Omnia.9/XE

Streaming & Encoding Software

Installation & User's Guide

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Omnia ®
THE TELOS ALLIANCE

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Introducing Omnia.9/XE

Omnia.9/XE is a software-only version of the processing and streaming encoder portions of the "Omnia.9 by Leif Claesson" hardware broadcast processor, including Undo - the revolutionary de-clipper and multiband expander that removes distortion and restores dynamics to music that has been clipped and over-compressed during the mastering process.

Broadcasters have always taken their on-air processing seriously, but too often streaming audio fails to get the same careful consideration. Sometimes the audio destined to the streaming encoder isn't processed at all (leading to listener complaints about wild swings in audio levels). Sometimes it is processed with a cheap wideband hardware processor rescued from the pile of de-commissioned equipment (resulting in low-quality sound with audible processing artifacts). Now and again, it may even be fed by the same processor as the FM signal path (which results in high levels of distortion and a correspondingly high tune out rate).

Omnia.9/XE is designed for terrestrial or streaming-only broadcasters who understand the importance of providing the highest quality audio possible to their listeners on a wide variety of devices and with varying amounts of available bandwidth.

This user guide is designed not only to take you step by step through the setup and installation process and through the various menus and controls but also to serve as a resource to make educated, thoughtful decisions about encoding in general and do help you understand what's really happening to your audio as it goes through the encoding process.

Chapter 1
General Processing Advice

General Processing Advice

Know Your Goals

The first step to successful processing – and this applies to ANY processor, not just Omnia.9/XE - is defining your goals. Whatever those goals are, Omnia.9/XE is capable of getting you there.

You may wish to maintain as much quality as possible while increasing loudness somewhat over your existing processor. Or you might want to create very consistent source-to-source spectral balance and develop a "signature sound" for your station.. In any case, it is best to start with a goal in mind and work toward it carefully and deliberately.

Tweaking and Fiddling

One big temptation when installing a new processor is fiddling with every available control, and there is no shortage of those in Omnia.9/XE! There's really no better way to find out what each control does, but experimentation and exploration are probably best done "on the bench" as it were before you put your new acquisition on the air. That way, you're free to tweak away without having to worry about jeopardizing your on-air sound or being unnecessarily timid about making adjustments because you don't know what affect they will have.

Another suggestion for getting to know your way around Omnia.9/XE is to find a factory preset that you like and then go exploring in the various menus to see how that sound was achieved. For example, let's say you find a preset with a smooth, open sound which matches your general processing goal. As you explore, you might find some controls set as you would expect to deliver that sound, but you may also find some unexpected settings that make more creative use of Omnia.9/XE's controls. There is much to be learned by studying.

Once you are comfortable with navigating through Omnia.9/XE's menus and have a good understanding of how its controls operate, you're ready to take it for a live test drive.

Choosing a Preset

Regardless of whether you end up using a supplied preset "as-is" or make adjustments to customize your sound, you must start with one of the factory presets. We recommend you find one that most closely matches your goals and then make any modifications from there as needed.

Most processors use radio format names to try and define their sound but then go on to explain there's no reason a rock station shouldn't try out the "Country" preset – which is very true. As a rule, Omnia.9 takes a deliberately different approach and most of its presets don't give you any hint as to who should use them (or how) in the hopes that you'll "step out of your box" a bit and explore.

If your Country station sounds great on the air with settings that might mimic an "Urban" preset - with a slamming low end and a nice open mid-range – the last thing we want to do is make you second guess yourself because the preset name doesn't match.

Making and Saving Changes

Human nature is fickle; the human ear adapts and tires quickly; and Omnia.9/XE is powerful! With those things in mind, here are some recommended "Do's" and "Don'ts" when making and saving changes to presets:

- DO resist the temptation to constantly fiddle with every control right after you put Omnia.9/XE on the air.

- DON'T make hasty, radical changes.

- DON'T make adjustments to too many parameters at once – that makes it difficult to determine which of the adjustments is actually responsible for the changes (for better or worse) you're hearing on the air.

- DO look to the Band Mix and Parametric EQ sections first to achieve the spectral balance (aka "signature sound") that you're looking for. These are powerful adjustments, and a little EQ can go a long way. This is generally "safer" than making adjustments to the target, attack, and release rate controls as it is sometimes difficult to know how the different compression stages interact with one another on all material.

- DO take breaks when adjusting your processing. Ears tire quickly, and if you stay at it too long, you're almost sure to make changes influenced by fatigue.

- DON'T turn your monitor speakers up too loud when making adjustments. High listening levels cause ears to tire even more quickly and mask both gain riding artifacts and distortion. Most listeners play the radio in the background, and problems tend to be more audible at comfortable listening levels.

- DO make small adjustments, particularly to critical controls like Limiter thresholds.

- DO take the time to calibrate a set of high-quality reference monitors (a process described in detail in this manual) so that any changes you make aren't skewed by colorations of the speakers or room.

- DON'T rush the process. Use the "sleep-on-it" method when you've reached a point where you are mostly satisfied with the sound, and then re-evaluate it the next day. If it still sounds good, STOP. If it doesn't, make a few adjustments and walk away for another day.

- DO use the "Save As" method of naming and saving your custom presets rather than over-writing them. This will allow you to return to any point in your adjustments if you get too far "off track" and keep you from having to remember (or guess) what changes you've made along the way.

Many Omnia.9/XE users have asked if they can make their own custom presets "from scratch." All custom presets start life as a factory preset, but the most straightforward, neutral-sounding factory preset is "Reference Settings," and as such provides the closest thing to a "blank canvas" possible for building your own preset from the ground up.
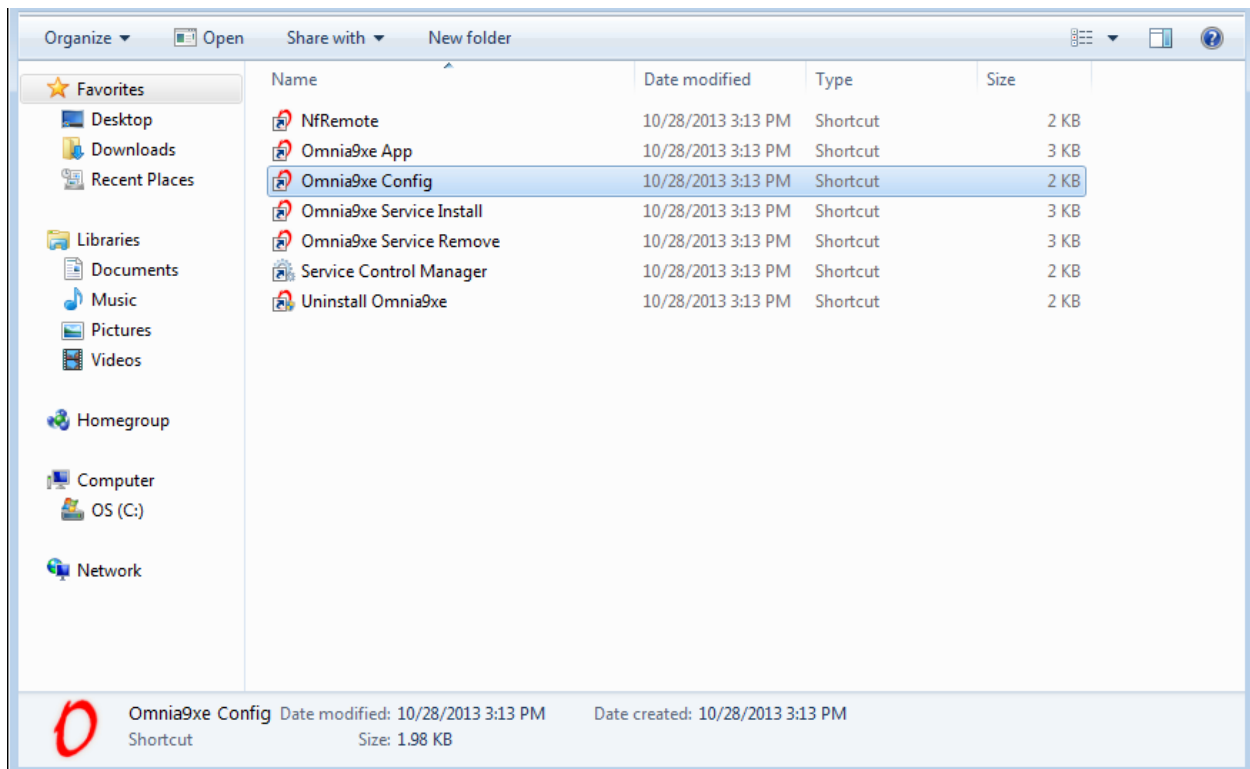
Chapter 2
Installing the Omnia.9/XE Software

