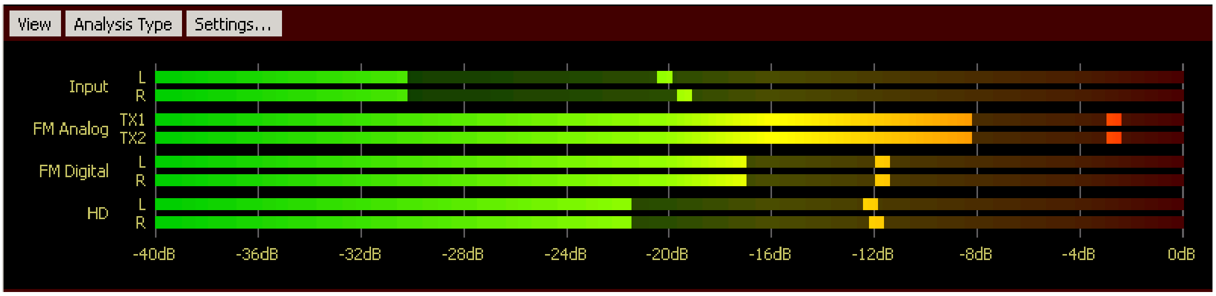
Once the previous controls have been adjusted for the desired sound texture, the *Loudness* control can be used to set the desired HD loudness. The Loudness slider intelligently communicates with the HD processing's 31-band limiter and its look ahead limiter.

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

The FM-531HD 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 the FM-531HD 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.

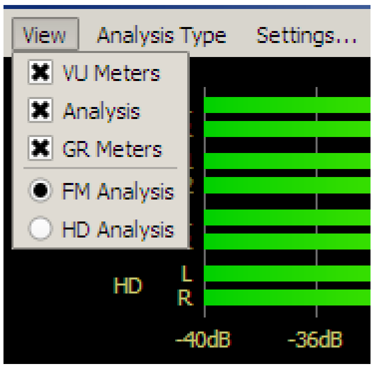
So, what are normal input levels? Because we are working in the digital domain great care must be taken to ensure that audio levels aren't run so high as to cause them to run out of headroom. Good engineering practice dictates that levels arriving from the studio or STL should be peaking between -20dB 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 FM-531HD'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.

The FM-531HD'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 we must do for input signals.

Audio Processing Gain Reduction Displays

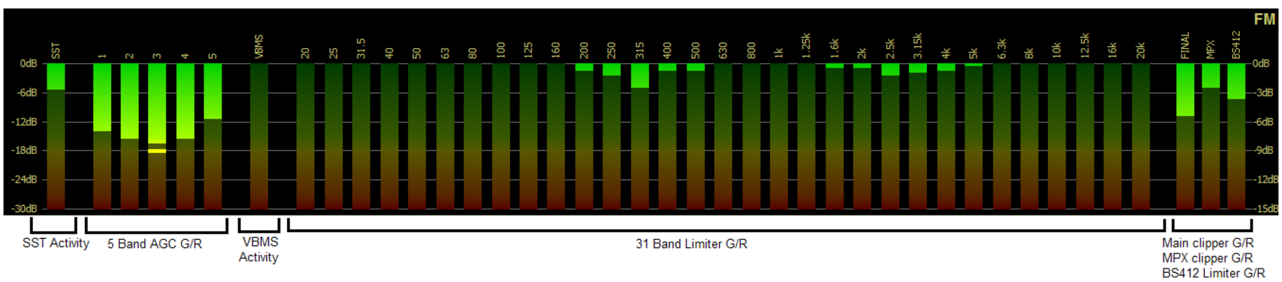
At the bottom of the GUI display are the vertical bars representing the amount of activity occurring in each of the FM-531HD's processing sections.



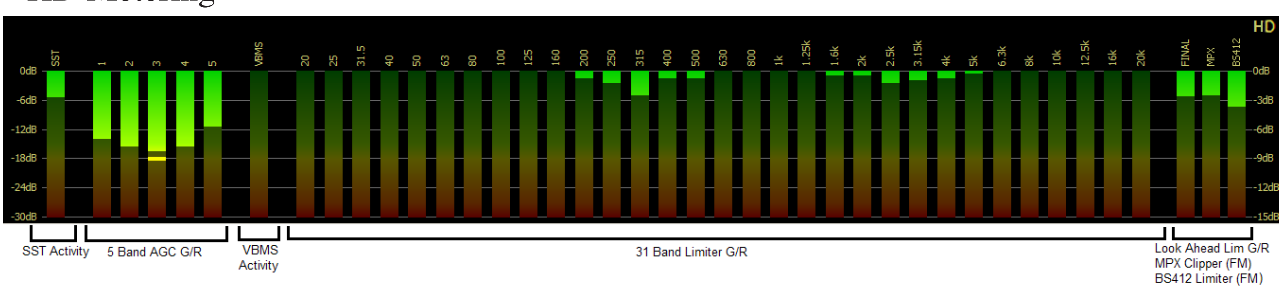
The *View* button on the top-left of the metering section allows you to scale the size of the GUI. This is done by "turning off" some or all of the metering and analysis displays. The check boxes allow you to pick the metering and analysis you want to view.

Also, there is an option for which processing path you want the analysis to display. The options are for the FM path or the HD path.

FM Metering



HD Metering



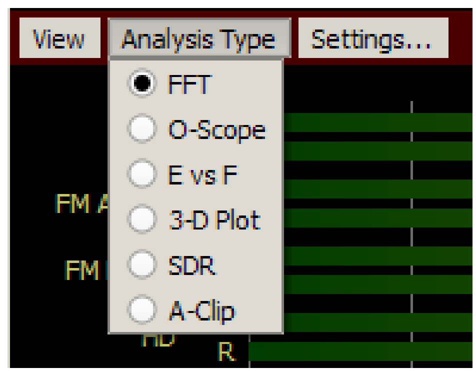
The meters are arranged in seven logical groups as follows:
- SST Activity meter;
- Five-band AGC gain reduction;
- VBMS Activity meter;
- 31-band Limiter gain reduction activity;
- Final Limiter gain reduction or Clipper activity;
- MPX Limiter gain reduction or Clipper activity;
- BS-412 MPX Power 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."

Analysis Menu



The FM-531HD is equipped with extensive processing analysis features that show what the processing is doing and by how much and when. The Guru GUI has a drop down menu where the analysis to be viewed may be selected. There are six options in this drop down menu:

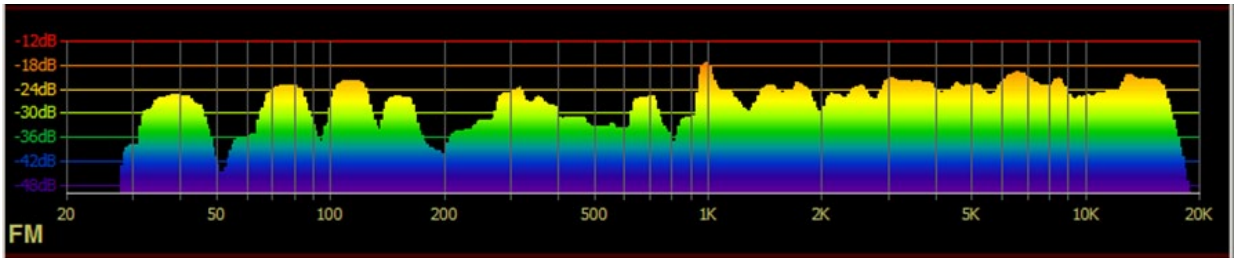
- **FFT** – Displays the spectral output of the FM or HD processing paths after processing. The FM display includes any pre-emphasis being used in order to present an accurate view of the signal actually being sent to the transmitter by the FM-531HD.
- **O-Scope** – Displays the audio waveforms at the output of the FM or HD processing chains.
- **E vs F** – Displays the measured *RMS Loudness* within each of the 31 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 Wheatstone 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 FM-531HD's final FM clipper activity display (another Wheatstone 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

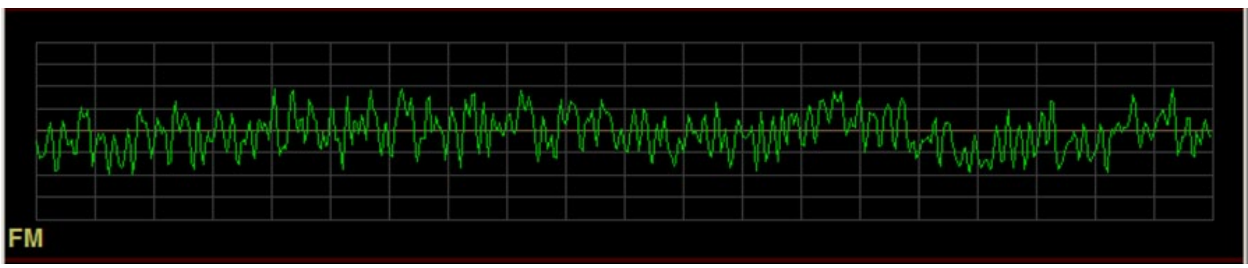
This display shows a Fast Fourier Transform of the audio spectrum after processing has been applied by the final FM clipper or HD limiter. The FFT display's high resolution is useful for locating where in the audio spectrum certain signals are heard.

With complex program material the FFT gives good visual feedback on the levels of audio residing in different parts of the audio spectrum.



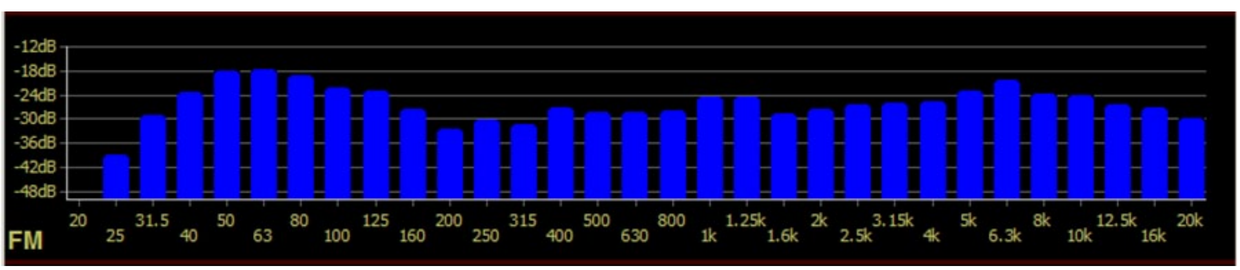
O-Scope

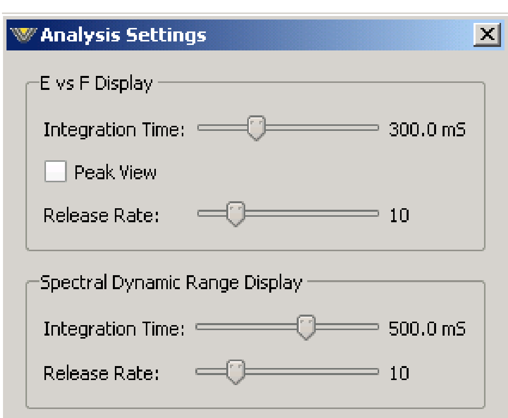
This is an oscilloscope-type time-domain display of the audio waveforms exiting the output of the FM final peak control sections. The FM O-scope display will reveal what the pre-emphasized and clipped audio looks like.



E vs. F

This display represents the RMS Loudness generated in each band of the FM or HD path's 31 limiter bands.





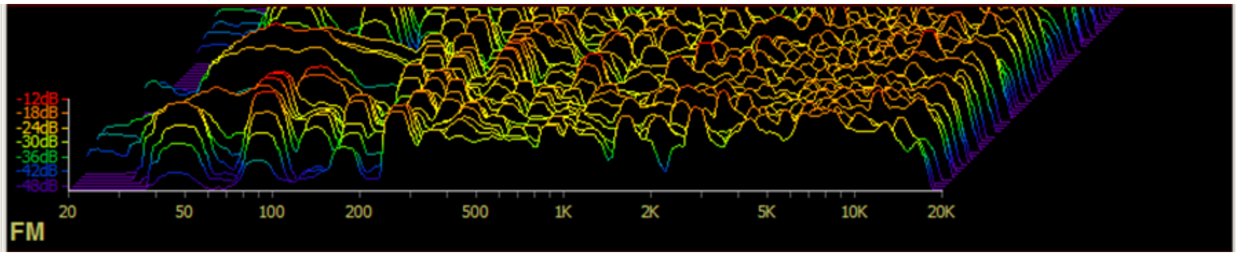
The E vs. F display has a submenu (shown left) that allows customization of the RMS Integration Time, the display Release Rate, and whether or not peak levels are to be displayed. The factory default Integration Time is 300mS – the same as a standard VU meter – which is also a fair estimator for how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

The submenu also allows you to adjust timings for the SDR display (below).

3-D Plot

A 3-D plot of the audio shows the last ten seconds of audio and what the audio levels were at each frequency in the audio spectrum during that time.

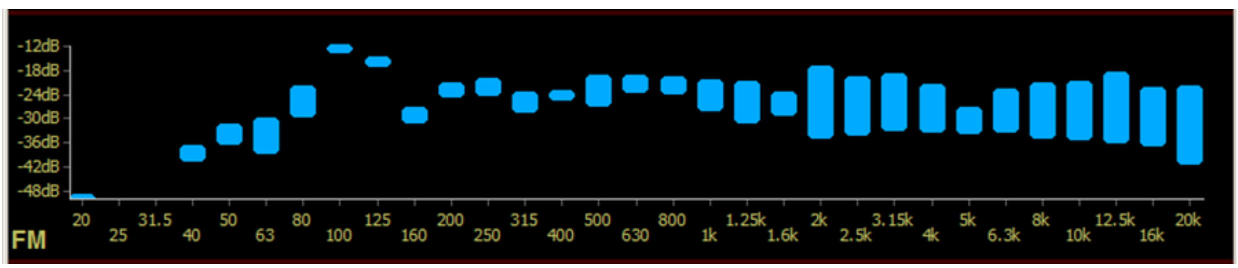
The 3-D plot can be very helpful when looking at how the audio spectrum has been behaving over time. The higher the signal is in the plot, the larger the signal amplitude at the frequency of the peak.



SDR™

During development of the FM-531HD product we invented the *Spectral Dynamic Range™ Meter* in order to assist us in evaluating what was happening in our new FM531-HD clipper as it managed the audio spectrum.

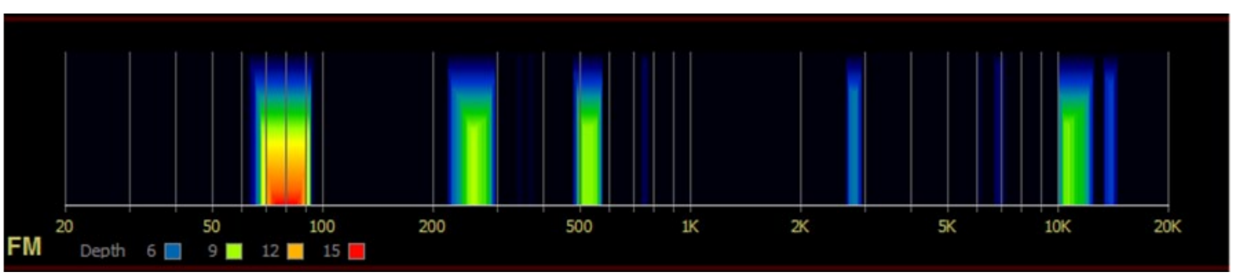
We made the SDR meter very easy to interpret – the longer the blue bars, the more dynamic range there is between the peak amplitude and its RMS (loudness) level inside each of the 31 limiter bands. Conversely, the shorter the bar's length is, the denser the audio is in that limiter band.



A-Clip

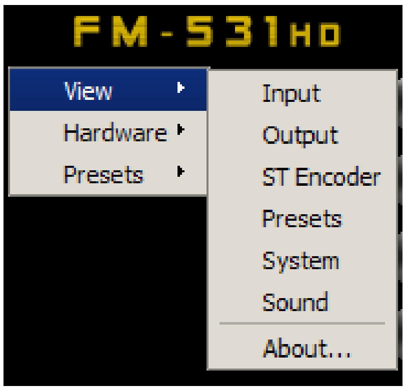
During development of our new FM clipper we realized that the only way to visually monitor what the FM531-HD clipper was doing was to invent a way to watch it work. The A-Clip display was born out of that idea.

This display shows in detail, *where* in the spectrum the FM531-HD's distortion management is being accomplished, and by *how much*. Like a weather radar image, the "redder" the display, the more active the work being done in that part of the audio spectrum.



Accessing Menu Options

Right clicking anywhere on the FM-531HD 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.

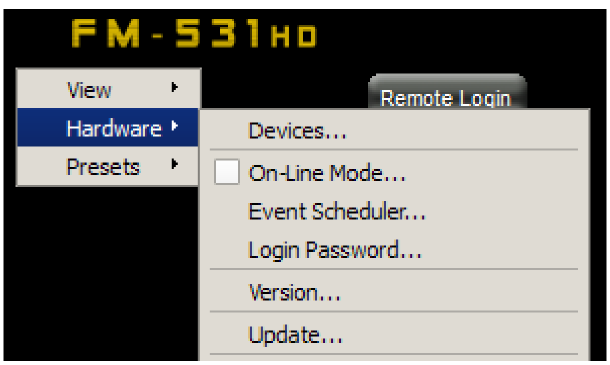
Presets: Makes the Preset management screen visible.

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 FM-531HD 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 FM-531HD.

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.

FM-531HD Pro GUI

Chapter Contents

FM-531HD Pro GUI

The *FM-531HD Professional Graphical User Interface* (or Pro GUI) allows you to adjust the processor and see, in real time, how the processor is reacting to the audio passing thru it. Just like all of the Wheatstone Pro GUI's, the FM-531HD Pro GUI has been carefully designed so that no adjustment is more than two mouse clicks away.

The FM-531HD Pro GUI software allows you to control an unlimited number of FM-531HD's via a standard Ethernet local area network (LAN). Utilizing TCP/IP and UDP protocols, the GUI can be used to control any FM-531HD processor from anywhere with an Internet connection, including those behind firewalls and through VPN tunnels.

Getting Started

The FM-531HD Pro GUI is a Windows™ based program and is designed to be intuitive and straightforward in use. It is supplied as a self-installing program executable that may be installed from any directory on the computer. The resulting files will be installed under "\Program Files\Wheatstone\FM531HD." After installation, select the shortcut in the Windows Start menu to start the Pro GUI program.