Installing the Omnia.9/XE Software

The Installer File

The installer is an executable (.exe) file that by default will install a shortcut to the Omnia.9xe start menu group on your Windows desktop. To begin installation, double-click on the "omnia9xe_installer_.exe" file. This shortcut will open an Windows Explorer window containing 7 individual shortcuts:

- NfRemote
- Omnia9xe App
- Omnia9xe Config
- Omnia9xe Service Install
- Omnia9xe Service Remove
- Service Control Manager
- Uninstall Omnia9xe

To upgrade, simply install the newer version on top of the old one. Uninstalling first serves no purpose as settings are kept in C:/programdata/Omnia9xe and are **not** removed by the uninstaller.

Chapter 3

Initial Setup and Configuration

Initial Setup and Configuration

Entering Your License Key

Begin the initial configuration by opening the "Omnia9xe Config" shortcut, selecting "License", selecting an available Slot #, entering your license key, and then selecting "Activate". The slot in which you entered the key will indicate the key number and the number of licenses in green. Please note that one license key can only be activated on one machine.

Using a Demo License Key provides full program functionality but inserts a voice-over periodically announcing that you are using a demo version. Stations running in demo mode are clearly marked as such in the user interface.
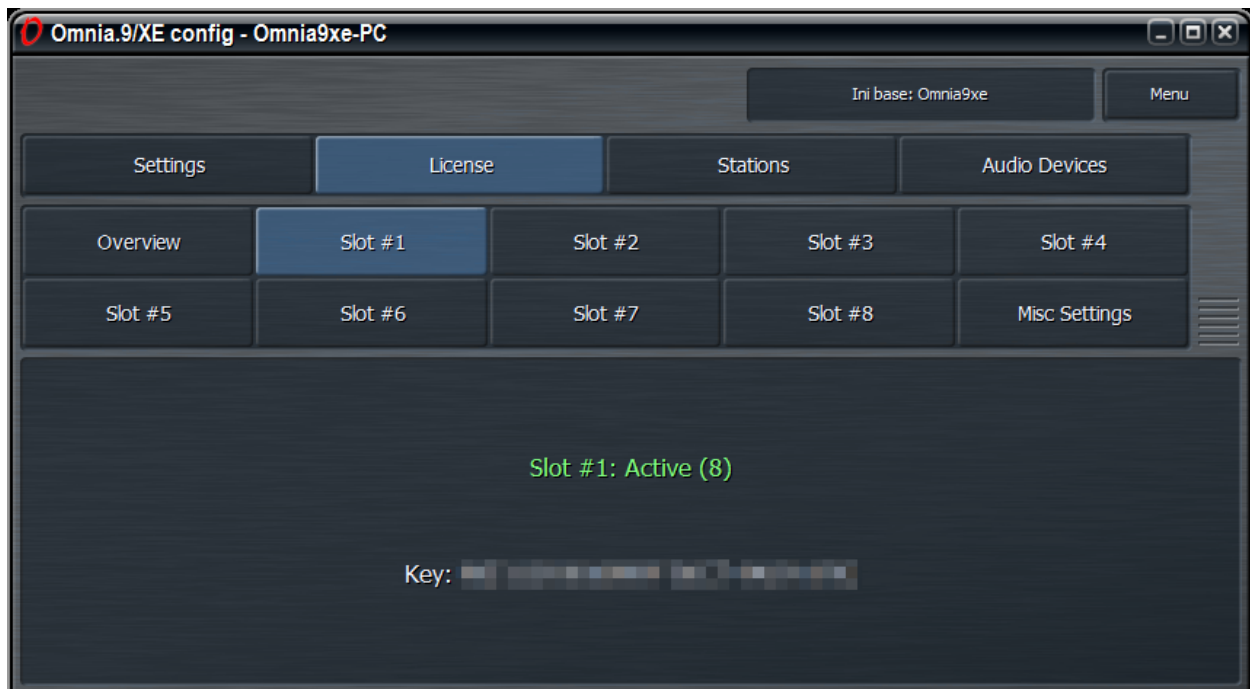
Setting the Number of Stations and Devices

A "Station" refers to a unique individual program source. A "Device" refers an audio interface. Multi-channel sound cards can have more than one device (for example Record 1-2, Record 3-4).

Begin by selecting the "Settings" and then "Base Configuration".
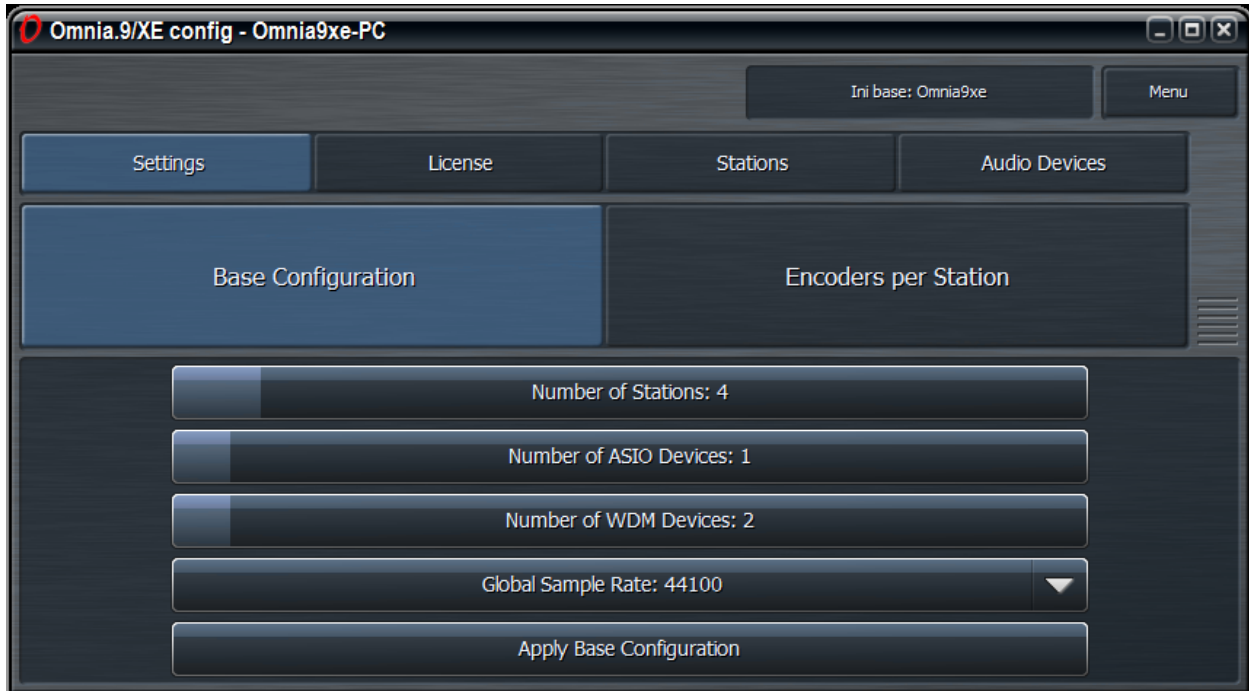
- Choose the number of stations by sliding the "Number of Stations" control. If you are using a mouse, hold down the left mouse button while dragging the slider. On a touch-screen tablet or PC, press and hold the slider while dragging it. Remember that any stations in excess of the number included with your license key(s) will run in demo mode and include a voice-over.

- Set the number of ASIO devices (Audio Streaming Input/Output) or WDM devices (Windows Driver Model) as appropriate for your particular configuration using the respective sliders.