To configure the FM-531HD network connection, a basic familiarity with Windows navigation techniques is assumed. If terms such as "left-click," "right-click," "double-click," and "drag-and-drop" seem alien, operating the FM-531HD's remote GUI could be a bit problematic.

Configuring The Pro GUI

The Pro GUI must first be configured so that it can find the FM-531HD on the network. Install and then start the Pro GUI. In the GUI, click *Devices>Add...* to bring up the *Edit Device* form (see page 2-4).

Fill in the pet name you'll be giving to the unit. In the IP Address field below the name fill in the IP address of the unit and then click OK.

Highlight the new the FM-531HD device in the Devices list and click on the *Select* button. Place the GUI online by clicking on the button to the left of the Status message – its indicator should turn green. Once the GUI has been made aware of the FM-531HD's existence in this manner, it will always appear in the list of the FM-531HD Devices and be instantly accessible from anywhere at anytime.

When connected to the FM-531HD the status bar will indicate *Online* and the "Devices" text will show the name of the FM-531HD that it is currently communicating with (in this example, the FM-531HD has been named *WERO*). The status bar will also display the name of the preset that the FM-531HD you're connected to is currently running.

When the Preset name is green it represents a preset (factory or user) that has been recalled and is running with no changes made to it. If the Preset name is scrolling in Red it means that the preset has had changes made or is a work in progress not yet saved to the processor.

Connecting Directly Without a LAN

You can work without a LAN by connecting the FM-531HD and the PC Ethernet ports together with a standard Ethernet cable – either a straight through or crossover cable may be used because the FM-531HD has an auto-sensing port.

NOTE: The controlling PC and the FM-531HD's network settings must be configured to place them both on the same subnet!

PC System Requirements

Constraints for running the GUI are not extraordinary – the PC should be preferably at least 1GHz in speed, running Windows 2000/XP or better. Screen resolution should be at least 1024 x 768 pixels in order to get the full benefit of the graphics used. You need about 15MB of space on the hard drive for the GUI software and presets. The PC's LAN interface (NIC) should be 100baseT (100MHz) capable for best performance.

The Really Remote Connection!

One of our customers routinely adjusts his station's Wheatstone processor from the comfort of his company car. In order to accomplish this he connects it to his station's business LAN's router, maps the unit's IP address over to a "public" IP address via NAT (Network Address Translation), and creates a hole in the firewall for this IP address. This gives the Wheatstone "Internet" connectivity.

From the laptop in his car he connects to the Internet using his Verizon wireless card, fires up the GUI, connects to the IP address that is running in the box (NAT takes care of the IP address differences), and Voila! He can adjust his Wheatstone processors from anywhere. We will leave the myriad other ways in which the FM-531HD can be remotely controlled "wirelessly" up to your imagination!

VPN

Another way to connect remotely to the FM-531HD is via VPN. A VPN, or Virtual Private Network, is a private data tunnel between computer devices. It behaves just as if the local and remote computers were sitting beside each other – devices connected via VPN act like they are on one big LAN regardless of where they are physically located. VPN tunnels can be configured both with and without data encryption. Setting up a VPN is not difficult, but certainly beyond the scope of this manual.

The FM-531HD and Internet Security Concerns

The FM-531HD 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 FM-531HD and use it as a pathway to the rest of the network to which it is connected.

About DHCP and The FM-531HD

The FM-531HD 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 local friendly network administrator to determine what static IP address is open and safe to use. 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 should suffice for all but the most complex situations.

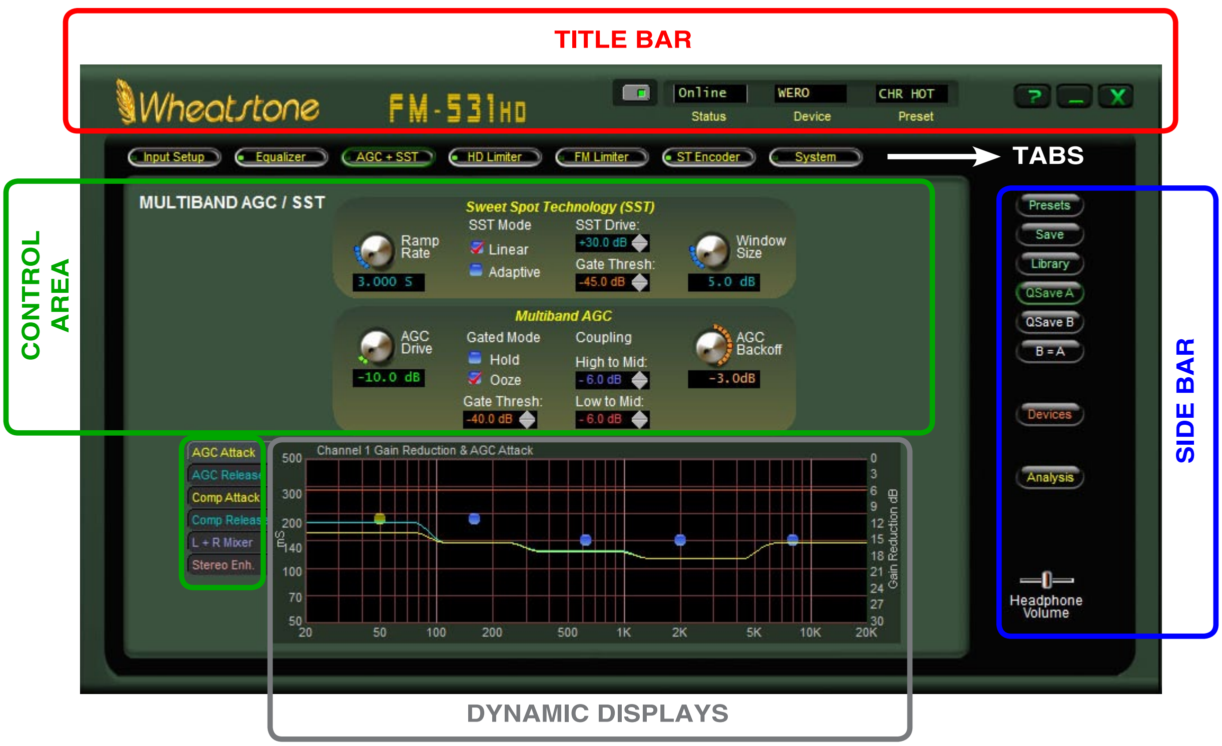
The FM-531HD Ports and Network Protocols

The FM-531HD uses both TCP/IP and UDP protocols to communicate with the remote GUI. TCP/IP is used from the GUI to the FM-531HD because its high reliability ensures that control changes sent to the FM-531HD will be received. UDP is used from the FM-531HD to send and update metering and other real time data. TCP/IP uses port 55900, while UDP uses port 60001, and, if 60001 is busy, tries the next highest port until an idle one is found. Both TCP/IP and UDP ports must be open for any firewall based connectivity.

Using The FM-531HD Pro GUI

The GUI may be positioned on the host computer's screen by left-clicking and holding the top edge of the Title Bar and dragging the GUI to the desired position. In true 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.

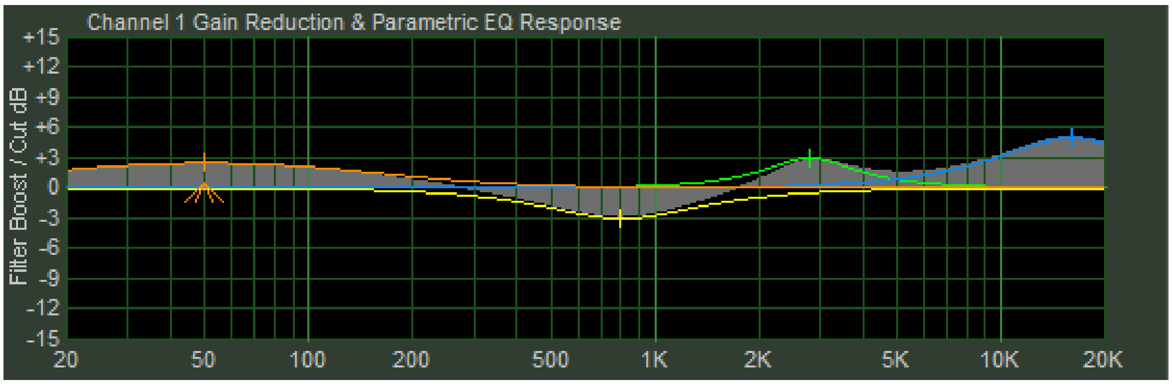
Descriptions of the features and functions of the GUI will follow different discrete regions of the screen; it will begin with the most eye-catching region – the Dynamic Displays – and then we'll explain the Control Area above it, followed by the Side Bar buttons, and lastly the Title Bar.



Dynamic Displays Region

Frequency-Domain Graph

Visually the most important element on the main screen, and in fact dominating the lower half of it, is a large graphical display depicting the standard audio frequency range of 20Hz – 20kHz arranged in logarithmic fashion with gradations at selected frequencies. The vertical axis scale changes according to its context.



FM-531HD Graphic EQ Display

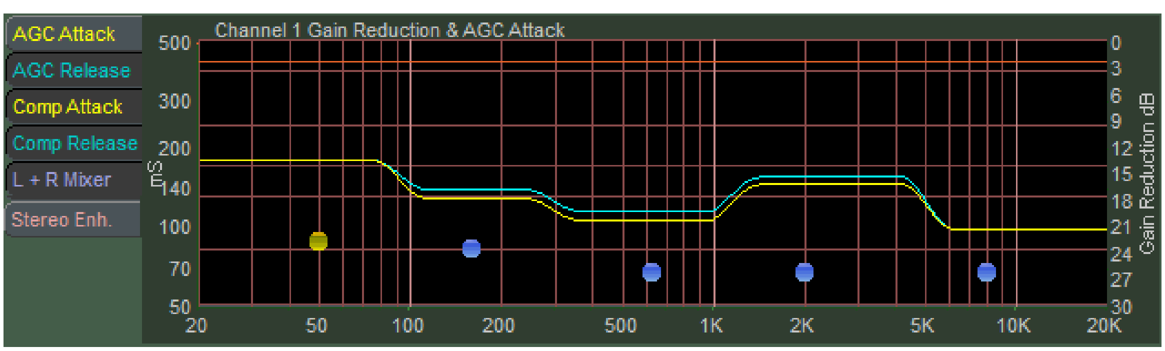
The FM-531HD Graphical Interface Operation

Select either AGC+SST (pictured below), HD Limiter, or FM Limiter from the buttons at the top of the Control Area. 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 the many number of controls the user has access to (well over 350).

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 of the blue dots on the screen represents the center frequency of a 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 each band can be adjusted.



FM-531HD 5 Band AGC/Compressor Adjustment Display

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

First, a short tutorial....

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.

If you double click a blue dot, it turns red to indicate that it has been selected, after which it may be moved individually to anywhere along the vertical scale.

If you now double click another blue dot somewhere else on the line, it too will turn red....

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 red.

At this point all of the dots can 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 red 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 **red** 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 **red**, regardless of any previous highlighting.

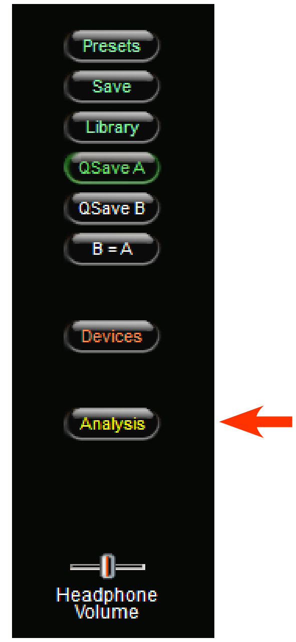
Flatten – Flattens into a horizontal orientation any dots not set to red.

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

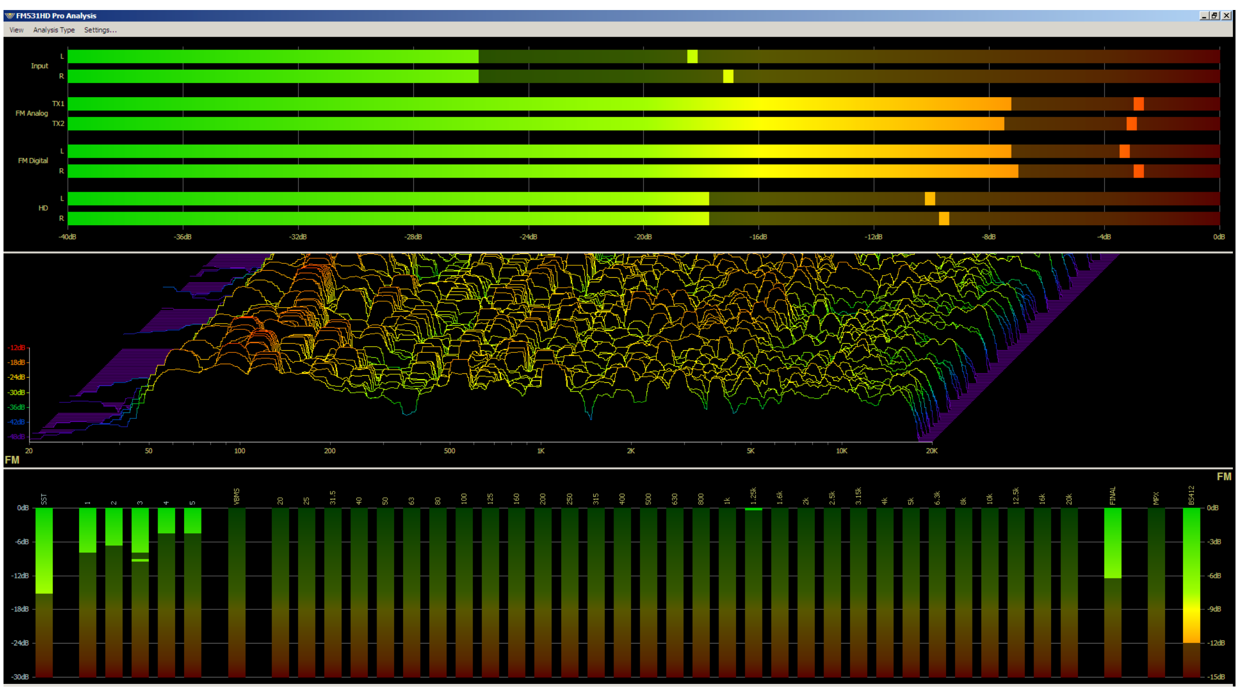
Flat Drag – Allows the dragging of dots together in a "flat" orientation.

Metering and Analysis

The FM-531HD 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 is a separate window. In true Windows fashion, the analysis window may be minimized on the desktop (taken off the screen) and/or closed by way of the familiar controls at the extreme top right.



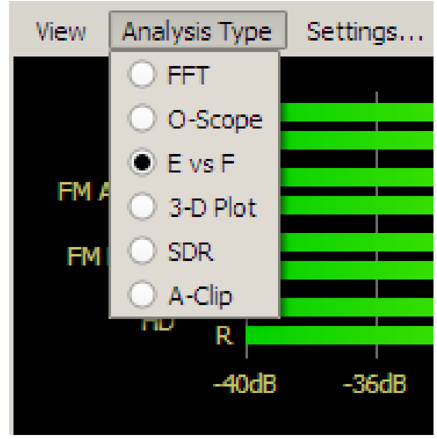
FM-531HD PRO GUI Analysis Window

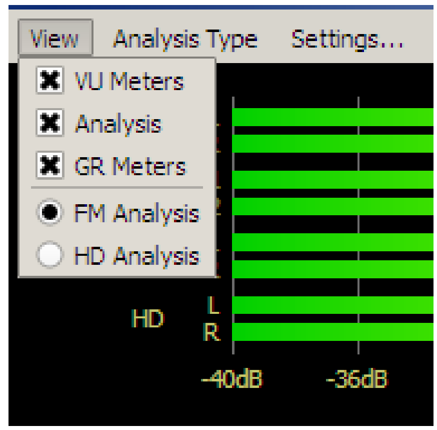
To learn more about the different types of analysis available, please see the section on Analysis for an in-depth look at what the different displays mean and how they can help visually adjust the sound of the FM-531HD. Accessing the different visualizations is simple. At the top of the analysis window is an option for *Analysis Type*. Click the text and dropbox will let you select the analysis you want to look at.

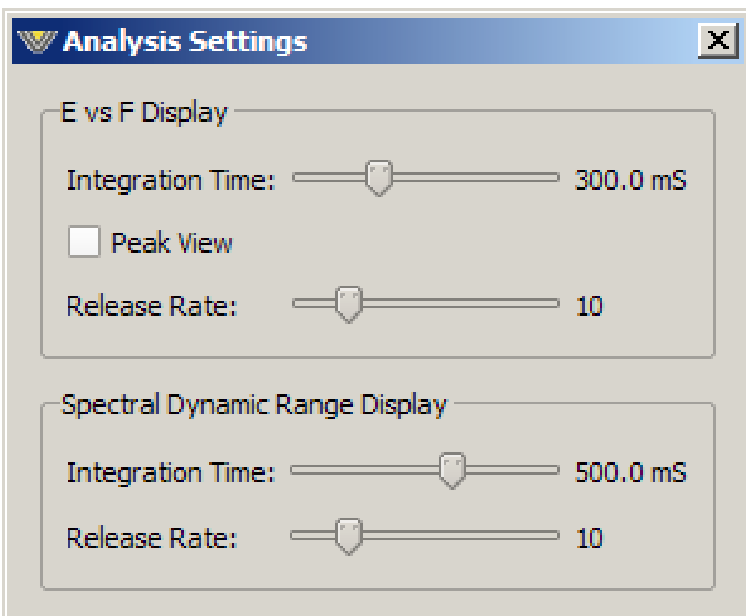
You can also decide to minimize the display by choosing which panels you want to monitor. The default is the I/O metering on top, the analysis in the middle window and the gain reduction meters in the bottom window. With the view tab, you can turn on or off any of those displays.

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.

The *Analysis Settings* window also allows you to adjust timings for the SDR display.