- Choose a sample rate (44.1 or 48kHz) keeping in mind that this is a global setting and all stations must use the same sample rate. If you are using the Axia Livewire driver, you must use 48kHz. Please note that all encoders have high-quality sample rate converters at their inputs, so it is still possible to output a 44.1kHz stream even when using a 48kHz global sample rate.

- Select "Settings" then "Encoders per Station" to select the type and quantity of encoder(s) needed. Encoding formats include AAC, MP3, WMA, and MP2. Each format can support up to four independent streams with different encoder settings per station. A detailed explanation of the benefits and drawbacks of each encoding format along with some recommended settings is included in the section on encoding.
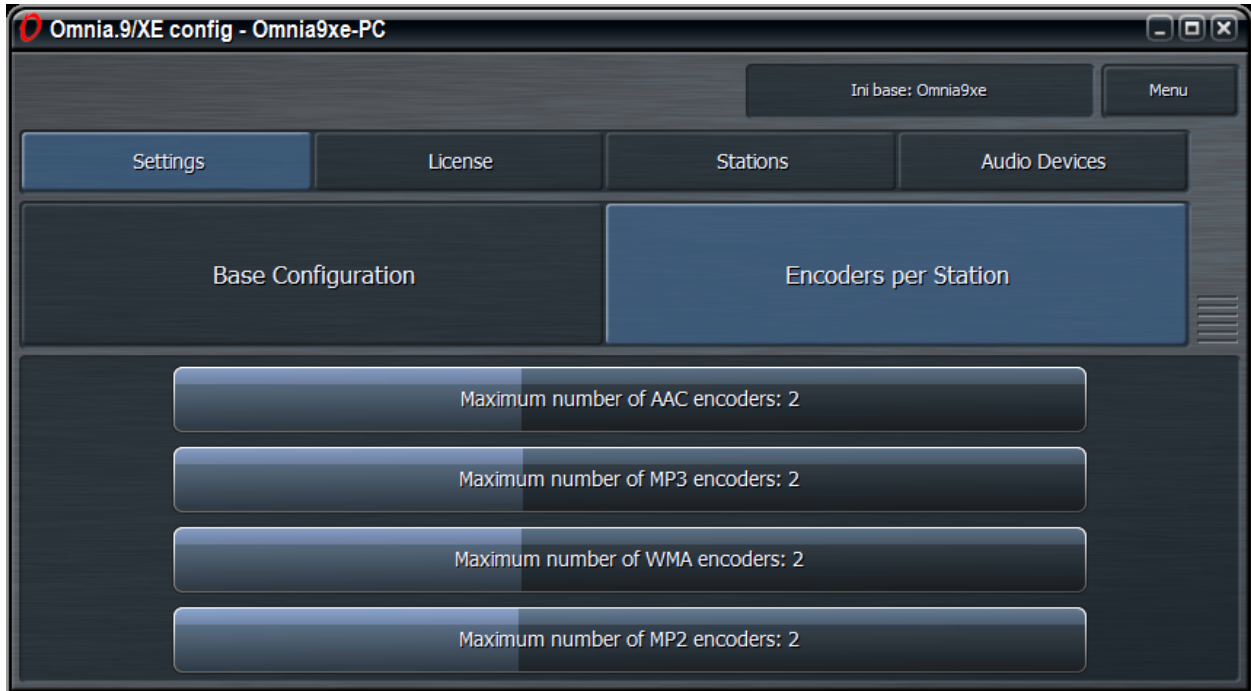
Configuring Audio Devices

Select Audio Devices to display an individual menu for each specified ASIO and WDM device.

ASIO Devices

ASIO is the cleanest and most robust option and is supported by nearly all professional sound cards. It provides a direct link between the software and the audio interface (sound card). It also provides access to all audio channels from within the Omnia.9/XE software.

WDM Devices

If ASIO is not an option, you may use a WDM audio interface instead.

- Choose Kernel Streaming (KS), DirectSound (DS), or Wave from the Interface drop down control.

  - Kernel Streaming is the preferred choice as it is a hardware-direct method of connecting to the WDM audio driver and therefore bypasses native Windows sound settings and sample rate conversion.

  - DirectSound is an emulation that provides greater compatibility by playing audio through the system mixer, but quality is often compromised and Windows will be responsible for converting audio to/from the native format of the sound card.

  - Wave is the default setting because it provides the greatest compatibility and allows multiple applications to access the sound card simultaneously. Like DS, it is an emulation. It is not as robust as KS, particularly on a system whose resources are being taxed.

- Choose the audio source from the Device dropdown control.

- Select the number of channels from the Channel dropdown control (Stereo, 5.1, or 7.1).

- Set the Buffer Size and Buffer Count from their corresponding dropdown menus. The default settings are conservative and "safe" and set to a buffer size of 1024 and a buffer count of 8.

- Check the "Run Test" box and verify a block jitter reading of under 50%. For DirectSound and Wave, you can reduce the block jitter by selecting a buffer size that results in a whole number of milliseconds per block. For example, 882 for 44.1kHz or 960 for 48kHz, both of which result in a nice even 50 milliseconds of audio per block.

WDM Speaker Output

While most monitoring will probably be done remotely through the NfRemote client, the WDM Speaker Output optionally provides a means for local monitoring. The speaker (monitor) output runs on a separate clock with an asynchronous sample rate converter to prevent periodic audio glitches resulting from the output being on a different clock than the input source.
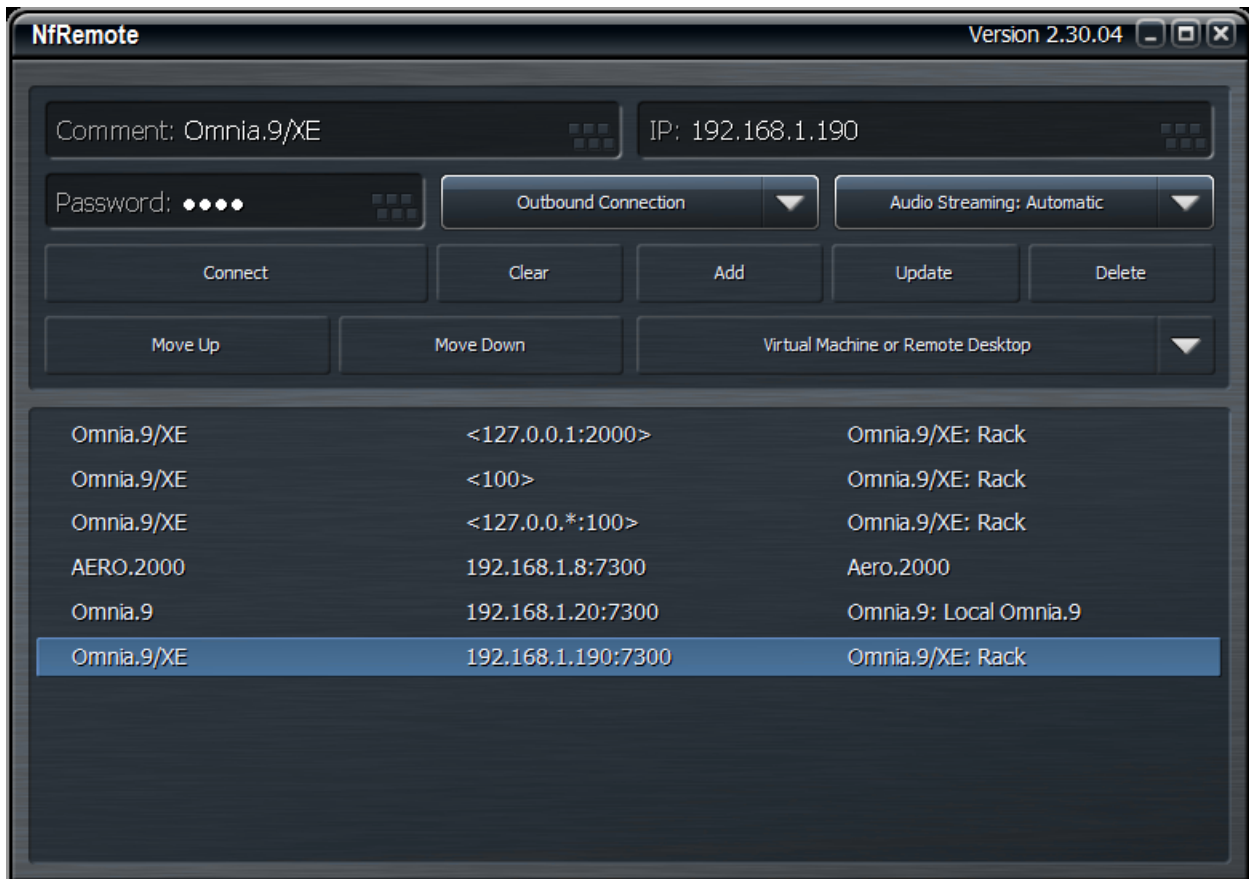
Opening the Omnia.9/XE App

Launch the "Omnia9xe App" to start the Omnia.9/XE software in order to verify system performance from several key hardware indicators.