Control Area Region

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

All of the knobs, faders, switches, and check boxes 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

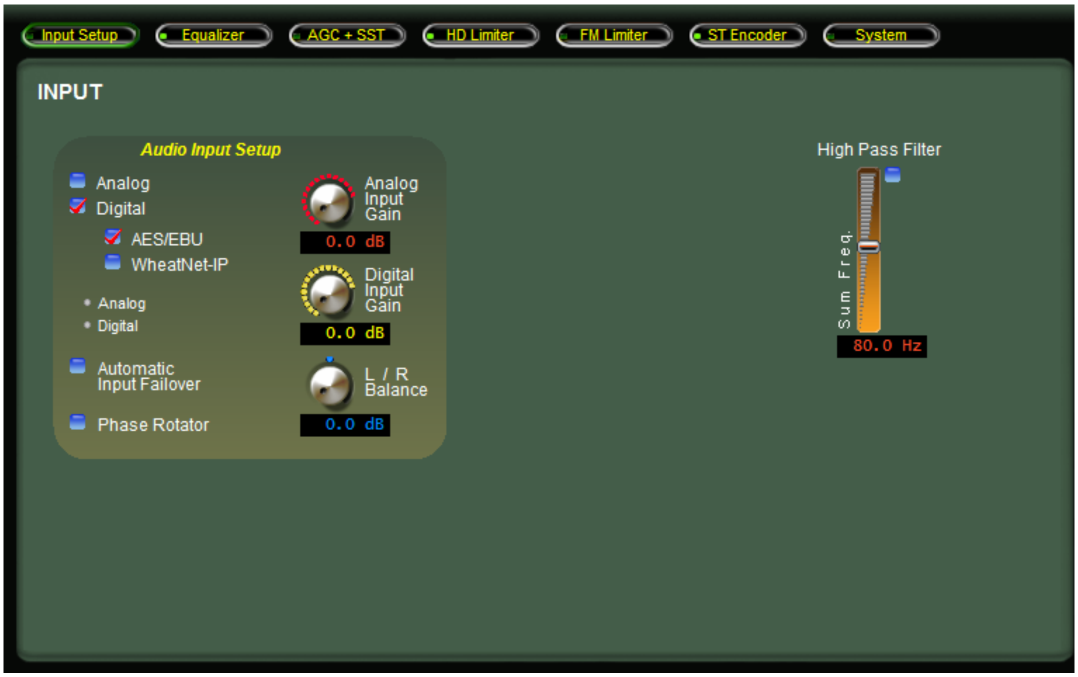
In most cases you can double click on the text that indicates the setting value (for example, the window that reads X.XdB under the *Analog Input Gain* knob on the *Input Section*) and a box will pop up where you can enter an exact value and click *OK*.

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 outline on the selection tab).



The *Input* screen contains controls for the input source selection: *Automatic Input Failover*, Analog and Digital input gain controls, input L/R gain balance, *Phase Rotator*, and the high pass filter.

Input Source

The Input Source can be selected as either Analog, which is the analog Left/Right input, or AES, which is the AES3 digital audio input. If enabled, an option for WheatNet-IP may also be selected.

The currently selected input is shown by a red check mark. In the example above, the input selected is DIGITAL with the AES/EBU option.

Input Signal Presence

Below the source selection check boxes are a pair of signal presence indicators. The Analog indicator lights if there is a signal present at the analog input that exceeds the threshold of -24dBu. The AES indicator lights if there is a valid AES signal present at the digital input and it is above -48dBFS.

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 there are invalid bits in the data stream or missing audio data. Also, the analog input will be selected after 30 seconds of a valid digital data stream having signal levels below -48dBFS.

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

NOTE: The input failover option can only switch between an analog input and a digital input. Failover from WheatNet-IP to AES/EBU (or vice versa) is not supported.

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 range of +12dB to -36dB.

Digital Input Gain

The AES/EBU and WheatNet-IP 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* can be adjusted over a range of +12dB to -36dB. This knob controls both the AES and WheatNet-IP input gain.

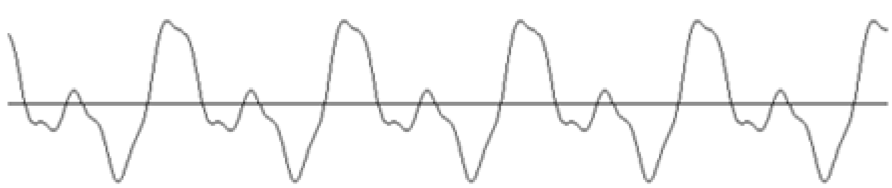
L/R Balance

Static level errors in Left/Right channel balance can be corrected by using the *L/R Balance* knob. The Left/Right balance can be adjusted over a +/- 12dB range. This control should only be used as a short term "Band-Aid." The relationship between left and right should be corrected upstream if it is found the audio is out of balance.

Phase Rotator

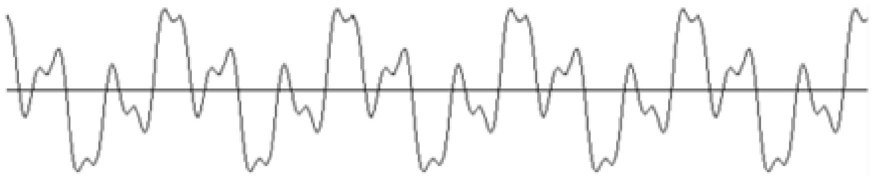
Human voice is usually asymmetrical by nature, which means that it is usually "peakier" in one polarity than the other. What the Phase Rotator algorithm does is fix these asymmetrical peaks. The way it works is by phase shifting the signal harmonics relative to their fundamental frequencies. This action 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 below. Notice how the upper and lower portions of the waveform now have equal distribution? No information is lost and, more importantly, no harmonic or intermodulation distortion is created by the Phase Rotator. The only thing that happens is that the signal's phase relationships are rearranged to make the waveform energy more uniform in polarity.



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.

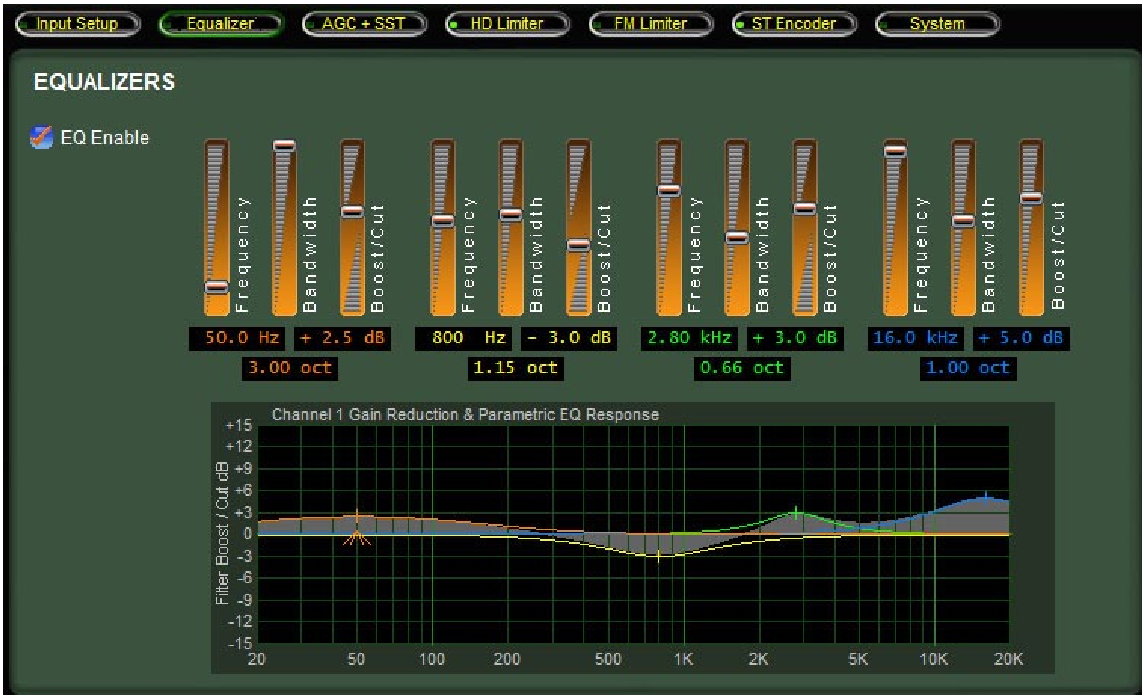
The *High Pass Filter* also removes subsonic energy that could upset the operation of equipment later in the audio chain, such as the AFC loop in the FM exciter.

The *High Pass Filter* has two operating modes:

Off – If the *High Pass Filter* box is not checked the High Pass Filter is not enabled.

Stereo – If the *High Pass Filter* box is checked the High Pass Filter operating mode is stereo and both left and right audio channels are treated equally by the high pass filter frequency.

EQUALIZER MENU

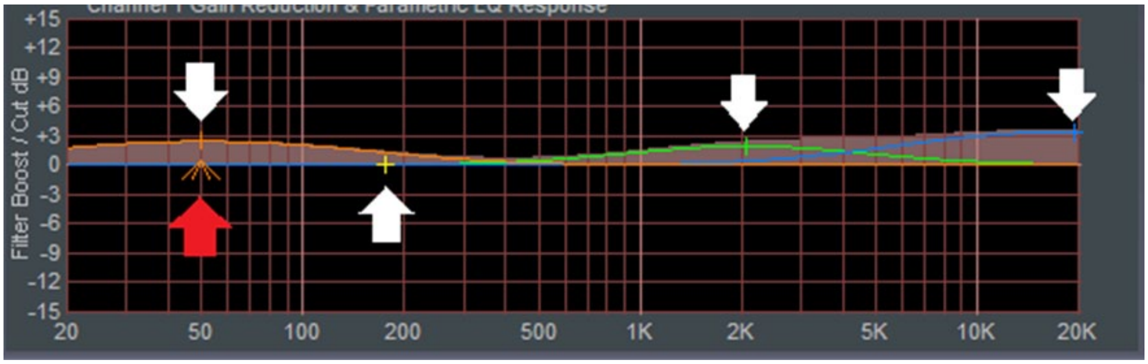


The *Equalizer* tab has four identical and independent bands. Each of the four bands can be switched in or out independently, and is fully adjustable in three ways:

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 equalizers 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 equalizers is by directly manipulating the actual curves on the graphical screen below using the mouse. 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.



Clicking on the crosshair of a curve (see white arrow) 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 level (Boost or Cut) above or below the 0dB reference line (up or down move).

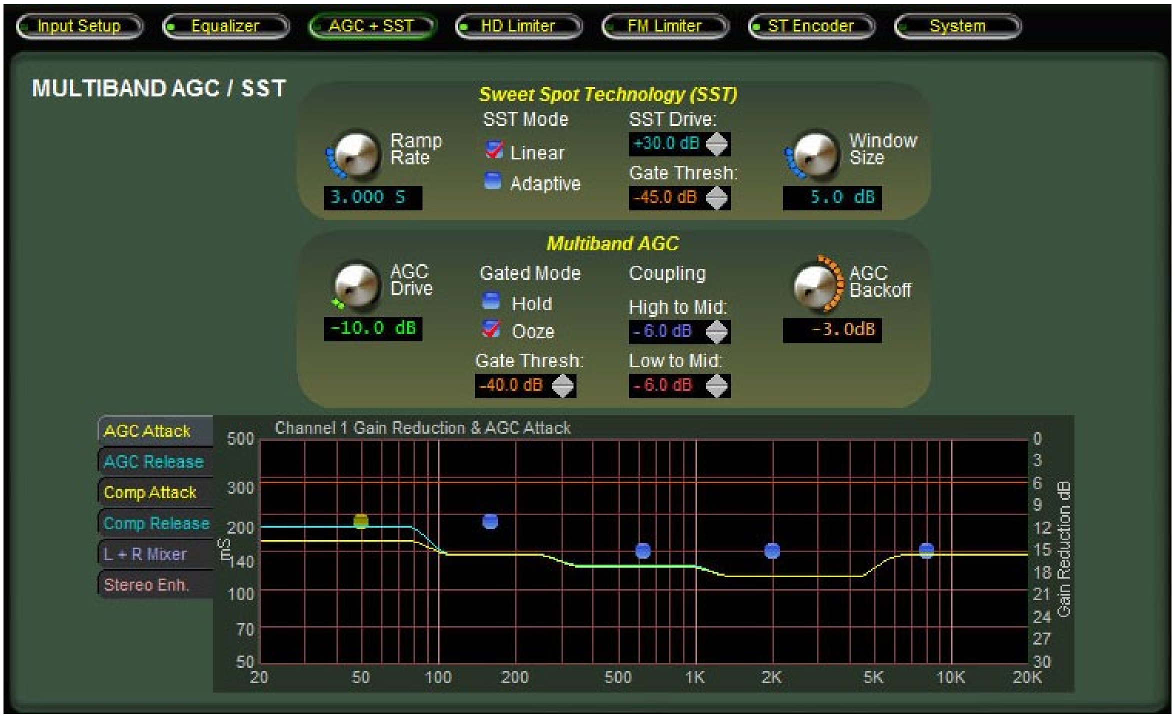
Clicking on the small "tent" underneath (see red arrow), and holding the left mouse button down while sliding the mouse left or right allows changes the bandwidth, or "Q," of the equalizer section.

EQ Enable

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

AGC + SST MENU

The FM-531HD 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.



What is SST?

Unique to Wheatstone models of processing is a special algorithm called SST, which stands for Sweet Spot Technology™. The SST incorporates control signals from the input audio, the multiband crossover, and the AGC and Compressor behaviors in order to derive a special control signal that is used to force the multiband AGC/Compressor to operate within a specified gain reduction window. By doing so, the medium and long term "sound" of the processor is quite stable as program elements and incoming audio levels wander about.

Without the contribution of the SST, the AGC and Compressor sections would not be able to maintain as consistent a spectral and dynamics texture over widely ranging signal input conditions. In fact, they would have little advantage over the technology used for generations in other audio processors which all suffer from the inability to manage long-term spectral consistency.

A "broadband AGC" is not used prior to the AGC and Compressor because such devices know nothing about the output of the multiband section, and it is this output that determines the overall texture of how a radio station sounds on the air. By using a multiplicity of control signals the SST is able to calculate a correction signal that exactly manages the behavior of the multiband AGC and compressor in order to create a consistent signature sound across a very wide variety 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 type of program-related processing occurring in the Compressor algorithms. While the SST section operates in Left/Right mode, the AGC and Compressors 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 signal fed to the algorithm that measures the dynamic behavior of the five band AGC. The higher the SST Drive setting the deeper the control depth in the algorithm – which allows it to operate over the widest possible program dynamic range.

A typical amount of SST control depth that affords the best tradeoffs in operation will be somewhere between -12 and -15dB as seen on the gain reduction display. Note that there is no harm in generating more or less control depth as long as one understands that more control depth affords greater long term control of audio levels and less control depth affords less. The overall control range is +30.0dB to +40.0dB.

SST Gate Threshold

The SST algorithm has been provided its own Gate Threshold so that its operation may be optimized when it is combined with the five band AGC. Typically the SST Gate Threshold will be set several dB higher (less negative) than that of the five band AGC. The overall control range is -60.0dB to -35.0dB.

The SST Gate Threshold control sets the audio input level where SST correction is turned off and its gain is frozen.

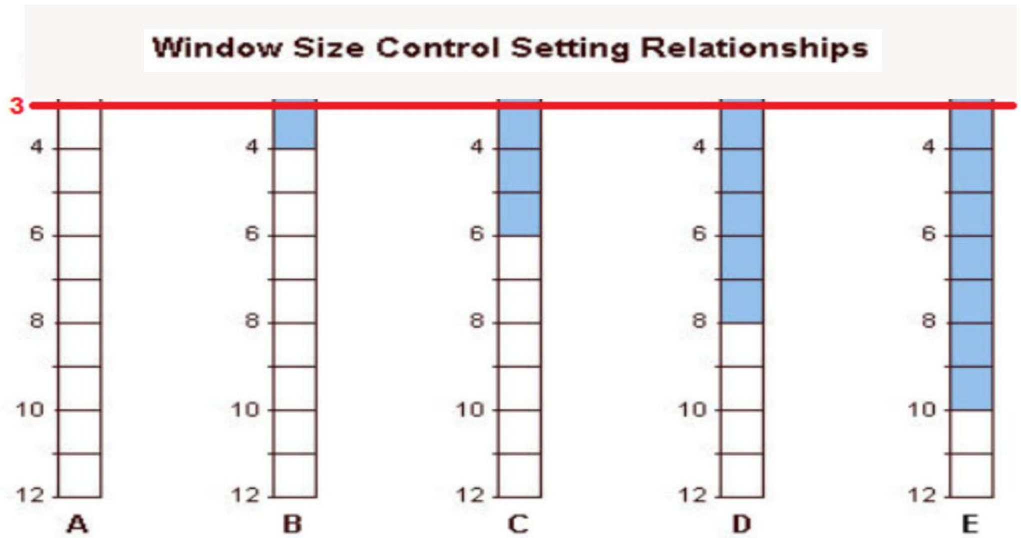
Ramp Rate

In order for the SST to behave with all types of program requirements it must have its speed of correction tailored for the application and this is what the Ramp Rate control does. The control is calibrated in units of time, and therefore describes the rate at which the SST causes more or less activity to occur within the five band AGC/Compressor.

A good typical starting value for SST Ramp Rate is 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" the size of that sweet spot needs to be able to be set for different kinds of program requirements. See the graphic below for some examples:



The graphic shows the full potential of the control for a visual explanation of how the control works. The red line indicates the user adjustable lower limit of the SST at 3dB. The upper limit is 12dB

The size of the shaded areas in the above graphic serve to demonstrate the size of the error (in dB) permitted to occur within the five band AGC before the SST corrects it. It is important to note that the SST has no control over allowing the five band error to go positive – that is specifically handled by the five band AGC itself. As the shaded areas show, the larger the Window Size, the larger the "error" allowed by the SST before correction takes place.

Very small Window Size settings can cause the SST to continuously try to correct for minute behavioral errors within the five band AGC. Likewise, very large Window Size settings will allow large AGC errors to go uncorrected. Settings at either extreme, while probably not doing anything wrong to the audio, are not particularly useful, especially when these settings are on the smaller Window size, which is why this control has been limited to 3dB as the lowest setting.

The usual setting of the Window Size control for the majority of program content will be between 4.5dB and 7.5dB. The full range of the control is 3dB to 12dB.

SST Linear/Adaptive Options

The two final controls are options for the SST, Linear and Adaptive. Only one can be chosen. In the Linear mode, the SST makes adjustments to the five band AGC/Compressor at the same user defined rate regardless of program level. In the adaptive mode, the SST makes adjustments to the 5 band AGC/Compressor at faster rate than selected by the user if there is a dramatic change in program level. The linear mode is advised in most cases. The adaptive mode is suggested in cases where program levels vary widely. To defeat the SST, make sure both options are unchecked.

A Word (or two) About Outboard Processors

There are a number of popular outboard AGC/Compressors on the market that have traditionally 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 of a poorly designed AGC in the main processor.**
- **Level protection for an STL path**
- **Level enhancement for ratings encoders.**

Of course, we feel that the SST algorithms in the FM-531HD eliminate most of the concerns mentioned above. By processing as we do with SST, we have removed the need for a wideband AGC directly in the audio path while keeping the benefit of what a wideband AGC can do (which is usually keeping the processing after it in its "Sweet Spot"... thus "Sweet Spot Technology" or "SST"). No other all-in-one processing solution like the FM-531HD processes with the accuracy of SST.

We also feel that the amount of control in our Pro GUI allows the user to virtually set the FM-531HD to whatever sonic texture they can think of, eliminating the need for an outboard box to emulate the same.

The advent of 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.

We also feel that if the levels in your plant are properly calibrated, there should be no need for an AGC in front of an STL path. Almost all professional broadcast gear that would be in-line with a console before and including the STL will have as much if not more headroom than your broadcast console. If non-broadcast or non-professional gear is in your audio path, it should be replaced ASAP with a suitable device. Consumer grade equipment in the program line is never advised and will only spoil the financial investment you have made in the FM-531HD.

All this being said, there may be some cases where, no matter what, an outboard processor is needed or desired.

- **Ratings Encoders** – We have found that ratings encoders do need a "steady diet" of incoming level in order to be happy in certain cases. We find this to be problematic at stations that are automated for much of their broadcast day (where a human is not present to make adjustments to the source level on the console). In many cases, levels can be as much as 10dB different from source to source when care is not taken when audio files are ingested into a playout system. While SST can easily compensate for these level differences, a 10dB drop in level will not make a barefoot ratings encoder happy over the course of a 3-4 minute song, causing it to alarm and email and page the world. On weekends, that can be a BIG pain.

The BEST way to fix this problem is to make sure there are standards in place for proper levels when recording into your playout system. Unfortunately, that isn't always practical in this era of downsizing as there aren't enough warm bodies to do the work. So, for stations that feel they need to use a processor ahead of a ratings encoder, we suggest setting the FM-531HD multiband control gate between -30dB and -20dB and placing the SST in Linear mode. If aggressive AGC is needed, adjust the AGC Backoff in the FM-531HD towards 0dB.

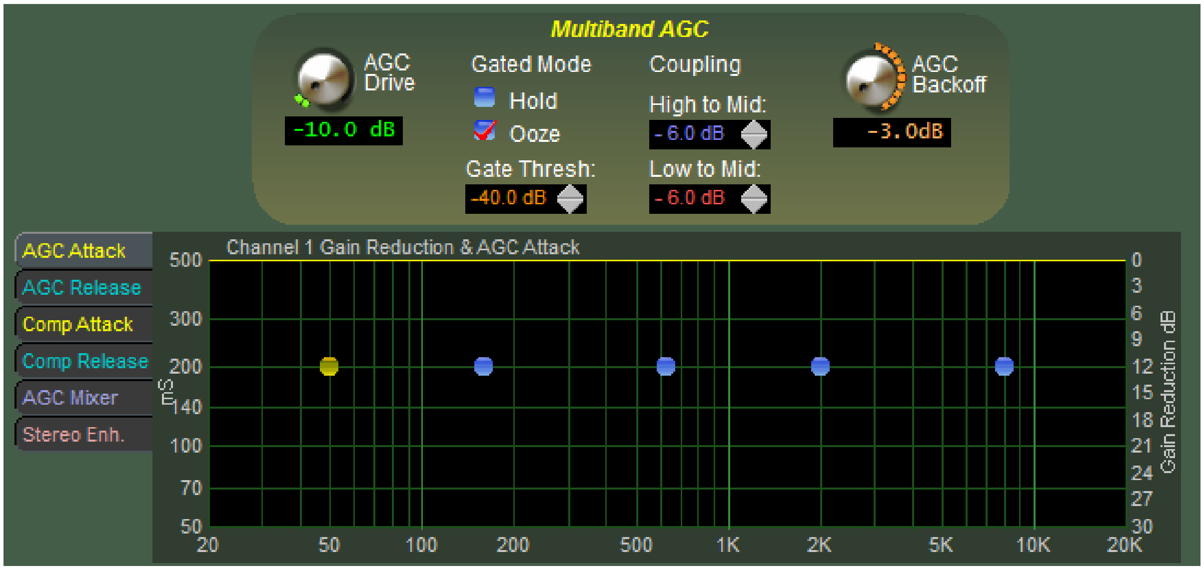
- **Reverb** – Reverb was originally used as a way to increase density into older AM transmitters when processing wasn't sophisticated enough to do the same. The effect of reverb not only did the trick at the time, it also added a pleasant sonic texture to the audio that is desired by many stations today. The problem with reverb units is that the input level controls everything about how the reverb will sound on the air if the unit is placed directly into the program chain.

A pre-processor is traditionally used ahead of a reverb unit to prevent excessive reverb due to sloppy levels from a console. A pre-processor also sustains program audio somewhat to get the most out of the decay effect in a reverb unit. Just like adding an AGC to enhance levels to a Ratings Encoder, the same steps should be taken to adjust the FM-531HD to "mate" with the outboard processor. If reverb is sidechained into a microphone processor and not directly in the audio path, no special adjustments of the FM-531HD need 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 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 FM-531HD 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, the gain control will become fixed at -12dB and adjustment of the 31-band limiter drive may need to be made for any offset that the fixed level creates. It is because of these changes that slowing down SST and the five band AGC/Compressor be exhausted before sections of the processor are bypassed.

Multiband AGC Controls



AGC Drive

The *AGC Drive* control adjusts the amount of signal into the AGC/Compressor, primarily determining the amount of compression that can be achieved. Driving harder (higher numbers) results in a more consistent on air sound, with an increased capability to pull up low passages in program material. Less drive creates a more “gentle” on air sound and has less capability to bring up low passages.

The *AGC Drive* control simultaneously adjusts the sum and difference channels. The range of this control is -12.0dB to +6.0dB.

Gated Mode

The *Gated Mode* controls determine how the AGC behaves in the absence of audio.

When *Ooze* is selected and the audio input falls below the AGC Gate Threshold, rather than the gains of the bands “holding” they will slowly return towards 0dB. This setting may be useful for formats that contain material with very wide dynamic range or for certain speech-based formats where the “Gate Delay” setting may not be appropriate.

When *Hold* is selected and the audio falls below the AGC Gate Threshold value the gains of the five bands will “freeze” at their current values and will hold at those values until audio is present that is above the AGC Gate Threshold. This prevents the increasing of gain in the absence of audio which would likely increase background noise.

AGC Gate Threshold

The AGC *Gate Thresh* control sets the audio level below which the audio gain reduction values are held until audio returns. The control can be set to OFF or adjusted over the range of -79.0dB to -20.0dB.

Band Coupling

The *Coupling* control has one specific purpose – to keep the gains of bands 1, 2, 4, and 5 from exceeding the gain of band 3 when those “outer” bands are “ungated.” This prevents unnatural gain rush-up by those bands when such behavior would be undesirable. Without the “Coupling” feature the gains of the bands at the frequency extremes can increase their gain and cause a spectrally incorrect sonic mixture at the output of the five band section.

Band 3 was chose as the “master” band for the high and low coupling control because that band is in the approximate center of the audio spectrum.

The *Coupling* control may be adjusted over the range of “0dB” to “-6dB” (High to Mid) or “0dB” to “-12dB” (Low to Mid). When each coupling setting is set for “0dB” bands 1, 2, 4, and 5 can never have more gain than band 3. Such a setting is useful in formats such as Classical and Jazz that typically would not wish the spectral “correction” caused by the outer bands increasing their gain but still need those bands to reduce their gains when appropriate.

In typical programming situations the best setting of the *Coupling* control will likely be somewhere between -2.0dB and -4.0dB. -6.0dB (-12dB Low to Mid) was chosen as the maximum setting. Any further decoupling at the point allows the outer bands to “wander” around too much, causing an unnatural spectral buildup on some material.

AGC Backoff

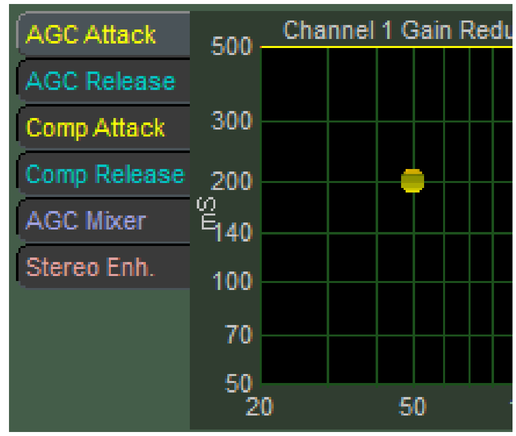
The *AGC Backoff* adjusts the processing balance between slower AGC action and faster compression.

With the *AGC Backoff* control set at “0,” the audio gain control is primarily due to slower AGC action, and therefore short term program dynamics are largely unaffected unless they exceed the AGC gain reduction by at least 1dB. In that case momentary gain reduction by the compressor will place the gain where it needs to be to return control to the AGC.

With the *AGC Backoff* control at the other extreme, or -6dB, gain control is primarily due to fast compression and short term dynamics will be aggressively controlled. The setting of the *AGC Backoff* control allows the audio compression texture to be modified from a gentle, almost unobtrusive control (*AGC Backoff* between “0” and about minus 2.0dB), to very obvious compression (*AGC Backoff* control at “-3.0dB” or greater negative numbers) creating a sound that is similar to 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 and there is no other method for changing the relative thresholds between the AGC and compressors. If for some reason “less” AGC control signal is desired, it may be achieved by lengthening the AGC attack times.

AGC/Compressor Sidebar Menu



It should be noted first that 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 "FM-531HD Graphical Interface Operation" section of this chapter.

AGC Attack

This control determines how quickly the AGC will respond to changes in the audio. The range is 50ms to 500ms. 300ms is a recommended starting point.

AGC Release

This control determines the rate at which the AGC will increase gain. The range is 1 second to 7 seconds. 7 seconds is VERY slow and recommended for fine-arts or classical programming. 1 second is very fast (for an AGC) and is recommended only if a certain sonic texture is desired. The recommended settings are between 2 and 5 seconds.

Compressor Attack

This control determines how quickly the Compressor will respond to audio not caught by the AGC. The compressor deals with short term peaks. The recommended setting for the compressor to sound "musical" is between 10 and 30ms. Compressor settings faster than 10ms can make the audio sound constricted. Conversely, settings slower than 30-50ms can prove to be ineffective and can make the 31-band limiter work harder than it needs to.

Compressor Release

This control determines how quickly the compressor will return to the processing "floor" established by the AGC. The five band AGC/Compressor are tied together, with the five band AGC establishing the level of gain reduction and the compressor adding additional compression to the AGC. The Compressor is never allowed to achieve more gain than the AGC; that is, it will release to the AGC's platform. The release time 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 should be 300ms. This is not a rule, just a good starting point if you are building a preset from scratch.

AGC Mixer

The *AGC Mixer* is the output of the L+R content of the processed signal. Very dramatic EQ changes can be made here, however, it is recommended that these controls are set flat and the parametric EQ is used for any EQ changes. If you find you need something in addition to the parametric EQ, these controls are here for your use.

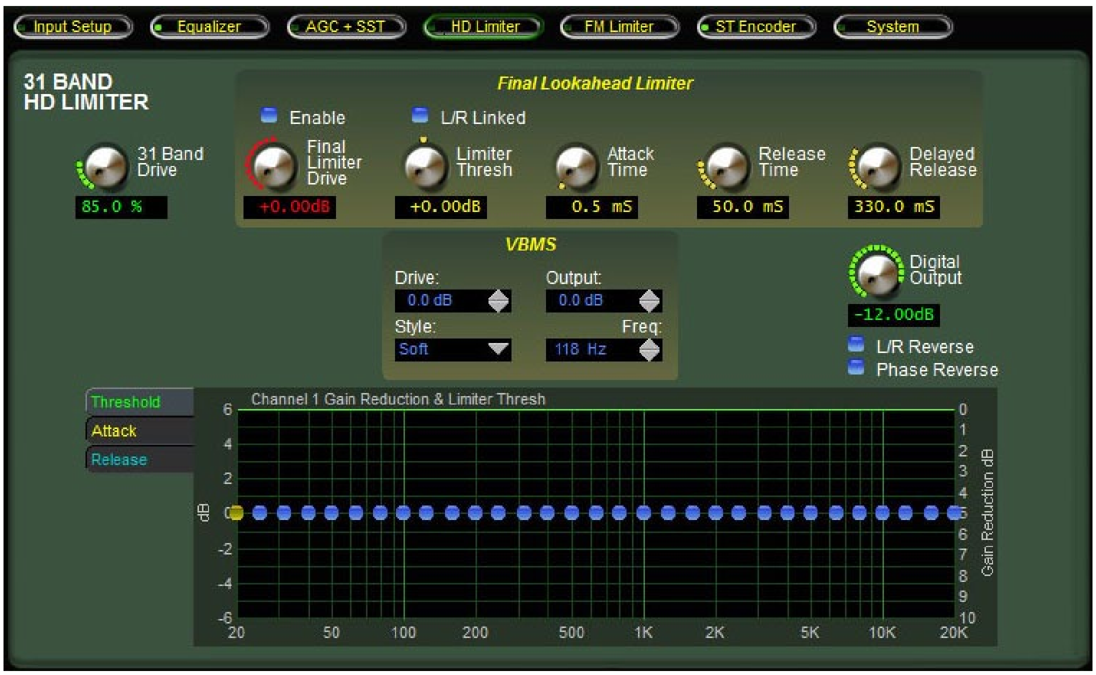
Stereo Enhance

Stereo Enhance is the difference output of the processed signal. Here, you can add stereo enhancement to specific parts of the audio spectrum without raising the overall

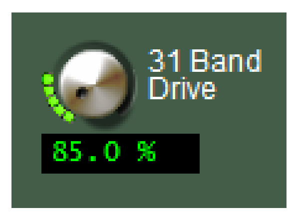
L – R gain. This simple step allows you to tailor the audio so that stereo enhancement will occur in bands most critical to the human ear without raising the L – R gain in the entire spectrum, which adds little audible benefit but can aggravate multipath. The most dramatic changes will be heard in Bands 3 & 4. A good way to add some stereo enhancement to the overall mix would be to raise Band 3 & 4 to between +2 and +3 dB.

HD LIMITER MENU

The *31 Band HD Limiter* screen contains the controls necessary to fine tune the sound of the HD output as well as adjust its peak control method.

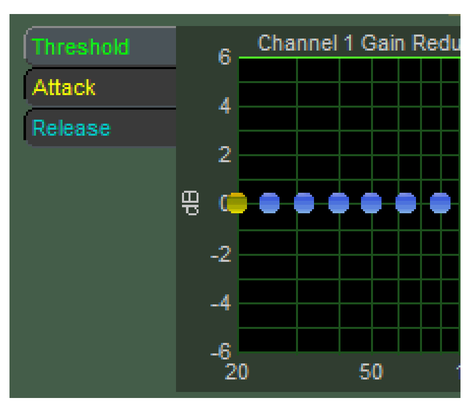


31 Band Drive



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. Higher settings make the sound more aggressive by increasing short term density. Extremely high drive settings which cause more than 5-6dB of continuous gain reduction are unnecessary, can become fatiguing, and may create a less than desirable operating condition for a perceptual codec. Drive is adjustable from 80 to 100%.

31 Band Limiter Sidebar



Just like in the five band AGC/Compressor, there is a sidebar with some special adjustments available to the user to tailor the sound of the 31-band limiter. It should be noted first that all bands of audio processing can be adjusted as a group or independently in the sidebar menus. For more information on how to adjust these controls, please see the "FM-531HD Graphical Interface Operation" section of this chapter.

Threshold

The *Threshold* control determines when limiting will start in a particular band. The range for the FM-531HD limiter threshold control can be adjusted over a 12dB range to offset the 31-band Limiter Drive.

Attack

This control determines how quickly the Limiter will respond to audio once it exceeds the limiter threshold. Recommended settings are between 30 and 10ms

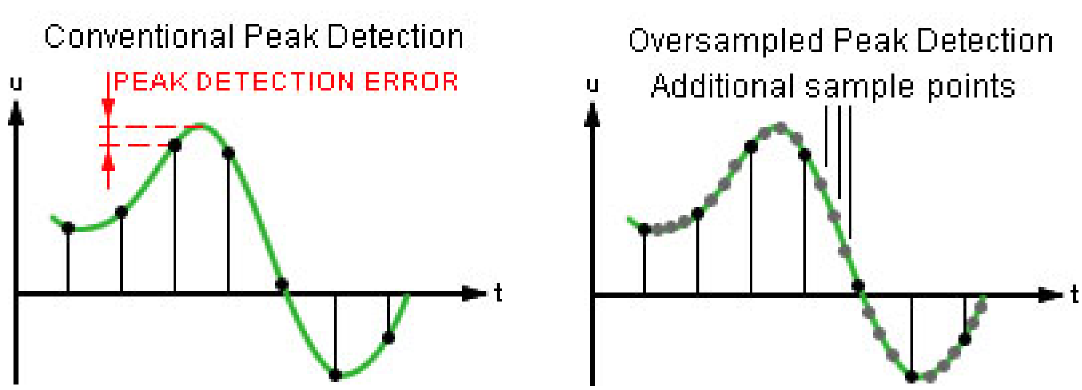
Release

The *Release* time determines how quickly the limiter will return to 0dB once audio has fallen below the limiter threshold. Faster release times will yield a louder sound while slower times will yield a smoother sound at the expense of loudness.

Because of the way the 31-band limiter works, there is a lot of leeway with attack and release times and these controls have been set so that the end user doesn't drive the car "off the cliff." Basically, only extreme settings will yield a totally undesired sound. That being said, these controls should also not be adjusted haphazardly.

Lookahead Limiter

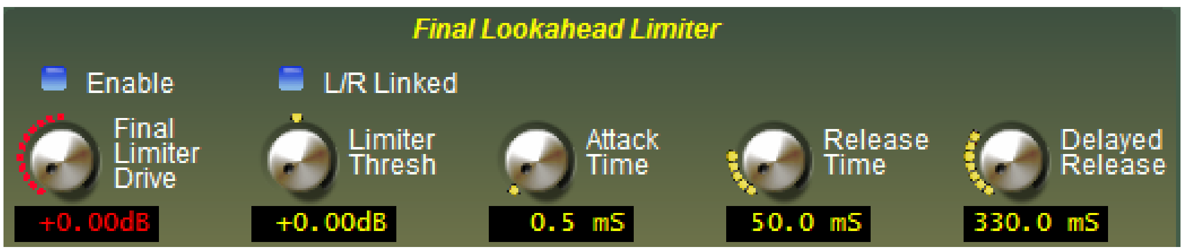
The output of the 31-band Limiter can be peak controlled by a special high-speed feed forward limiter with a 0.5 millisecond look ahead time. The control path, and not the audio path, is oversampled in order to achieve extremely precise peak control. Because the goal of oversampling is better peak control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



In a conventional peak detector, audio excursions occurring between individual samples are "missed," and therefore the audio signal is "under read" during this instant. This results in inaccurate peak control.