- Examine the CPU Load meter on the top Menu Bar. The maximum safe reading is 70%.

- Select the Common menu, then System, then Audio I/O Status for each device to see the current actual sample rate and the amount of block jitter. This value should be less than 40% with the default settings. Adjusting buffer size as described above can result in values of under 10%.

Installing and Configuring the NfRemote Client Software

The Omnia.9/XE remote interface software, NfRemote, is a separate "client" software application that can be installed on nearly any desktop, laptop, or tablet PC of reasonably modern vintage running on Windows 2000, Windows XP, Vista, Windows 7 or Windows 8. A shortcut is included in the Installer file.

NfRemote		Version 2.30.04
Comment: Omnia.9/XE		IP: 192.168.1.190
Password: ••••	Outbound Connection	Audio Streaming: Automatic
Connect	Clear	Add	Update	Delete
Move Up	Move Down	Virtual Machine or Remote Desktop

Omnia.9/XE	<127.0.0.1:2000>	Omnia.9/XE: Rack
Omnia.9/XE	<100>	Omnia.9/XE: Rack
Omnia.9/XE	<127.0.0.*:100>	Omnia.9/XE: Rack
AERO.2000	192.168.1.8:7300	Aero.2000
Omnia.9	192.168.1.20:7300	Omnia.9: Local Omnia.9
Omnia.9/XE	192.168.1.190:7300	Omnia.9/XE: Rack

It is possible to configure Omnia.9/XE from the host computer, but this can only be accomplished when the Omnia9xe Service is not running and your streams are not up and running live. The NfRemote client

allows remote access to the Omnia.9/XE service while it is running on the host PC. With few exceptions, it looks and operates just like the interface in the Omnia.9/XE Application.

Before you can connect remotely to Omnia.9/XE, the host PC - that is, the computer running the actual Omnia.9/XE software - must connected to your network and must have an IP addressed assigned to it. Using a password is not required when you are running NfRemote on the host PC, but is necessary if you wish to install NfRemote on any other PC in order to connect to Omnia.9/XE.

Double-click on the NfRemote shortcut to begin the installation.

Once installed and running, enter the IP address and password of the host PC into the corresponding fields of NfRemote and click Connect.

A complete explanation of NfRemote controls and functionality is provided in Chapter 14, "Using the Omnia.9/XE NfRemote Client Software".

Understanding and Installing the Omnia.9/XE Service

The actual processing and streaming by Omnia.9/XE is done as a Windows Service rather than as an executable program. The final step in launching your stream is to start this service. When the Omnia.9/XE service is running, you must use the NfRemote client software to access the user interface.

However - please read and understand the warning below. It's the only time you'll see red text in this manual, so you know we're serious. Really. We're not kidding. We've seen the future if you don't and it's not pretty.

Warning: It is imperative to understand that it is not possible to open the Omnia.9/XE Config or the Omnia.9/XE App applications to make changes to basic configuration settings (such as number of stations or number of encoders) while this service is running.

Therefore, DO NOT launch the Omnia9xe Service install file until you have have verified that your configuration is correct and your CPU and hardware can handle the workload as described above.

Because the service for Omnia.9/XE initiates before the Windows user interface, if the CPU "pegs out" due to excessive demands from Omnia.9/XE, the Windows GUI may never appear at all, leaving you with no way to terminate the very service causing the overload and resulting in a bricked PC.

Once you understand the importance of testing your system before starting the service and have verified that your hardware is up to the task, you can install the Omnia.9/XE service by opening the shortcut labeled Omnia9xe Service Install from the installation shortcut folder.

A shortcut to the Windows Service Control Manager is also included in the installation folder. You can start and stop the Omnia.9/XE service from within the manager. If you want to remove the service completely (either temporarily as an extra safeguard when running the Config App or Omnia.9/XE app, or permanently) a shortcut to do so is provided to the Omnia9xe Service Remove file in the installation folder.
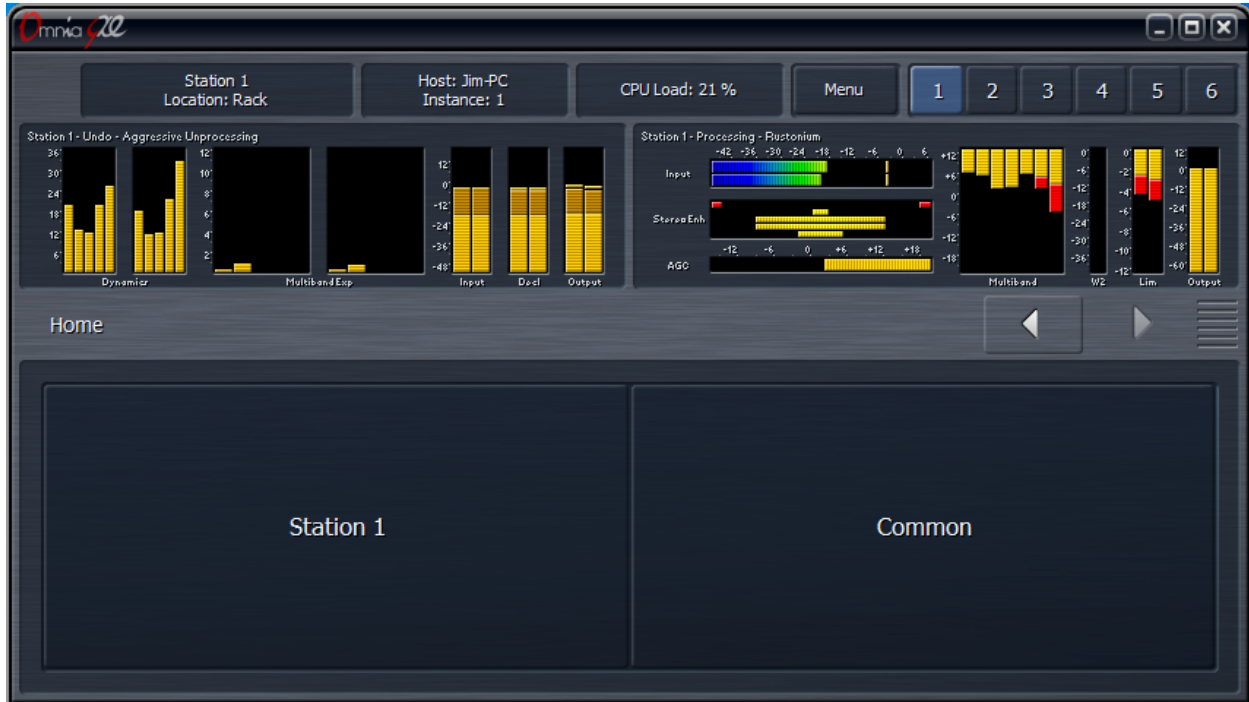
Chapter 4

Omnia.9/XE Home Screen and User Interface

Omnia9/XE Home Screen and User Interface

Opening the Omnia9xe App displays the user interface which is divided into four basic sections: The Menu Bar, the Display Windows, the Navigation Bar, and the Controls for the current menu.

The Menu Bar

The Menu Bar, visible at the top of every screen, shows the station currently being viewed, a "friendly" customizable location reference field, the name of the host PC, the current instance, the CPU load, a Menu button, and 6 display preset buttons.