By increasing the sample rate of the detector to oversample the audio waveform only at the detector it becomes possible to read the audio waveform with much higher precision. This results in far more accurate peak control.

Note that it is not only unnecessary to oversample the main audio path, it is actually undesirable; it would result in the same problem a conventional detector has because the ratio of audio to detector sample rates would be the same as in the conventional case. Additional DSP resources would have been used with no benefit.



Limiter Enable

When the *Enable* check box is checked the look ahead limiter is enabled.

L/R Linked

When this box is checked the left and right channel control signals are common. This prevents unnatural stereo image shifts when one channel needs to reduce the signal to control peak excursions.

Final Limiter Drive

This control drives the output of the 31-band limiter into the final peak limiter. Care should be taken to monitor the FINAL meter in the analysis display to view the amount of gain reduction in the final HD limiter. Too much gain reduction can cause pumping.

Limiter Threshold

The *Limiter Thresh* control adjusts the relative threshold of the look ahead limiter compared to the setting of the Final Limiter Drive control and is adjusted until the onset of limiting is reached with normal program material as indicated by the gain reduction bargraph. Adjustable over a +/- 6dB range.

Attack Time

The look ahead Limiter *Attack Time* control adjusts 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 conventionally.

For example, if the attack time control is set to 1.0 milliseconds, the actual attack time is 0.5 milliseconds. This is because the 0.5 millisecond look ahead is always in circuit and its value must be subtracted from any adjustment of 0.5 milliseconds or greater in order to know the actual attack time.

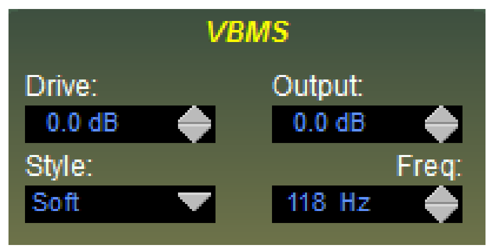
Release Time

The *Release Time* controls the primary (fast) release time of the look ahead 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 which can "muddy" the sound. The control ranges from 33.0 to 330.0 milliseconds

Delayed Release

The *Delayed Release* controls the secondary (slow) release of the look ahead limiter. Here, 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 ranges 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. Its primary purpose is to allow the thumps of kick drums and other percussion instruments to sound natural in their "percussiveness." The algorithm operates in left/right mode treating both channels identically.

Our laboratory research has revealed that it is perfectly acceptable to process very low frequency material using the VBMS prior to HD exciters provided that the harmonics caused by the VBMS are well controlled spectrally. Since our algorithm does not allow significant harmonic levels above the third or fifth harmonic (depending on the Style setting) there is no audible downside to using the VBMS 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 drive to the VBMS algorithm and is adjustable over a range of +/- 6dB.

Style

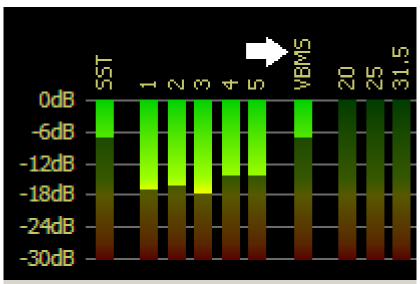
Two VBMS operating styles, plus Off, are available. The *Hard* setting has a transfer function occupying approximately 1dB of transition amplitude. The waveform control is fairly abrupt and generates a fair amount of harmonics; however, we eliminate everything above the fifth harmonic. The *Soft* setting operates with a transfer function occupying approximately 2dB of transition amplitude. Therefore the control is less sharp than the hard setting, generating even fewer harmonics.

While the two settings may appear to be almost the same, because the transition amplitude is 1dB greater on the soft setting there is a *significant* difference in how the two VBMS transfer functions *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 higher than 0.0dB add more bass sound to the overall mix, while settings lower than 0.0dB reduce the amount of bass. Both the VBMS *Drive* and the VBMS *Output* controls may be generally adjusted to taste. However, care should be exercised in the setting of the VBMS Output control because settings above about +2dB may cause the amplitude of bass to reach the final limiter and added distortion may result. Like VBMS Drive, the VBMS Output is adjustable over a range of +/- 6dB

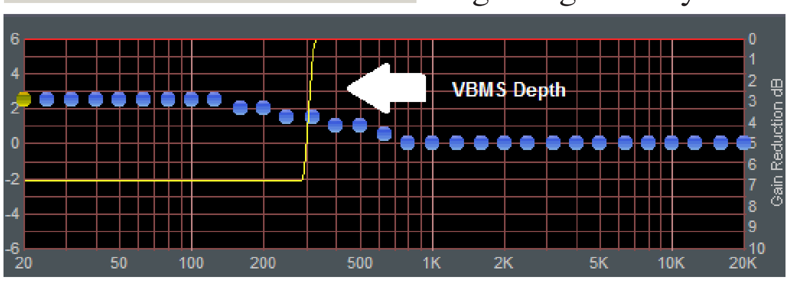
VBMS Frequency



The VBMS *Freq* control tunes the VBMS algorithm’s frequency of operation from 80Hz to 300Hz with the program energy that is not passing through the algorithm time aligned for coincidence regardless of the settings.

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 or in the Guru GUI. 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. This does not mean VBMS is not working or acting on the audio.

Digital Output

The *Digital Output* control adjusts the precise peak audio signal level at the HD Limiter of the FM-531HD. The resulting peak output levels in dBFS are indicated by the front panel and remote GUI bargraph meters. This control can be set to OFF or adjusted over the range of -79.95dBFS to 0.00dBFS.

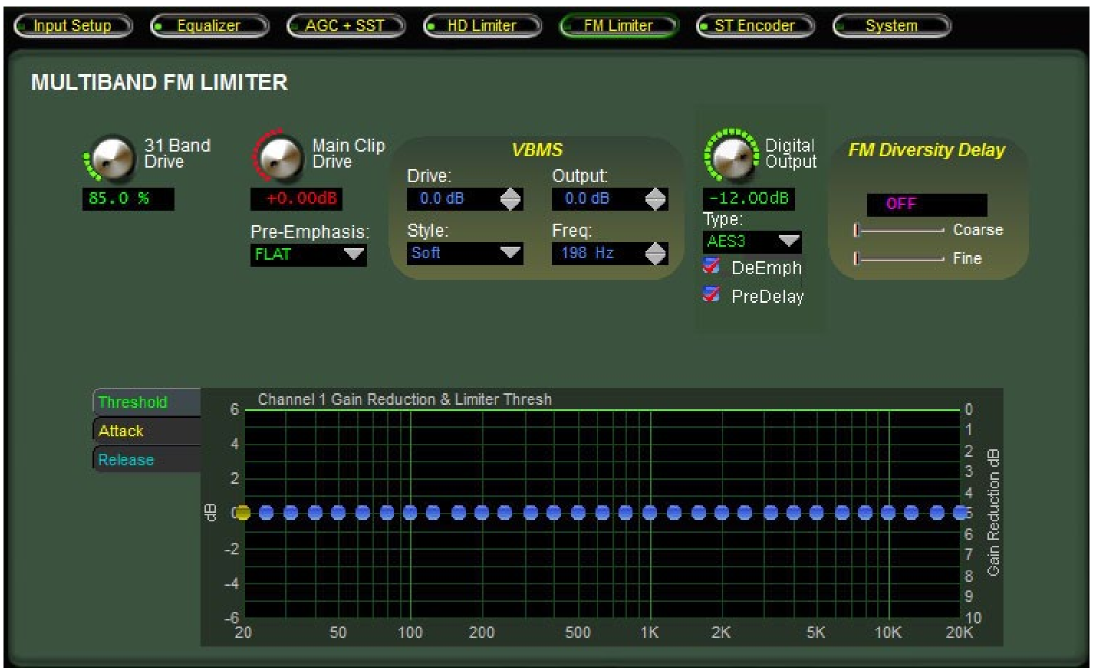
L/R Reverse

The *L/R Reverse* flips left and right on the HD output to correct for any wiring issues between the two channels. Phase reverse compensates for a known bug in certain software versions of older HD exciters.

Phase Reverse

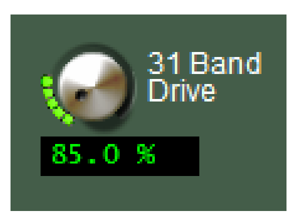
Certain software versions within the iBiquity HD Radio system have an inadvertent signal polarity inversion putting the analog and digital signals 180 degrees out of phase resulting in audible flanging as the two signals crossfade in a listener's HD receiver. When the *Phase Reverse* check box is checked,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

FM LIMITER MENU



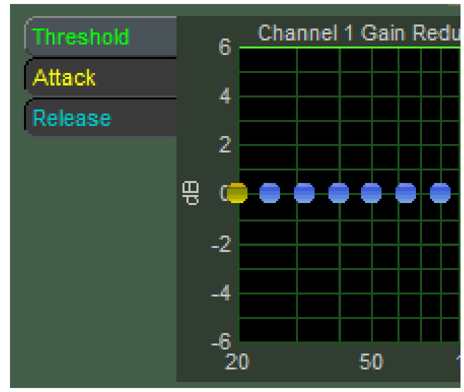
The 31-band *FM Limiter* screen contains the controls necessary for fine tuning the sound of the FM output and adjusting its peak control method. These controls allow the user to make many tradeoffs between loud and clean, in ways that have never been available before in an FM peak limiter.

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. Higher settings make the sound more aggressive by increasing short term density. Extremely high drive settings which cause more than 5-6 dB of continuous gain reduction are unnecessary, can become fatiguing, and may create a less than desirable operating condition for a perceptual codec. The *31 Band Drive* control is adjustable from 80 to 100%.

31 Band FM Limiter Sidebar



Just like the five band AGC/Compressor, there is a sidebar with some special adjustments available to the user to tailor the sound of the 31-band limiter. It should be noted first that all bands of audio processing can be adjusted as a group or independently in the sidebar menus. For more information on how to adjust these controls, please see the "FM-531HD Graphical Interface Operation" section of this chapter.

Threshold

The *Threshold* control determines at what level limiting will start in a particular band. The range for the FM-531HD 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.

Attack

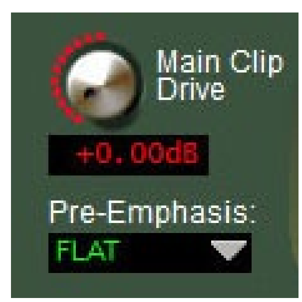
The *Attack* control determines how quickly the limiter will respond to audio once it exceeds the limiter threshold. Recommended attack time settings are between 30 and 10ms. Longer attack times are possible for frequencies below the VBMS frequency setting. Overall, longer attack times will cause the look ahead limiter to work harder and/or overshoot.

Release

The *Release* control determines how quickly the limiter will return to 0dB once audio has fallen below the limiter threshold. Faster release times will yield a louder sound while slower times will yield a smoother sound at the expense of loudness.

Because of the way the 31-band limiter works, there is a lot of leeway with attack and release times and these controls have been set so that the end user doesn't drive the car "off the cliff." Basically, only extreme settings will yield a totally undesired sound. That being said, these controls should also not be adjusted haphazardly.

Main Clip Drive



The *Main Clip Drive* controls the absolute peak output level of the 31-band section and is adjustable +/-6dB. The resolution of this control is 0.05dB, fine enough for the most exacting output level adjustments.

Pre-Emphasis

FM *Pre-Emphasis* is selectable as 50 or 75 microseconds, or Flat (OFF).

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. Its primary purpose is to allow the thumps of kick drums and other percussion instruments to sound natural in their “percussiveness.” The algorithm operates in left/right mode treating both channels identically.

Our laboratory research has revealed that it is perfectly acceptable to process very low frequency material using the VBMS prior to all modern FM exciters provided that the harmonics caused by the VBMS are well controlled spectrally. Since our algorithm does not allow significant harmonic levels above the third or fifth harmonic (depending on the Style setting) there is no audible downside to using the VBMS 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 drive to the VBMS algorithm and is adjustable over a range of +/- 6dB.

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; however, we eliminate everything above the fifth harmonic. 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 even fewer harmonics.

While the two settings may appear to be almost the same, because the transition amplitude is 1dB greater on the soft setting there is a *significant* difference in how the two VBMS transfer functions *sound* on actual program material.

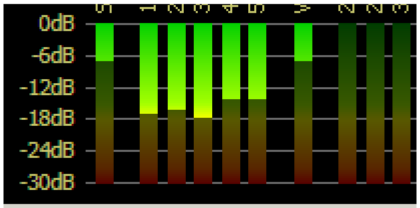
Output

The VBMS *Output* control adjusts the output of the VBMS algorithm and can be thought of as a “low bass mix” control. Settings higher than 0.0dB add more bass sound to the overall mix, while settings lower than 0.0dB reduce the amount of bass. Both the VBMS *Drive* and the VBMS *Output* controls may be generally adjusted to taste. However, care should be exercised in the setting of the VBMS *Output* control because settings above about +3dB may cause the amplitude of bass to reach the master clipper and added distortion may result. Like VBMS Drive, VBMS Output is adjustable over a range of +/- 6dB.

VBMS Frequency

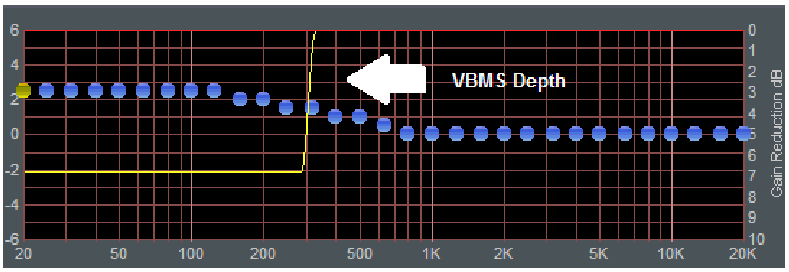
The VBMS *Freq* control tunes the VBMS algorithm’s frequency of operation from 60Hz to 300Hz with the program energy that is not passing through the algorithm time aligned for coincidence regardless of the settings.

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 or in the Guru GUI. 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 see much activity with VBMS. In fact, only when a lot of bass is present in source material will VBMS show any activity on the meters. This does not mean VBMS is not working or acting on the audio.

Digital Output

The *Digital Output* control adjusts the precise peak audio signal level at the FM Limiter of the FM-531HD. The resulting peak output levels in dBFS are indicated by the front panel and remote GUI bargraph meters. This control can be set to OFF or adjusted over the range of -79.95dBFS to 0.00dBFS.

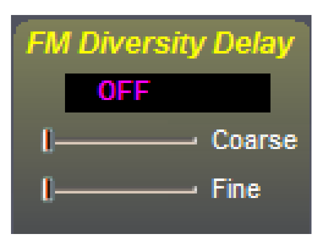
Digital output available as AES3 digital signal or baseband192 digital multiplex signal.

AES3 Digital

DeEmph – When checked, this option applies de-emphasis to the FM digital outputs. The type of de-emphasis is determined by the pre-emphasis selected. If no pre-emphasis is selected, this button is defeated.

PreDelay – When checked, the digital FM output will always be fed before the diversity delay.

FM Diversity Delay



The FM-531HD contains an internal digital delay to compensate for the corresponding delay in the HD Radio codec. The delay section is capable of providing up to ten seconds of compensating delay insteps of 100 microseconds.

The *Coarse* delay is set first to align the analog and HD signals as closely as possible, and then the *Fine* delay control is used to trim the delay to precisely compensate for that in the HD channel. Generally it is a good idea to set the Fine delay to midrange before adjusting Coarse delay. Resolution of the Coarse delay is 50 milliseconds.

The *Fine* control covers a range of 50 milliseconds in 0.1 millisecond (100 microsecond) steps. If the *Fine* control does not set the increment exactly to where it is desired, double clicking on the delay value text box will open it for manual entry of the delay time.

When the Delay controls are both set to their minimum settings, "OFF" appears in the delay window to signal that there is no delay at all in the circuit.

Entering Delay Values

You can double click on the delay value window to directly enter a numeric delay value in milliseconds. This is useful when you know roughly how much delay is required or wish to try various values quickly. You may also hold down the Shift key on your keyboard while adjusting a delay slider to effect faster adjustment rates, or hold down the Ctrl key to achieve finer control.

STEREO ENCODER MENU

The *Stereo Encoder* screen contains the controls required to configure the FM-531HD's internal DSP-based precision stereo generator.

MPX Processor Style

The FM-531HD is provides *two* separate ways to process the composite stereo waveform:

Composite Clipper

Uses a high ratio transfer function clipper to precisely control peaks without generating very high order distortion. This clipper has a brighter sound than the look ahead limiter and also creates more high order 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 more tightly defined peak amplitude.

Look Ahead Limiter

Uses 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 look ahead peak limiter in the FM Limiter screen, only at a far higher sample rate.

If you place the MPX Processor in the Look Ahead Limiter mode and set the drive to 0dB (or lower), all composite processing is basically defeated.

MPX Process Drive

The *MPX Process Drive* control adjusts the input level to whichever composite processor is selected by the *MPX Processor Style* check box. It has a range from -6.0dB to +6.0dB.

Multipath Controller

The Stereo Encoder (Stereo Generator) contains a special algorithm called Multipath Controller. 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 as this is the purpose of stereo enhancement, to exaggerate the stereo image width by increasing the level of L-R energy. When taken to the extreme the extra L-R energy can cause either unnatural sounding audio on some program material (the hole in the middle effect), odd behavior in most stereo receivers, or both.

Another thing to remember is that in FM Stereo broadcasting each dB of L-R increase results in a like decrease in the L+R, and it is the L+R signal that is solely 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 Multipath Controller does.

There are eleven operational choices available under the *Multipath Controller* 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 *Multipath Controller* 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 good starting point for this control is 70% which constrains the L-R to about 3dB below the L+R.

With the Multipath Controller 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 Stereo, -3dB, -6dB, and Mono settings within its stereo generator control menu. In that product, setting the control to -3dB causes a static 3dB reduction in stereo separation regardless of the incoming program material! The Wheatstone approach is more intelligent in how it works; it does not decrease stereo separation on program material that does not need it.

BS412 MPX Power Controller

The Wheatstone 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.

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."

To prevent accidental activation of the BS 412 Power Controller, one must *Enable* the controller first even before making adjustments from the OFF position.

TURNING ON THE BS412 MPX POWER CONTROLLER

IMPORTANT! ***The BS412 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.***

When the BS412 control is moved away from its OFF position the MPX Power controller is engaged and the algorithm begins measuring and controlling the processor's total MPX power according to the ITU-R BS.412-7 standard. Because the measurement is essentially an average value integrated over a very long time the MPX Power controller will not make quick adjustments to the MPX Power when first engaged.

As the controller integrates the MPX energy the drive to the clippers will be modified until the measured MPX power satisfies the reference level as set by the Stereo Encoder menu's 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

If the BS412 Controller has been on for more than a few minutes and the BS412 control is set back to the OFF position it will take some time for the MPX Power to return to an unregulated state. This is because the BS412 controller is not turned OFF by bypassing it (which could cause abrupt changes in output level!), but is instead defeated by raising its MPX Power detection threshold well above any conceivable or achievable audio density.

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

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 compared to that of the 38kHz subcarrier allows the FM-531HD to compensate for nonlinear time delays in the transmission system that would otherwise disturb the phase relationships between the 19kHz stereo pilot and the 38kHz stereo subcarrier and cause degradation of stereo separation.

Stereo/Mono

The stereo encoder may be operated in Stereo or Mono by selecting the desired operating mode with this switch.

SCA 1 & 2 Gain

Inputs applied to the SCA connectors are digitized at 192kHz. Prior to being digitized the signals are high pass filtered at 1kHz to reduce hum and then brick wall low-pass filtered at 94kHz. These controls can be set to OFF or to levels from -79.95dB to +10.0dB in 0.05dB steps.

TX 1 Level

The *TX 1 Level* controls the output appearing on the BNC connector for TX 1. This control can be set to OFF or to levels from -79.95dB to +6.00dB in 0.05dB steps. Reference is +2.40dB.

This control also adjusts modulation levels for the baseband192 digital multiplex signal, noting that as of this writing transmitter manufacturers only support signals on the TX-1 channel.

TX 2 Level

The *TX 2 Level* controls the output appearing on the BNC connector for TX 2. This control can be set to OFF or to levels from -79.95dB to +6.00dB in 0.05dB steps. Reference is +2.40dB.

This control also adjusts modulation levels for the baseband192 digital multiplex signal, noting that as of this writing transmitter manufacturers only support signals on the TX-1 channel.

If TX 2 Pilot Only is checked, the TX 2 output becomes a 19kHz sine wave source.

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 wideband, line level composite stereo audio and, as in the L/R case below, 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* check box is selected. We have also included a special frequency, “Bessel,” which is a 31.250kHz test signal that can be used when the first Bessel null is being utilized for modulation monitor calibration.

SYSTEM MENU

The *System* screen allows system-wide settings to be configured. In the first third of the screen is a very important box 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 FM-531HD SYSTEM, or global, parameters, which are saved with presets but not restored when presets are taken, unless the *Input/Output Settings Follow Preset Takes* check box 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

BS412

SCA 1

SCA 2

TX 1

TX 2

Pilot Only

Analog - L/R, L/R Deemph, and MPX output style choices

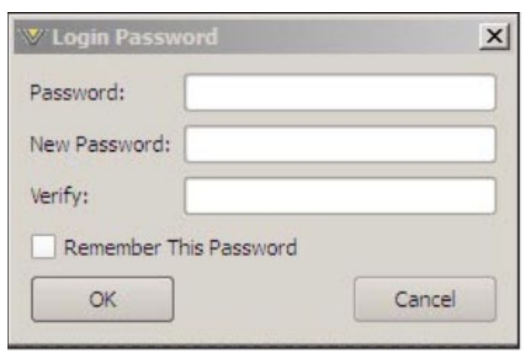
The Input/Output Settings Follow Preset Takes choice allows the FM-531HD to modify its input and output gain settings whenever presets are recalled. Input and output gain settings are always stored with the presets when processing values are stored, but those settings will be ignored upon preset recall when the *Input/Output Settings Follow Preset Takes* check box is unchecked.

Metering Data Over TCP/IP

The FM-531HD utilizes the UDP protocol by default for transmitting metering data to the remote GUI (TCP with its guaranteed packet delivery is always used for control).

In some instances UDP packets are blocked by Internet Service Providers, and when this occurs the GUI can successfully connect to the FM-531HD 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 FM-531HD and its GUI to using 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 FM-531HD over Ethernet. When a GUI TCP connection session is initiated, the FM-531HD accepts the connection and then asks the remote Pro GUI for a password.

If the remote GUI cannot provide the correct password, then a dialog box will pop up asking the user to enter the correct password.

Until the login password can be verified the FM-531HD will not send its current settings to the remote GUI nor will it accept any changes from the connecting GUI.

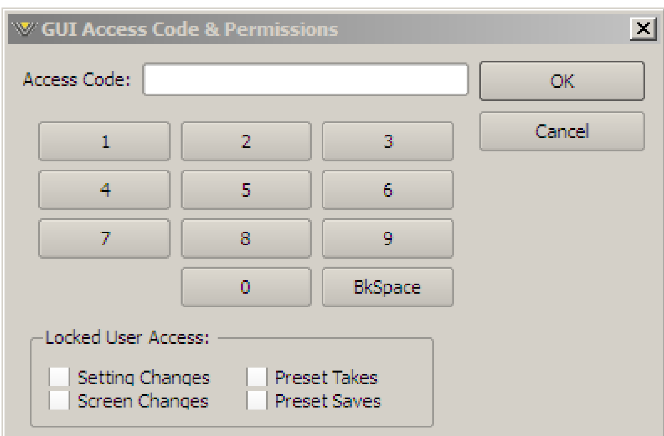
The dialog box show above also allows a user to change the login password if one has been set. The current password will need to be known beforehand, and it will need to be entered twice for verification. The password can be any combination of characters, numbers, or symbols. If the *Remember This Password* 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 FM-531HD and Remote GUI are handshaking on the password string that is being sent.

Pro GUI Access

The 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 FM-531HD. 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.

Unlocking the FM-531HD Pro GUI When the Passcode Has Been Lost

There are two ways of gaining access to the device which has been locked by the GUI:

1. Re-install the GUI

- Re-install the PC Remote GUI but this time into a NEW folder on your computer.
- Start the GUI from the new location, recreate the previous FM531HD device in the GUI and assign it the same name and IP address.
- Select the new device and click the GUI’s online button.

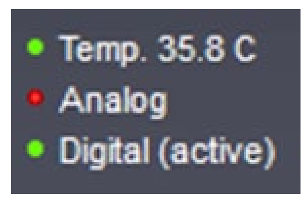
2. Remove the "Locked" properties from the current GUI installation.

- Make sure the GUI is not running.
- Navigate the directory structure of the host PC to find the folder where the GUI has installed its support and configuration settings. By default this will be:

C:\Documents and Settings\(logged on user)\My Documents\Wheatstone\FM531HD\maps

- Inside the MAPS folder are one or more .cfg (configuration) files named for the devices that the GUI has previously been configured for.
- Locate the .cfg file with the name of the device you are unable to access.
- Delete this .cfg file.
- Restart the GUI.
- Recreate the device.
- Highlight the new device and click *Select*. The *Devices* box will close.
- Click on the *Online/Offline* button left of the *Status* window at the top of the Pro GUI.

Status Indicators



Temp – When the *Temp* indicator is green, the operating temperature inside the FM-531HD is satisfactory. When it is red, the temperature is too high. The displayed value is the current operating temperature in degrees Celsius of the FM-531HD motherboard. Normal temperatures should be in the range of 25ºC to 50ºC, and will vary somewhat depending on ambient temperatures near the FM-531HD.

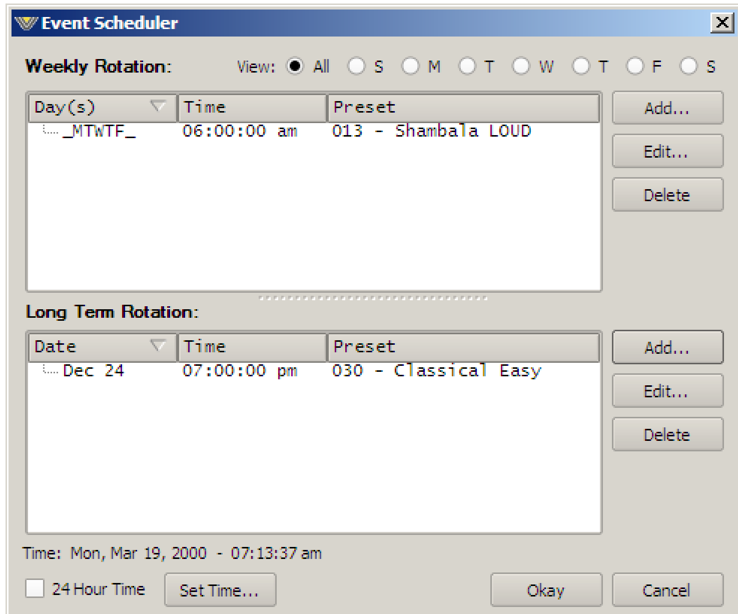
Analog – When green, this indicator shows that audio is present on the analog line inputs and that its level is above -24dBu. When the indicator is red it means that audio is missing or is too low to be detected.

Digital – When green, this indicator shows that valid data is present on the AES3 digital input, and that it 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 audio is present but is too low.

Scheduler

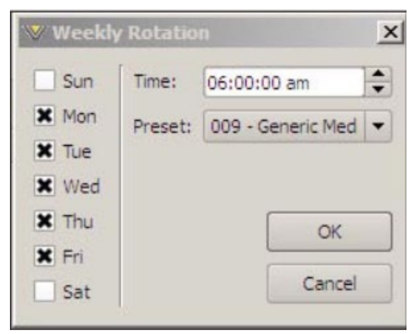
The FM-531HD contains a comprehensive scheduling utility that allows presets to be automatically changed on desired dates and times. There is also a *Long Term Rotation* utility to allow one time preset changes to be scheduled for any time in the future. To create, edit, or view entries in the *Event 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 on the right.
2. On the left column, select the days that the preset should be selected. Any combination of check boxes 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*.

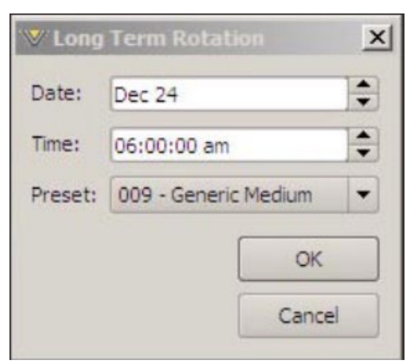


As many preset change events as desired may be programmed in the *Event Scheduler*.

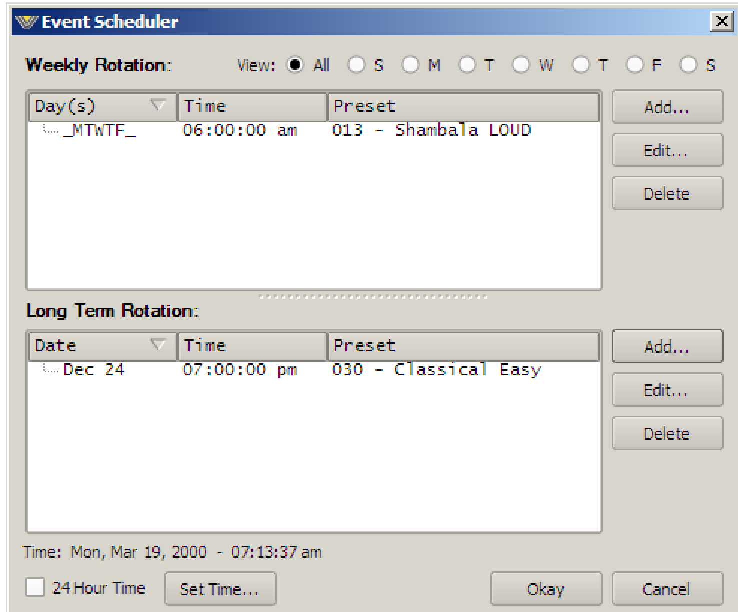
If a one-time preset change is required, such as might be necessary for a special holiday or other event, the *Long Term Rotation* routine should be used.

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 shown on the right:



Deleting a Scheduled Event

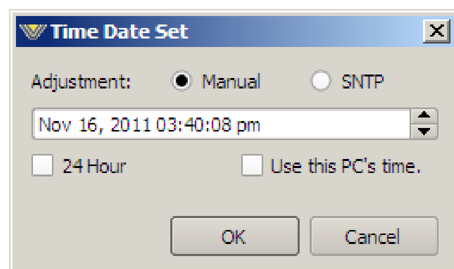


Deleting a scheduled preset change is as simple as adding one. Open the *Event Scheduler* window by navigating to the System menu and clicking *Scheduler*. The *Event Scheduler* window will open.

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

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

System Time



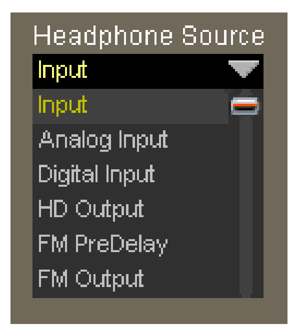
Allows the system time of the FM-531HD onboard computer to be set or changed. The FM-531HD system time can be synchronized to external time servers (SNTP, or Simple Network Time Protocol) if desired.

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.

See "Setting System Time" in chapter 2 for additional information on time servers.

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 FM-531HD headphone routing selector allows you to monitor an un-selected 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 FM-531HD 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.

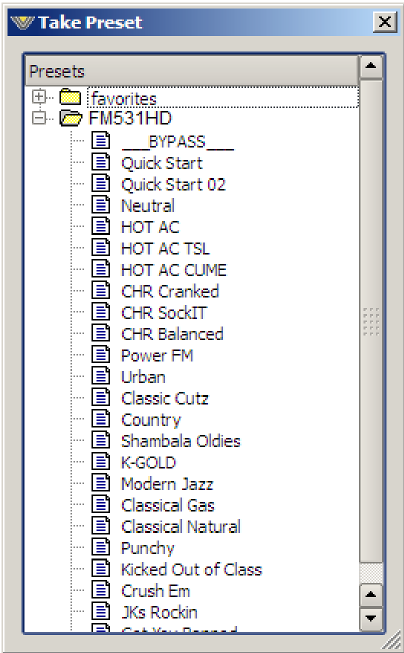
To adjust the headphone volume slide the fader right to increase volume and left to decrease it.

Note: The headphone outputs of the FM-531HD 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 FM-531HD, which lists all presets stored on the hardware.

Double-clicking on a preset brings it immediately into use in the FM-531HD. 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 FM-531HD 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. Alternatively, by nudging the preset number (Preset #) up or down, it is possible to save the present settings in place of an existing preset's settings. Needless to say, this erases, as in loses, as in destroys, the overwritten preset's contents.

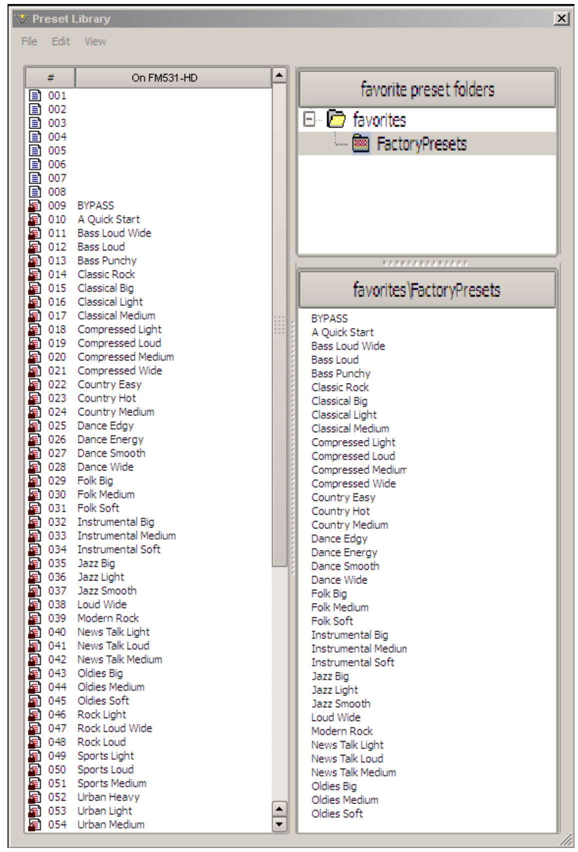
Factory presets are protected from being inadvertently overwritten.

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 the FM-531HD.

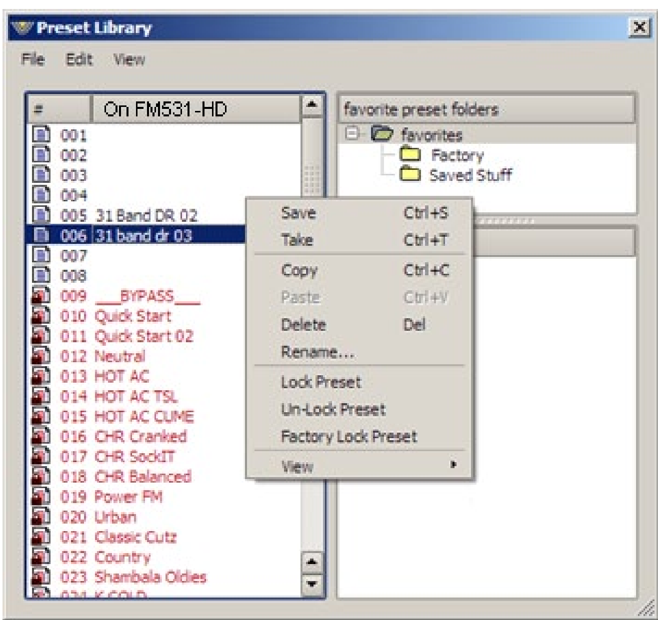
The upper right panel shows the preset folders on the PC hosting the GUI, and the lower panel shows the presets stored in the folder chosen in the above panel.