- The **Menu** button on the front panel has six options: Displays Only, Expert Mode, Lock Position, Maximize, Full Screen, and Close Window.

    - Selecting **Displays Only** allows whichever displays you have selected to occupy the entire window while hiding the controls, useful when you want a more detailed view of the processing meters or an oscilloscope or FFT display. Clicking or tapping anywhere on the screen returns you to the normal view, which shows both displays and controls.

    - The **Expert Mode** control toggles between Expert and Non-Expert views and is explained in more detail below in the "Navigation Modes" section.

    - **Lock Position** makes the window non-moveable and secures it in its current position on the screen. This is especially handy on a touch-screen device if you prefer not to maximize the window.

- Selecting **Maximize** resizes the window to occupy the entire screen and duplicates the function of the standard Windows "maximize" button. Once maximized, the window can be resorted to its previous size by selecting "Restore" or using the Windows restore button.

- **Full Screen** also maximizes the window but removes the standard Windows frame and its minimize, maximize, restore, and resize buttons as well as the task bar and the start menu. Selecting "Restore" from the Menu exits the Full Screen mode.

- Selecting **Close Window**, not surprisingly, closes the application window.

- The **Display Preset** buttons, numbered 1 – 6, let you create and save six separate Home screens. For example, you may wish to devote one display preset to nothing but the Undo and Processing Meters to display these parameters in greater detail. Or, you could devote an entire display preset to the oscilloscope and monitor various points along the processing path to see what the various stages within Omnia.9/XE are doing to the audio signal.

Navigation Modes

The Omnia.9/XE User Interface (UI) is set up in "tiers".

The first tier always contains the highest level (most general) menu options in the form of buttons. When selected, these buttons take you to a second tier of menu options, which are also buttons, but of a more specific nature. The third tier is made up of controls.

There are two navigation modes available from the Menu button: **Expert** and **Non-expert**. The default mode is "Non-Expert." Both modes give you access to the same number of controls; the terms refer only to the way those controls are displayed and laid out.

The **Non-Expert** mode shows fewer controls at once and has a cleaner, less cluttered look, especially in a small window. However, it requires more steps to navigate through the menu system.

The **Exper**t mode shows all tiers and available controls for selected menu sections at once. This provides more direct access to items located deeper in the menu structure but may be visually more intimidating to inexperienced users. It works especially well on a large screen where you have more real estate to work with.

Non-Expert Navigation Mode

The non-expert mode is the default view.
In this mode:

- The top portion of the screen is devoted to the display windows.

- The bottom portion of the screen shows the buttons or controls for the current tier.

- In between these two areas is a navigation bar that shows you exactly where you "are" within the menu system along with navigation arrow buttons that when selected allow you go forward or backward to places you have recently visited.

- Clicking and holding either of the arrow buttons will bring up a list of these places so you can re-visit them directly.

- You can make individual portions of the window larger or smaller to devote more or less on-screen real estate to display windows or control windows by dragging up and down on the re-size bars.
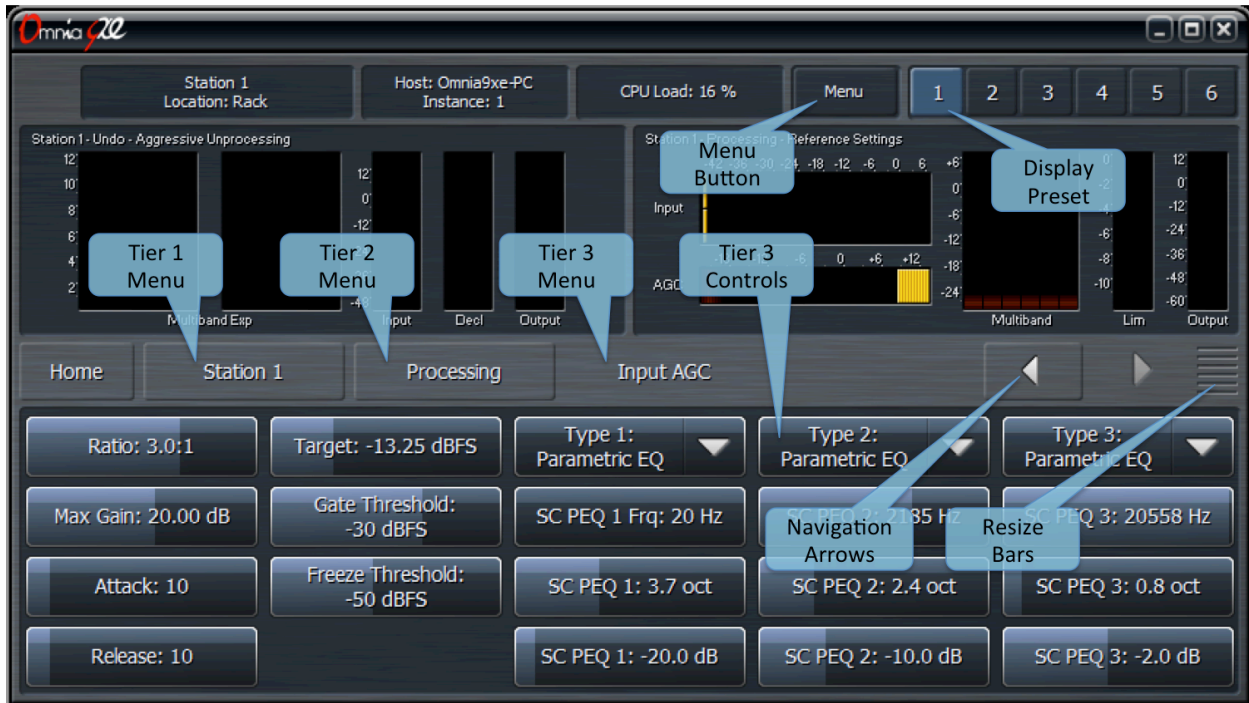
Expert Navigation Mode

Clicking on the Menu button and selecting "Expert Mode" allows more direct control by displaying more information at once.

In this mode:

- The navigation bar and arrows are eliminated.

- The top portion of the screen is devoted to the display windows.

- The middle portion of the screen shows the first, second, and third tier buttons.

- The bottom portion of the screen shows the third tier controls.

- You can make individual portions of the window larger or smaller to devote more or less on-screen real estate to display windows or control windows by dragging up and down on the re-size bars.
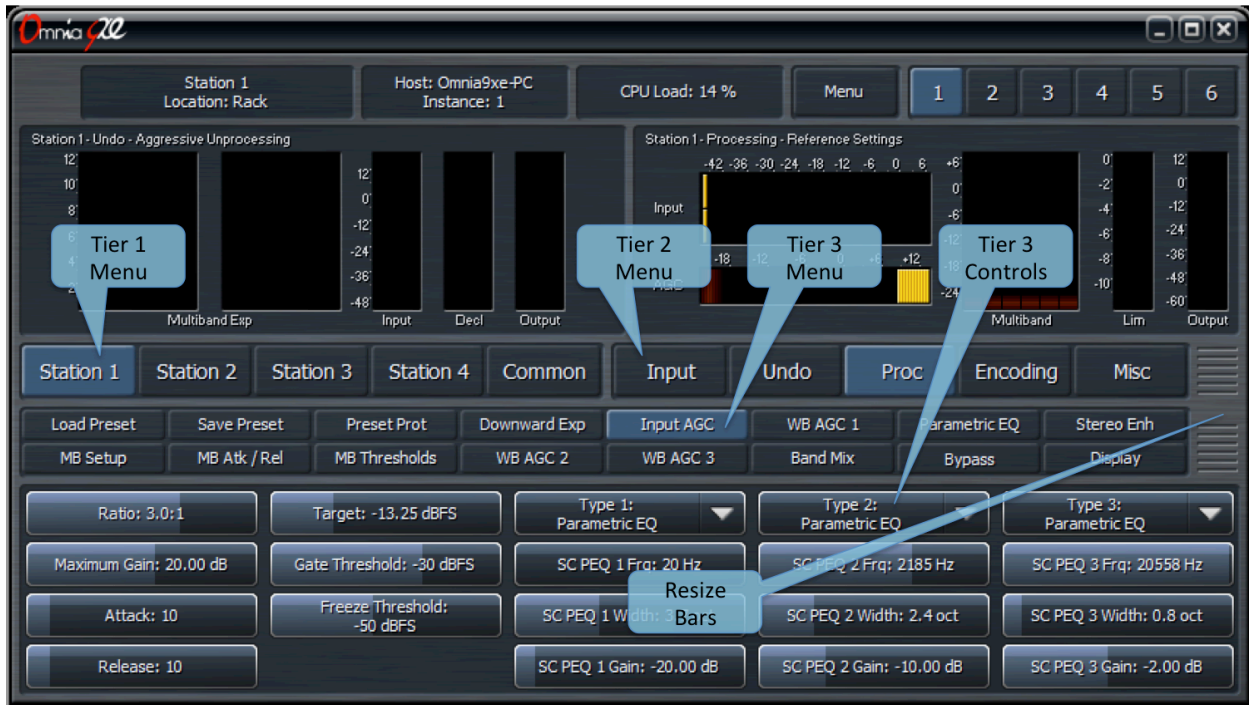
The Three Basic Control Types