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



Locking Presets

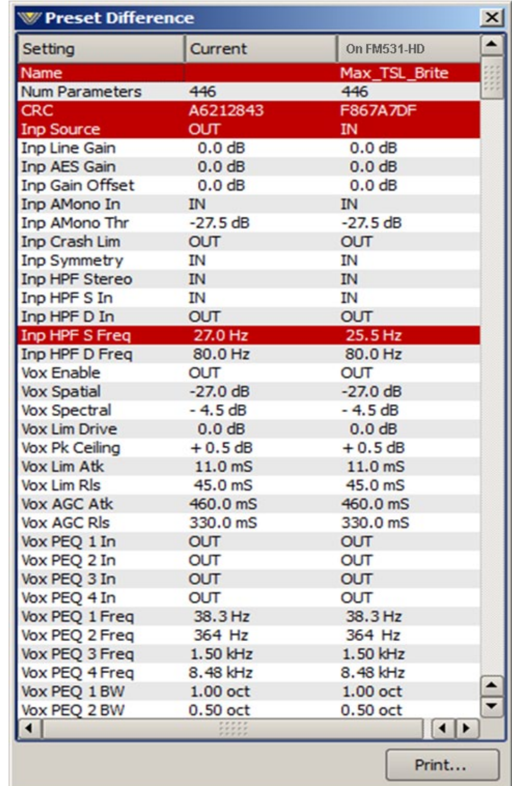
Presets stored on the FM-531HD may be locked by the user to prevent inadvertent overwriting, renaming, or deletion. This is accomplished by opening the *Preset Library* by clicking on the *Library* button in the GUI. Once the list of presets is open, the ones actually stored within the processor hardware 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 FM-531HD with a special feature called Preset Difference.

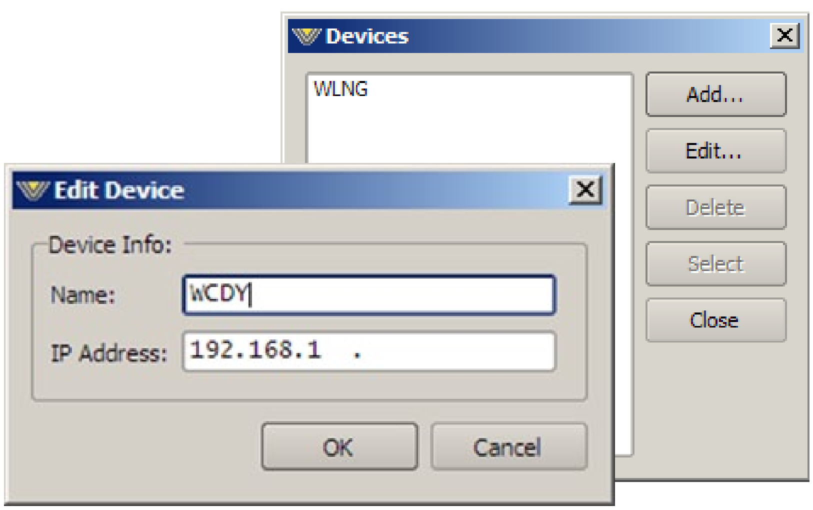
When the *Library* tab is opened, highlight any preset you'd like to compare with the current preset running on the processor. Then right click and select *View* and *Diff* from the sub-menu. You will see the window on the right open.

Any parameter that is listed with a RED background indicates a parameter that is different 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.

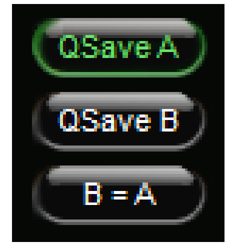
Devices

Clicking on the *Devices* button opens a list of the FM-531HD devices known to the GUI (if any). Highlighting the desired FM-531HD's name and hitting *Select* causes the GUI to attempt to connect to it if you are Online.

In the event no FM-531HD'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 FM-531HD 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 FM-531HD 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 FM-531HD 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 FM-531HD 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 FM-531HD 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 FM-531HD Pro GUI screen (in line with the *Wheatstone FM-531HD* 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 FM-531HD Pro GUI is capable of controlling multiple FM-531HD 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 FM-531HD.

Offline – The GUI and FM-531HD are disconnected.

Trying – The GUI is attempting to find the FM-531HD on the LAN and connect to it.

Pending – The GUI has found the FM-531HD and is trying to complete the connection.

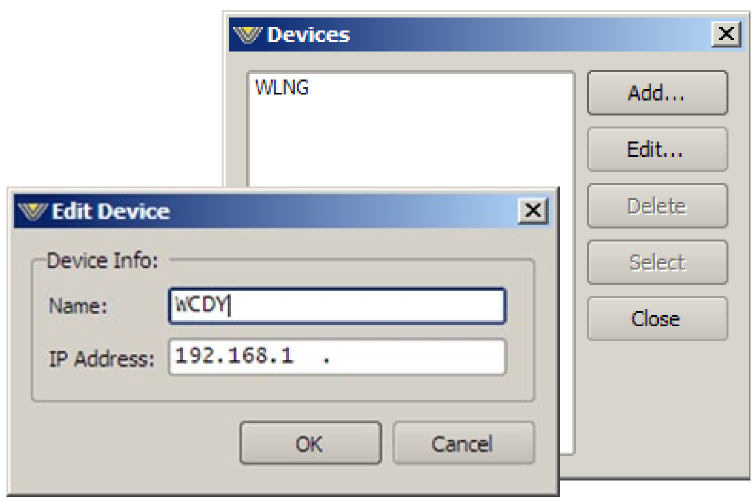
An adjacent and illuminated button toggles between *Online* and *Offline*.

Device

The name of the FM-531HD 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 FM-531HD devices that the GUI knows about is shown in the box. Highlighting the desired FM-531HD'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 FM-531HD 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.

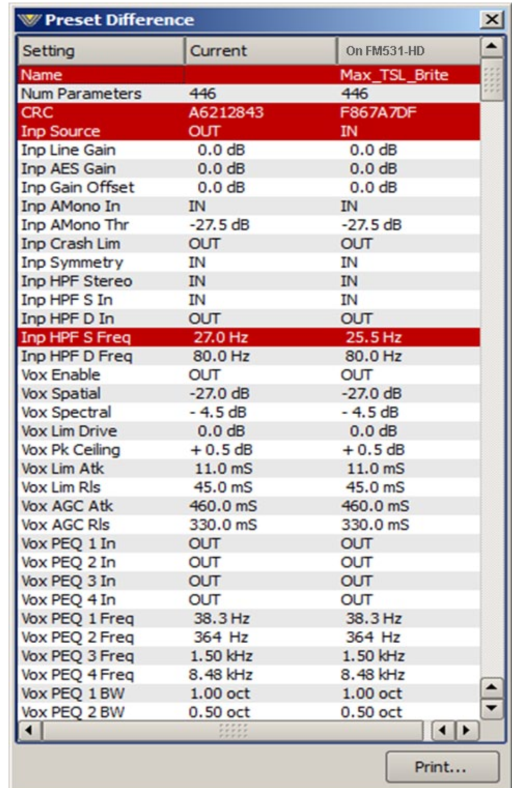
When the preset name is displayed in GREEN text the currently running preset reflects exactly what has been last saved to that preset and that no changes have been made.

When the preset name is showing in RED text, the current preset has been modified and is not saved on the FM-531HD 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 3-52), it is possible to visually compare two different presets on the FM-531HD 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 FM-531HD settings and the recalled preset.

Any parameter that is listed with a RED background indicates a parameter that is different 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 FM531-HD
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 the FM-531HD 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 FM-531HD.

Changes can be made to presets and configurations without affecting the sound of the processing occurring within the FM-531HD.

Importantly, any adjustments made on the GUI *Offline* will not take effect until the preset changes have been saved, the FM-531HD is *Online* to the GUI again, and the preset is made active by selecting it.

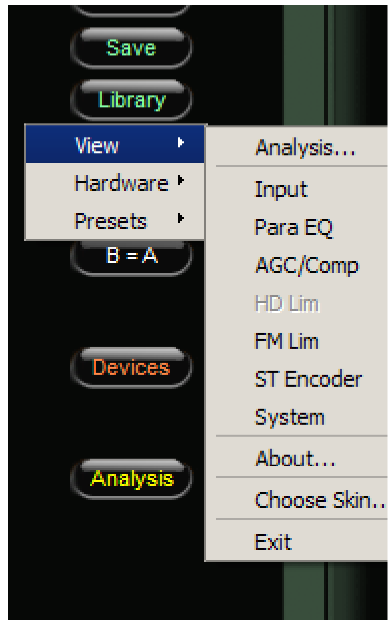
Invoking a preset – by double-clicking a preset from the box brought up by clicking the *Presets* button – does one of two things, depending on whether the Pro GUI is in *Online* or *Offline* mode:

- *Online* – the double-clicked preset is immediately sent to the FM-531HD 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 FM-531HD's audio.

Accessing Menu Options

Right clicking anywhere on the FM-531HD 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 FM-531HD 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 FM-531HD Control Panel.

Analysis... – The FM-531HD Pro GUI offers the same analysis options as the Guru GUI. You can access these analysis features by clicking *Analysis*.

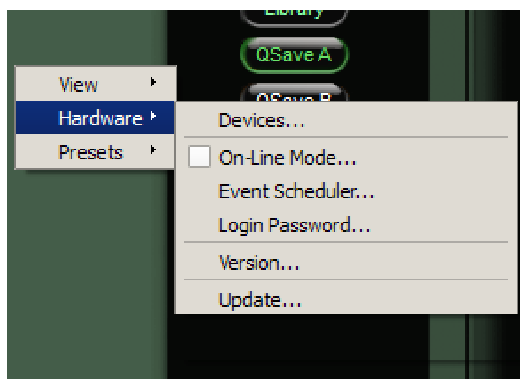
The next set of options in this window (*Input – System*) correspond to the tabs along the top of the GUI. This is just another place to access those options.

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 FM-531HD Control Panel. Please note that most of the Hardware-related items require you to be connected (Online) to the FM-531HD 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 Wheatstone 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 FM-531HD 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 FM-531HD hardware.

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

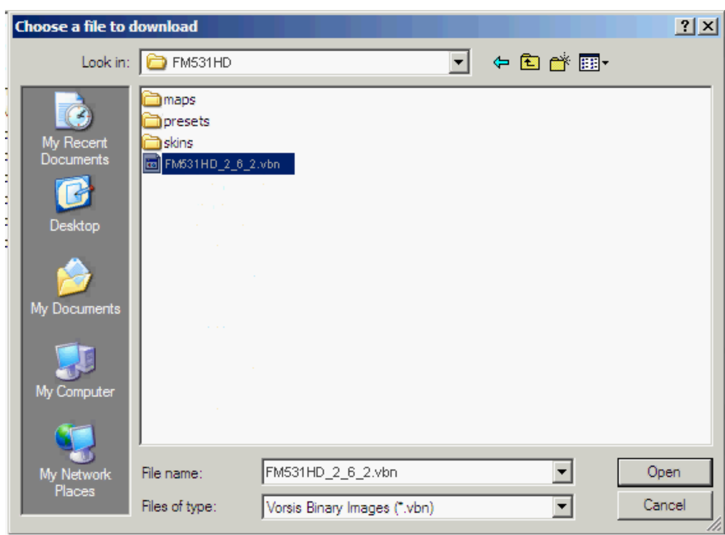
New or updated FM-531HD software may be released at any time to implement new features, change specifications, or correct known bugs. The Hardware Update choice will open a separate dialog box which will prompt the user for the correct file type to be uploaded to the FM-531HD. Upon completion of the update you will be asked to restart the FM-531HD (cycle its power).

FM-531HD Hardware Update

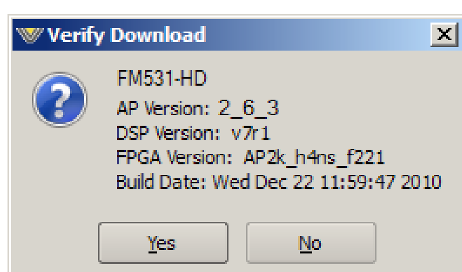
The FM-531HD is extremely easy to update when new firmware becomes available from Wheatstone. The file involved in updating the FM-531HD is called a *VBN* (Virtual Binary Nugget). This file contains motherboard application code, FPGA images, and DSP code.

Before performing updates to the FM-531HD make sure an auxiliary processor is available for when FM-531HD needs to be rebooted after the upgrade as there will be 15-20 seconds of silence until FM-531HD 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 FM-531HD, 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. The screen at right will appear.

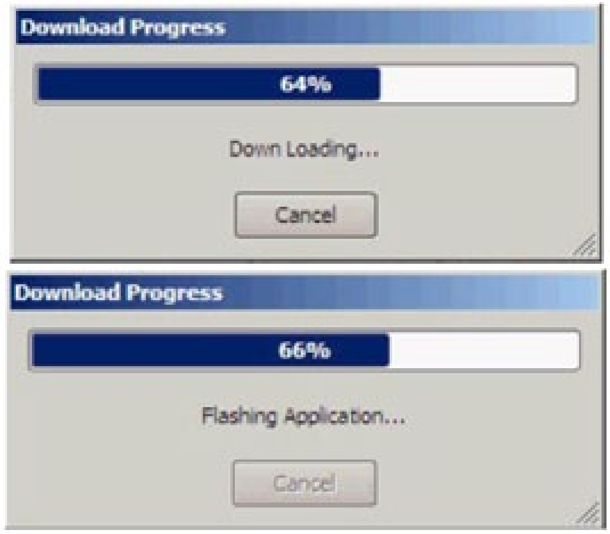


Navigate to the location of the VBN file that was downloaded from the Wheatstone website or if a new GUI version was just installed, the VBN that was unpacked during its installation. Verify that the version number is correct in that it matches the version number listed in the documentation about the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

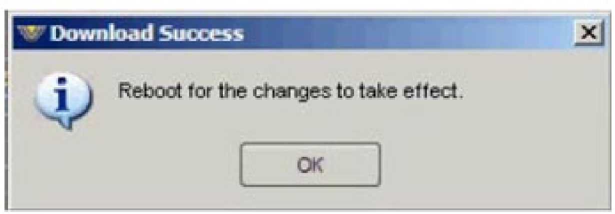


Recheck that the versions match the documentation! Then click *Yes*.

Progress boxes will pop up as the download progress occurs:

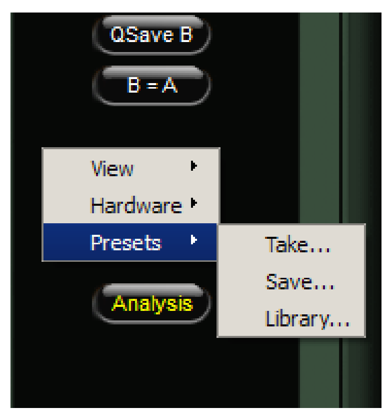


When the system has finished downloading, you will get a prompt to reboot the FM-531HD.



After the FM-531HD reboots the new VBN code will be running on the motherboard.

Presets Menu Items



The *Presets* menu tree may be accessed by right clicking anywhere on the main FM-531HD 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 A

Contents

General Specifications

System Level

Headroom:	20dB
Nominal Operating Level:	-20dBFS

Analog Line Input

Type:	Electronic Differential
Impedance:	> 10Kohm (bridging)
Optimum Source Impedance:	< 1Kohm

Digital Line Input

Type:	AES/EBU (AES3) or Wheatstone AoIP
AES Data Amplitude:	Per AES3-2003 assuming minimum allowable output signal amplitude of 2V and minimum allowable input signal amplitude of 200mV

Input Gain Adjustment Ranges

Gain Adjustment:	+12dB to -36dB
Gain Adjustment Resolution.:	0.5dB
Gain Calibration:	A gain control setting of 0.0 aligns an external 0dBFS signal with the FM-531HD's 0dBFS internal reference

Input Failover

Type:	Automatic
Analog Fail Cause:	Audio level below -48dBFS (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)

Audio Level Balance

Type:	Common to Analog or Digital inputs
Analog/Digital L/R Balance Range:	+/-12dB
Control Resolution:	0.5dB

Voice Symmetry Phase Rotation

Operating Modes:	In/Out
Filter Type:	4^{th} Order Allpass

High-Pass Filter

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

Equalizer

Frequency:	20Hz – 20kHz
Bandwidth:	0.2 – 3.0 octaves
Boost / Cut:	+/- 14.0dB
Options:	Bypass All bands

Multiband AGC/Compressor with SST

AGC Drive:	-12.0dB to +6.0dB in 0.5dB steps
AGC Backoff:	0dB to -6dB
AGC Attack:	50mS – 500ms
AGC Release:	7 seconds to 1 second (7000 – 1000ms)
AGC Band Coupling:	0dB to -6.0dB in 0.5dB steps mid to high 0dB to -12.0dB in 0.5dB steps mid to low
Compressor Attack:	3mS – 100mS
Compressor Release:	50mS – 1000mS (1 second)
AGC Mixer:	6dB Boost/Cut at 50Hz, 160Hz, 600Hz, 2kHz, 8kHz
Stereo Enhance:	6dB Boost/Cut at 50Hz, 160Hz, 600Hz, 2kHz, 8kHz
Gated Modes:	Ooze/Hold
AGC Gate Threshold:	-20.0dB to -79dBFS, plus OFF
SST Drive:	+30.0dB to +40.0dB in 0.5dB steps
SST Ramp Rate:	2 to 7.0 seconds
SST Gate Threshold:	-35.0dB to -60.0dB in 0.5dB steps
SST Window Size:	3dB to 12.0dB
SST Mode:	Linear or Adaptive

31 Band HD Limiter

31 Band Drive:	80 – 100%
VBMS Drive:	-6.0dB to +6.0dB
VBMS Style:	Off/ Soft/ Hard
VBMS Output:	-6.0dB to +6.0dB
VBMS Frequency:	80 - 300Hz

Multiband Threshold:	+/-6dB relative to Multiband Drive setting
Multiband Attack Time:	10ms to 100ms
Multiband Release Time:	3.0ms to 100ms
Look ahead Limiter Drive:	+/- 6.0dB
Look ahead Limiter Threshold:	+/- 6.0dB
Look ahead Limiter Attack Time:	0.2ms – 30ms
Look ahead Limiter Release Time:	33ms – 330ms
Look ahead Limiter Delayed Release:	100ms – 1000ms (1Sec)
Digital Output:	-79.95dBFS to 0.0dBFS, or OFF
Phase and Polarity:	Left and Right channels may be phase-reversed or swapped

Multiband FM Limiter

31 Band Drive:	80 – 100%
Main Clip Drive:	+/- 6dB
VBMS Drive:	+/- 6dB
VBMS Style:	Off/Soft/Hard
VBMS Output:	-6dB to +6dB
VBMS Frequency:	60Hz - 300Hz
Multiband Threshold:	+/-6dB relative to Drive
Multiband Attack:	10ms to 100ms
Multiband Release:	3ms – 100ms
Pre-Emphasis:	Flat/50μS/75μS
FM Diversity Delay:	OFF to 10 seconds
Diversity Delay Coarse Resolution:	50ms
Diversity Delay Fine Resolution:	100μS
FM Digital Output:	-79.95dB to +0.0dB, or OFF
FM Digital Output Types:	AES3 Digital / baseband192 digital multiplex;
FM AES3 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.