There are three basic types of controls in Omnia.9/XE:

- **On/Off buttons** - These controls toggle on and off, thereby enabling or disabling the associated function.

- **Drop-down menus** provide a drop-down list of options. Scrolling over each option highlights it, while a tap or left-click on a mouse selects it.

- **Sliding controls** - Selecting a slider control highlights it. Tapping and holding (or left-clicking and holding with a mouse) a slider allows you to drag it back and forth to adjust its parameter. When using a touch interface, dragging the slider faster will accelerate the rate of change. As long as you haven't lifted your finger (or released the left mouse button) you can continue to drag the slider beyond the edge of the control all the way out to the very edge of the display. Because it is often difficult to make fine adjustments on a small screen, in a virtual machine or through remote desktop, when using a touch interface you can double-click a slider to bring up a pop-up window with two large sliders, one for "coarse" adjustment and one for "fine."
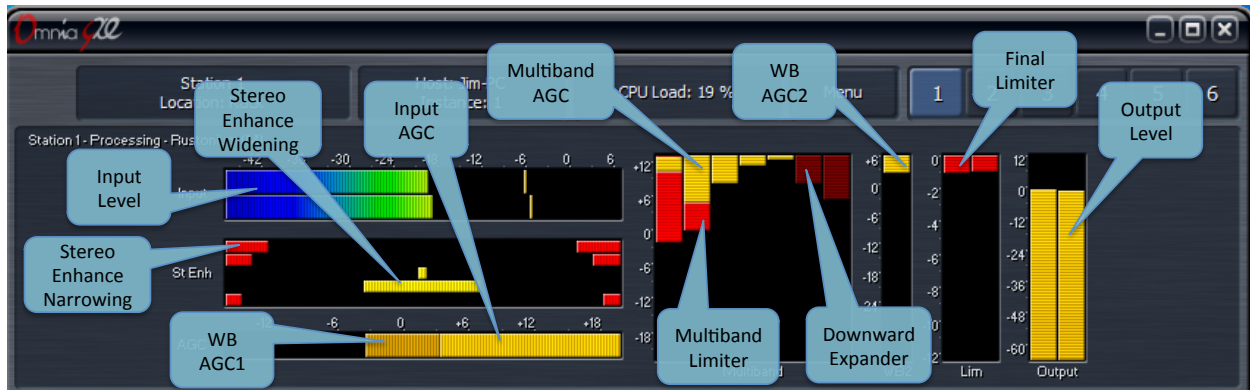
Chapter 5

Reading the Processing Meters

Reading the Processing Meters

Understanding the Processing Meters Display

The Processing Meters Display provides a tremendous amount of information about what's going on inside each of Omnia.9/XE's processors, but in a very efficient and easy-to-read manner.

- The **ITU BS.1770 Input** meters indicate the level of the unprocessed source audio coming into each station processed by Omnia.9/XE with floating peak indicators above each bar.

- The **AGC** meter window shows the levels and action of the Input AGC and the Wideband AGC1 compressors. It also shows the level of the Wideband AGC2 section when it is enabled and positioned before the multiband section.

  Depending upon the settings of the various Input and Wideband AGC1/AGC2 controls, the meter scale can display levels below threshold (positive numbers) as well as levels above threshold (negative numbers). When the meters dip down below "0" the compressors are above threshold and therefore in a state of gain reduction. When they rise above "0" they are below threshold and in a state of gain increase. The combination of Input AGC, AGC1, and AGC2 gain is added together to determine the total amount of AGC gain in place.

- The **Input AGC** levels are indicated by a bright yellow bar. The Wideband AGC1 levels are shown as a dark orange bar extending beyond the Input AGC meter. Wideband AGC2 levels (if this compressor is situated before the multiband section) are shown as a second, darker orange bar beyond the AGC1 meter.

- The **Multiband** meter window shows the levels and action of the AGC, limiter, and noise reduction sections for each band (ranging from 2 to 7 depending upon the preset employed). As is the case with the AGC metering section, the scale will change based upon the settings of various multiband controls.

- The **AGC** level in each band is indicated by a bright yellow bar. The multiband limiter for each band is represented by a bright red bar that flashes beneath the AGC level. The multiband noise reduction for each band is shown by a dark red bar that appears to drop down from the very top of the meter. Gating - the condition in which release is significantly slowed down - is indicated by a dark red ribbon at the bottom of the multiband AGC meters. Freezing - the condition in which

release stops completely - is indicated by the same red ribbon, but slightly brighter. Dry voice detection is indicated by a green ribbon in the same location.

- If the **Wideband AGC2** compressor is enabled and placed after the multiband section, or it is used in one of its Bass modes, its meter will be displayed separately to the right of the multiband display. The Wideband AGC3 meter will appear after the multiband section as well if it has been enabled.

- The **Multiband Stereo Enhancer** meters will be displayed if the current preset employs this feature. Yellow bars extending from the middle outward toward the left and right of the display indicate a widening of the stereo image. Red bars extending from the left and right edges toward the middle indicate a narrowing of the stereo image.
- The **Final Limiter** meters indicate the amount of limiting activity in the final look-ahead peak limiters.

- The **Output Meters** indicated the final processed output levels.
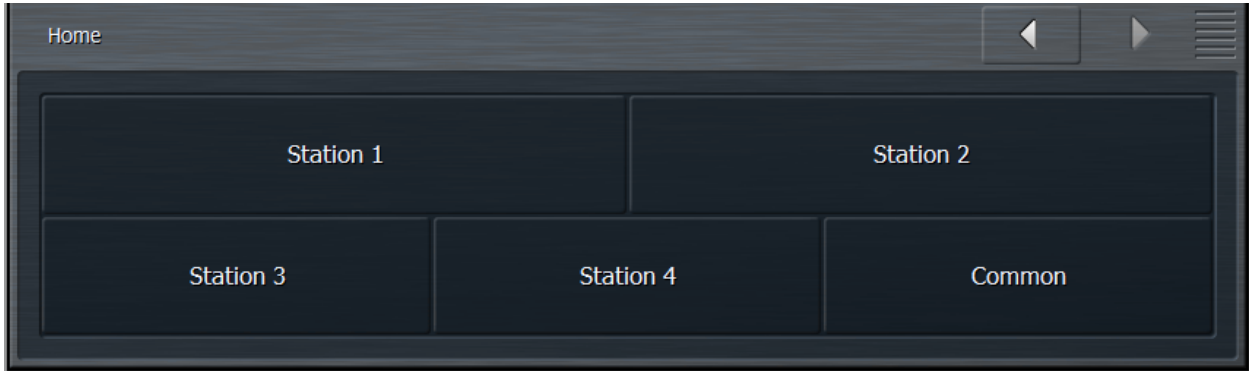
Chapter 6
The Home Menu

The Home Menu

The **Home Menu** represents the highest level menu available in the user interface. It includes sub-menus for each individual **Station** and for **Common** settings.

If you ever lose you way when navigating the menu system, going back to the Home Menu will get you quickly back on familiar ground.

Chapter 7
The Common Menu

The Common Menu

The **Common Menu** contains two sub-menus, **System** and **License**, which provide access to global and configuration settings including licensing information. Some of this information is also found in the Configuration app but duplicated here for convenience. If you specified a device as a **WDM Speaker Output** during configuration, it will also appear in the Common menu.
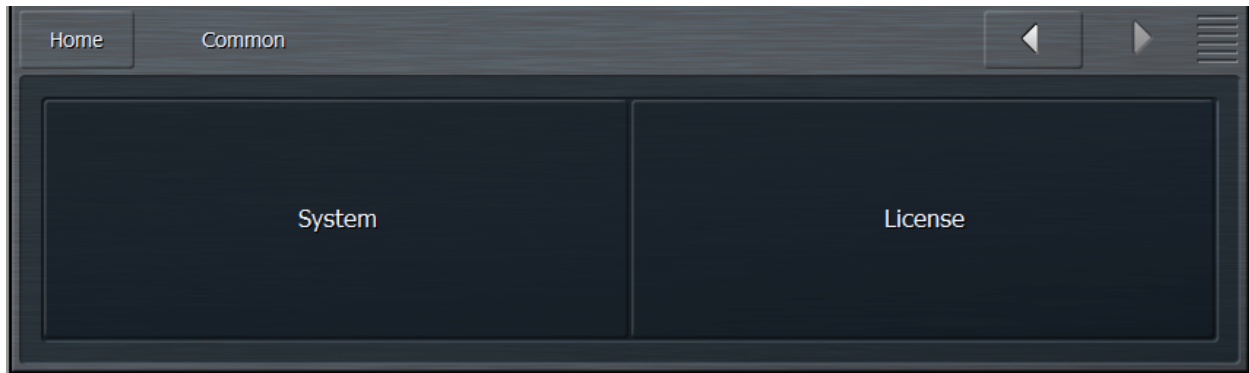
System Menu

The **System Menu** provides access to the System Information, Network Information, Audio I/O Status, Base Configuration, Password, Outbound Connection, and HTTP Access sub-menus.
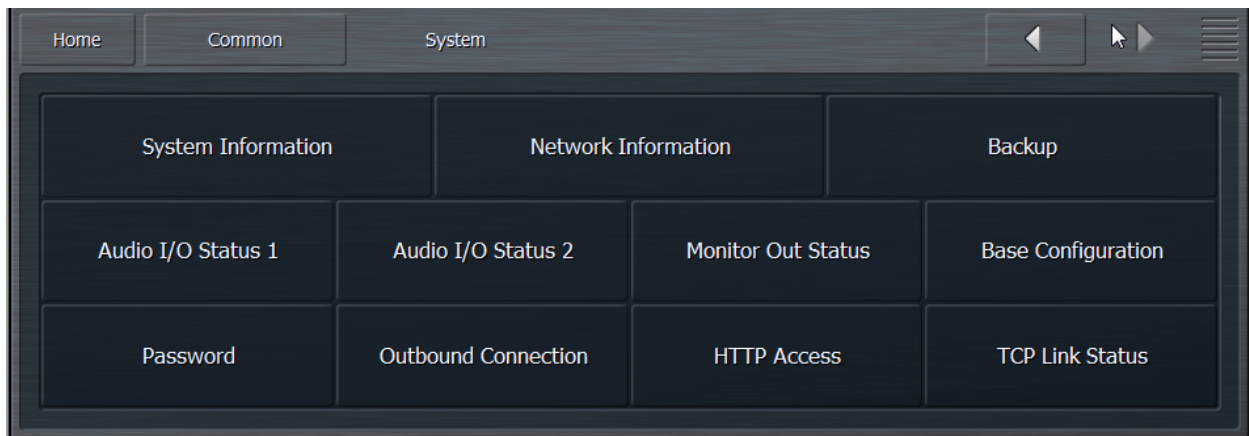
System Information Menu

Statistics for CPU Load, available memory, system up time, software version and GUI version are located here in the **System Information** menu.

Home	Common	System	System Information	◀	▶	☰

CPU Load: 0 %

Available RAM: 3587 MB

Up time: 0 years, 0 days, 00:03.03

Software Version: 0.53.24
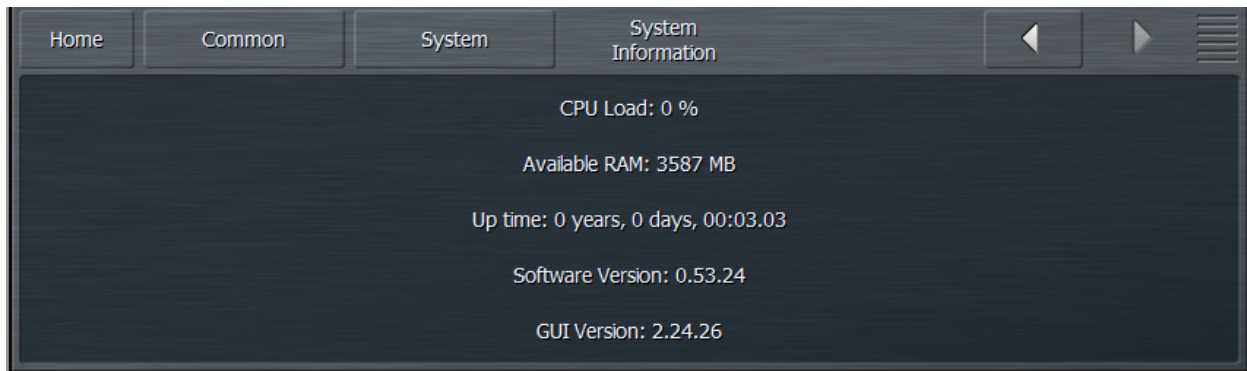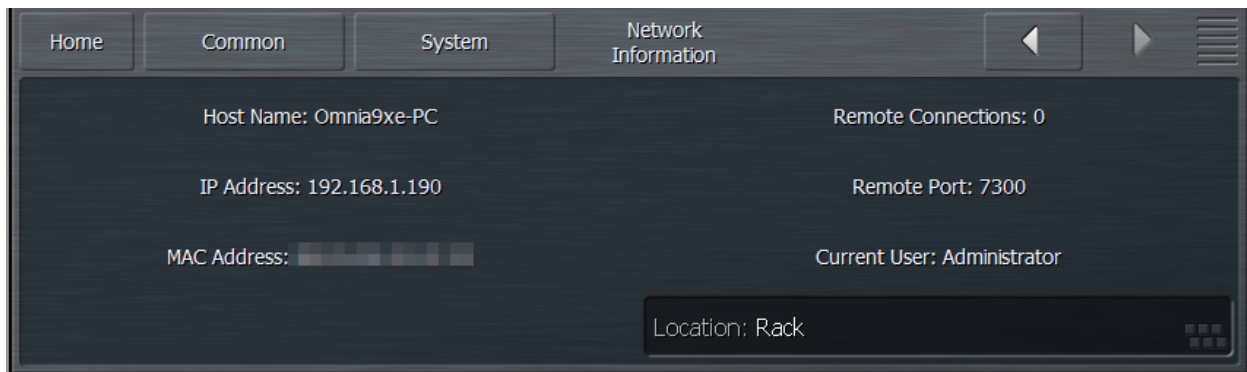
GUI Version: 2.24.26

Network Information Menu

Networking information including Host Name, IP Address, MAC Address, number of active remote connections, Remote Port number, and current user is listed in the **Network Information Menu.** There is also a user-defined text field for **Location** which changes what is displayed in the Location window in the Menu Bar. This location text is also automatically displayed in your connection list when using the NfRemote client.