MPX Processor Styles:	Composite Clipper or Look Ahead Limiter
MPX Process Drive:	+/-6dB
Multipath Controller:	10% to 100% in 10% steps, plus Off
Stereo Pilot Injection:	0-20%, 0.1% steps
Stereo Pilot Phase:	+/- 22.5 degrees referenced to 38kHz
SCA 1 & 2 Input:	Analog, 10Kohm input impedance

Maximum SCA Input Level:	+24dBu
SCA 1 & 2 Gain Control:	-79.95dB to +10dB; OFF
TX 1 & 2 Level:	-79.95dB to +6.0dB in 0.05dB steps corresponding to baseband192 levels of -85.5dBFS to 0.0dBFS respectively; or OFF
Analog Output Type:	Analog L/R Analog L/R De-Emphasized Unbalanced and Balanced Multiplex
Test Oscillator:	
Frequencies:	50Hz to 80kHz plus 31.25kHz Bessel
Output Level:	0 to 100% in 1.0% steps

System

Distortion:	No greater than 0.01% THD 20Hz – 29kHz system bypass
Signal to Noise Ratio:	100dB system bypass
Dynamic Range:	>144dB
Stereo Separation:	>60dB (FM) 100dB (HD)
Crosstalk:	>80dB, 20Hz – 15kHz (FM) 20Hz – 20kHz (HD)
Power Requirements:	100-250 VAC (auto sensing) 50/60Hz
Power Connector:	EMI suppressed IEC male
Operating Temperature:	0 to 50 degrees C (32 to 122 degrees F) Alarm sensor activates at 50 degrees C

All specifications subject to improvement or change without notice.

Presets

An advanced preset management system allows the creation, storing, and recall of 80 presets within the FM-531HD itself and an unlimited number on a host PC. All signal-processing parameters are contained in these presets, allowing the preservation of the complete processing environment. Optionally, Input and Output settings may be stored and recalled with presets.

Preset Encryption

Presets are encrypted using the FM-531HD internal 32 bit serial number as the encryption key.

Onboard Real-time clock

A highly accurate internal real-time clock allows the automatic recall of presets at pre-determined times. This clock may be set to free-run or be synchronized to an external on- or off-site SNTP server.

Day-Parting and Long-term Scheduling

A weekly preset scheduling establishes a weekly cycle of preset changes, allowing for automatic programmed day-by-day exclusions or additions.

Additionally, a separate long-term 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 slaved to the first eight 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 with the GPI pin being the most positive voltage. Please see manual text for the required current limiting resistor for voltages above 3.3VDC.

Ethernet Interface

The FM-531HD contains an auto-sensing 10/100BaseT Ethernet switch with one port available on the outside of the unit for field use. Because the switch is auto-sensing it is not necessary to utilize a crossover cable when connecting a PC directly to the FM-531HD.

Software Remote Control

Supplied Windows® FM-531HD GUI software affords control of all system and processing parameters via a 10/100BaseT Ethernet interface.

Appendix B

Contents

FM-531HD GPI Interface

The FM-531HD is equipped with a GPI (General Purpose Input) control interface – an eight input optoisolated interface for selecting presets in the first eight memory slots.

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 the FM-531HD 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.

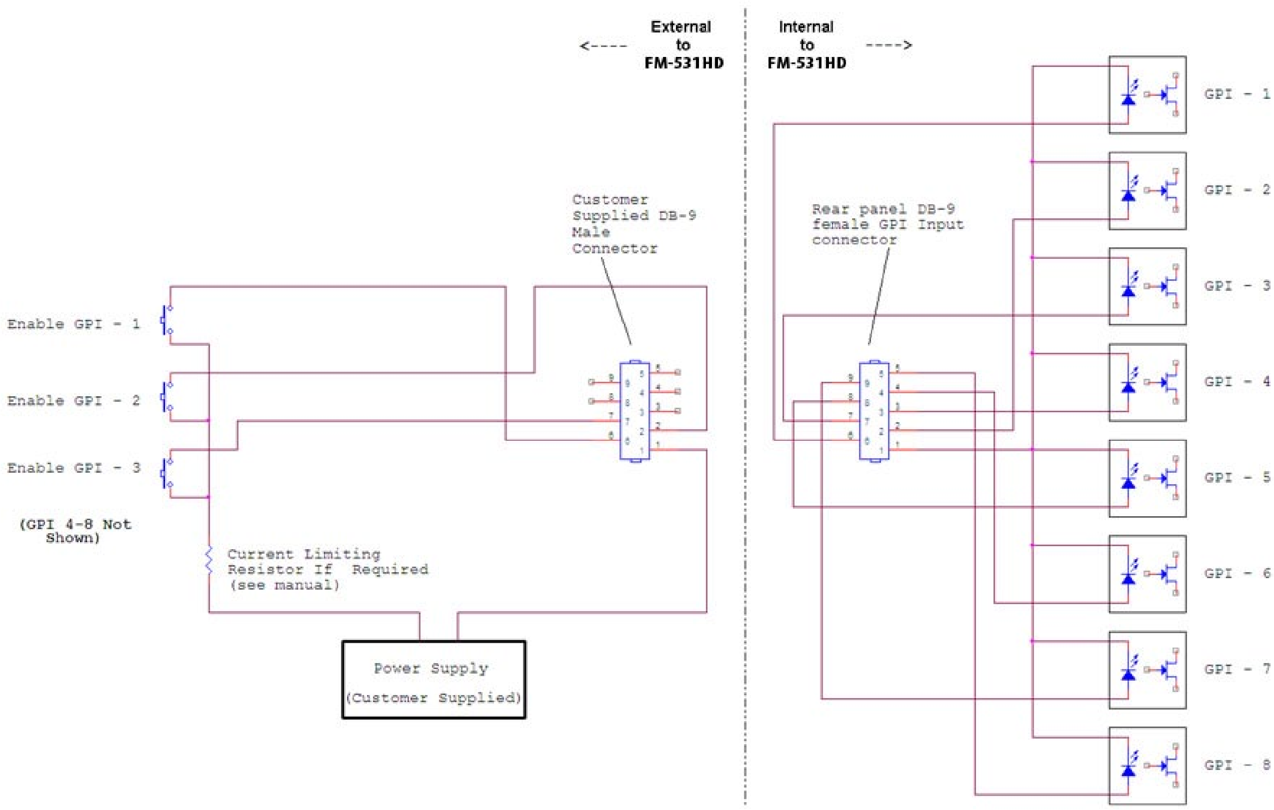
Each of the GPI inputs is current limited by a 475 ohm internal resistance. The LED device inside the optoisolators is rated at a maximum forward current of 50mA and a normal operating current of 5mA.

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

Appendix C

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).

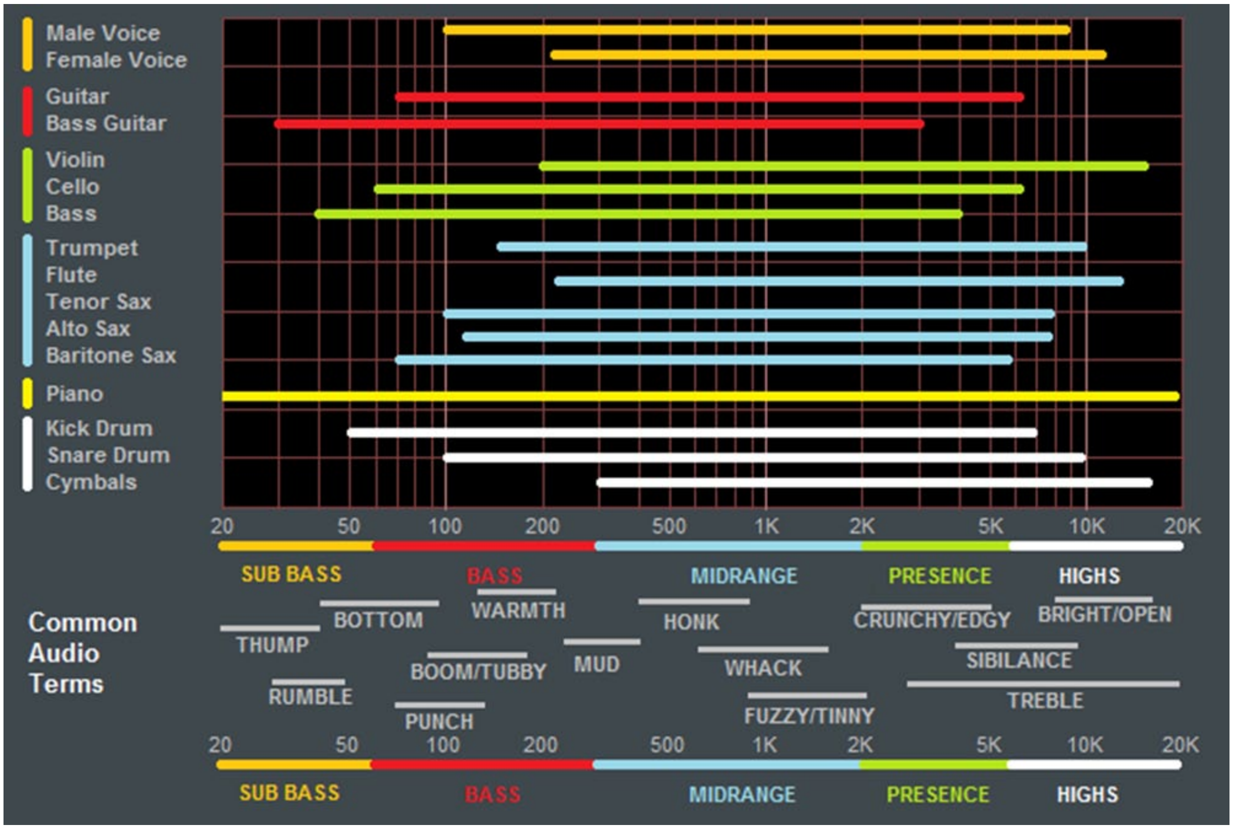
The *language* of audio processing more often than not needs the benefit of an interpreter. Some oft-used terms can easily have multiple meanings and be confused with others. Some of the terms we use simply don't describe very well what is actually happening to the audio. For instance, one listener might 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."

Please note that some of the controls mentioned on the following pages may not be available on the FM-531HD. The discussion is generic to a wide variety of Wheatstone processor models.



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 FM-531HD 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." .

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 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 the FM-531HD is much more forgiving in this regard and most users have an easier time being loud with the FM-531HD 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 us a call at 252-638-7000. We know our processors inside and out and can make them sit up and dance in any market or format.

Appendix D

Contents

Using The Factory Lock Utility

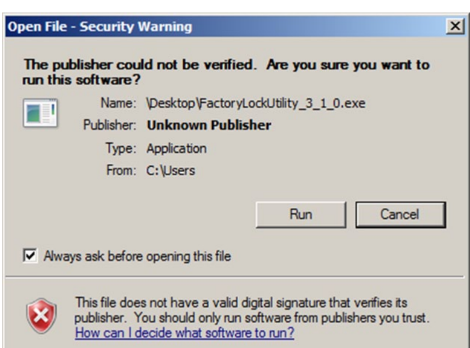
Background

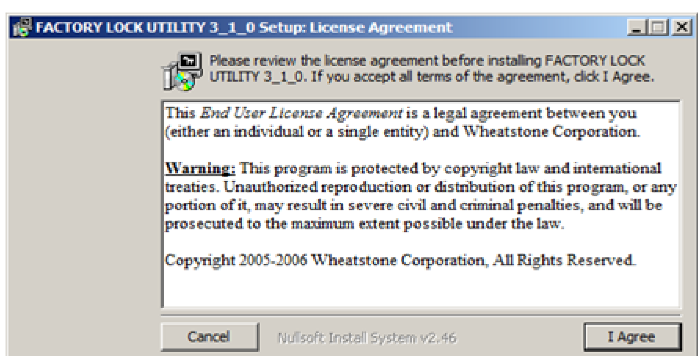
The Factory Lock utility tool was originally designed for audio processing consultants who use and recommend our processors to their clients. The utility allows them to unlock our factory presets, create and "factory lock" their own presets, or even remove all but their own custom presets from an Audio 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 made the utility also available to customers.

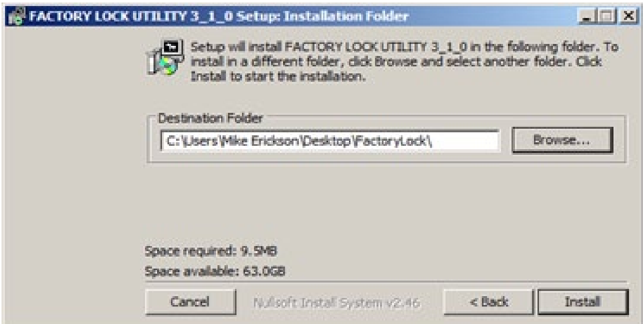
Installation

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.

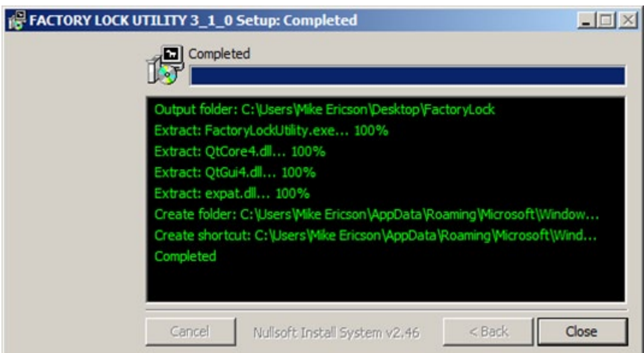




Next up is the License Agreement prompt. Please select the *I Agree* option to begin the installation process.



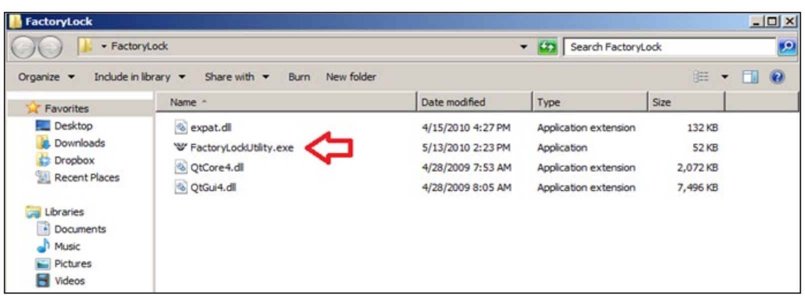
The next prompt will ask where you would like to install the utility on your computer. Please either accept the installation default or navigate to the location where you would prefer it to be installed by using the Browse button to open the Browse for Folder Windows dialog.



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.

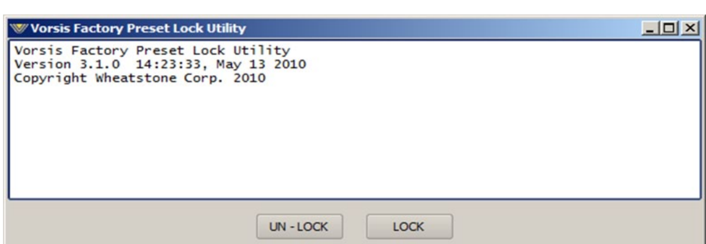
Using The Utility

STOP! ***ALL WHEATSTONE PROCESSING GUI'S MUST BE CLOSED BEFORE RUNNING THIS UTILITY!***

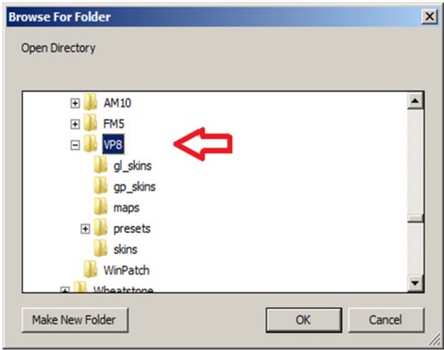


There are four files installed into the *Factory Lock* utility's folder. The only one you should be concerned with is the actual *FactoryLockUtility.exe* file.

Double click on this file to start the utility and the Lock Utility choice menu will appear.



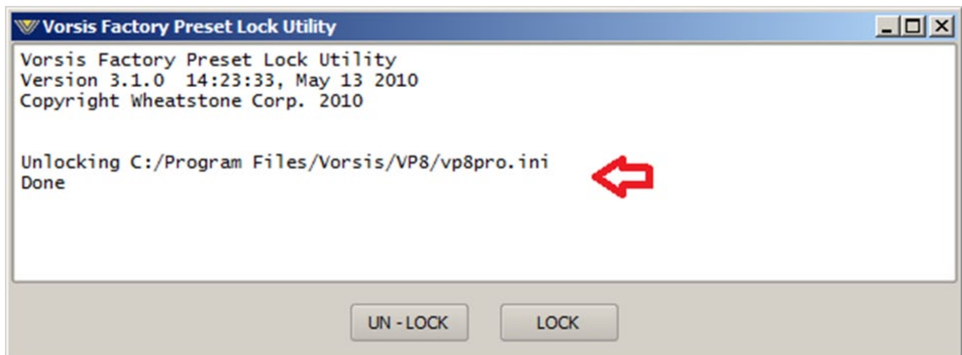
In order to grant your GUI the Factory Lock privileges click on the *UN–LOCK* button. A new navigation box will open and if the Wheatstone processor GUI has been installed in its default location the path you want to point the utility to is: "...Program Files\Wheatstone\Processor Name".



Once you have reached the folder with the name of the processor, STOP!

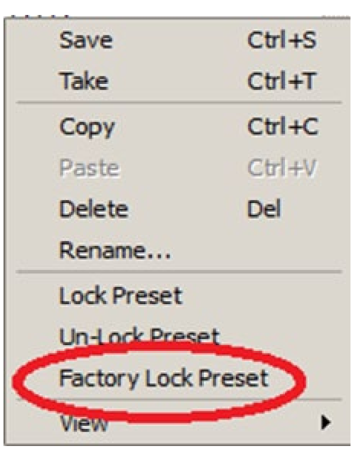
Click *OK* to proceed.

The utility's progress window should now say "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.