Home	Common	System	Network Information	◀	▶	☰

Host Name: Omnia9xe-PC   Remote Connections: 0

IP Address: 192.168.1.190   Remote Port: 7300

MAC Address: Current User: Administrator

Location: Rack

Audio I/O Status Menu

Each ASIO and WDM audio device specified in the Configuration App has a corresponding **Audio I/O Menu** that indicates its exact current sample rate, % of block jitter, and buffer size. Jitter for ASIO devices will include the number of jitter warning events ("W" - over 40%), high jitter events ("H" - over 70%), and resets ("R" - if the device stalls completely and needs to be reinitialized).
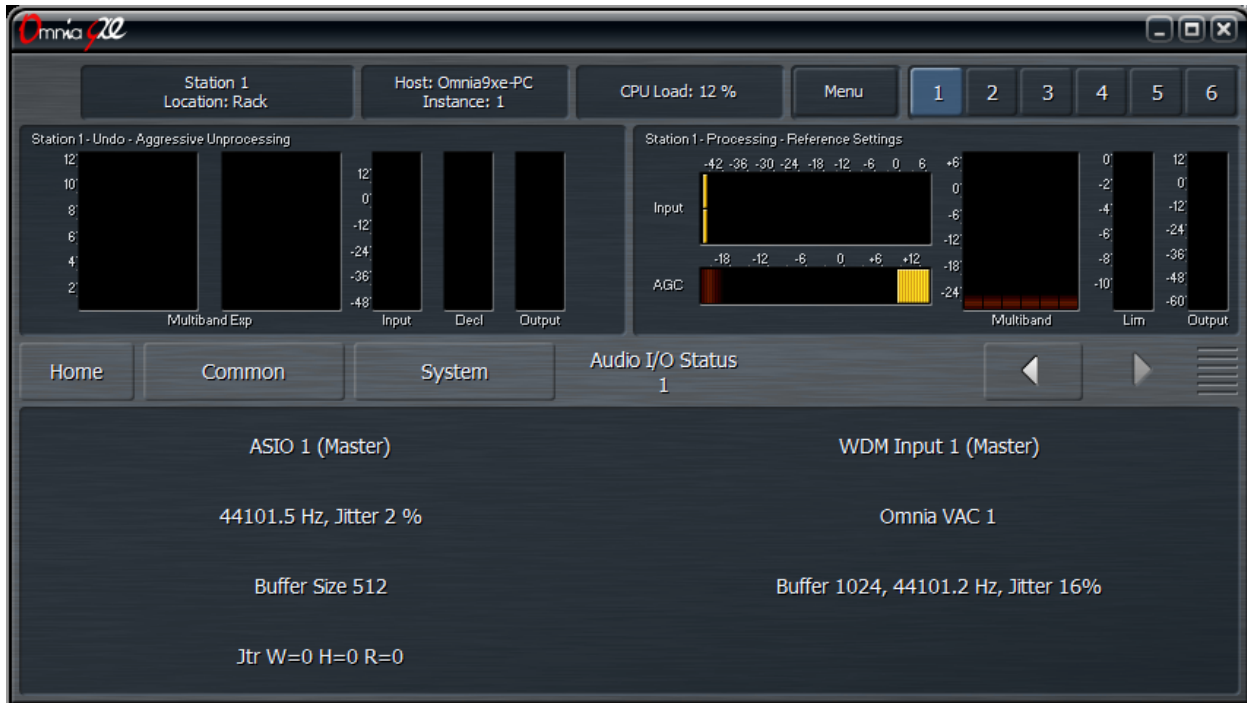
Base Configuration Menu

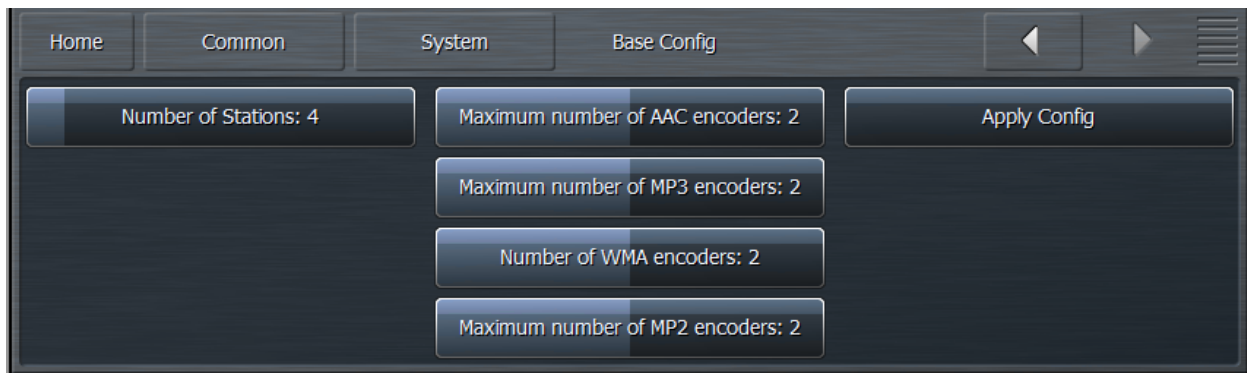The number of stations and the number of each encoder instances for each format can be set in the **Base Configuration** menu as described earlier in the section on Configuration.

Password Menu

In order to access Omnia.9/XE remotely from the NfRemote client it is necessary to enable password protection by entering and re-entering the password in the text fields from the **Password Menu**.

Note that this password ONLY provides protection when using the NfRemote software. It does NOT secure the PC running the Omnia.9/XE software. Good security practice (and common sense!) dictate that the host PC be secured with a Windows password of adequate strength.
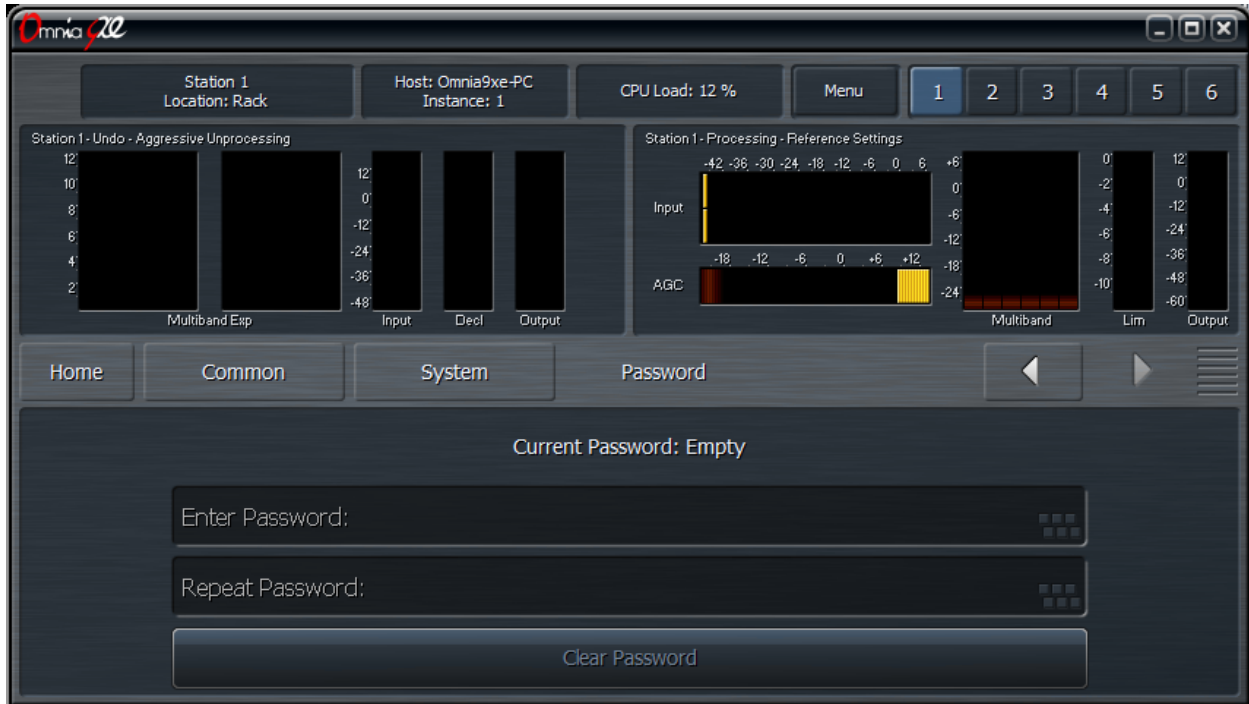
Outbound Connection Menu

Connecting remotely to Omnia.9/XE is normally involves using the NfRemote client software to initiate a a connection. The **Outbound Connection** menu allows this to happen in reverse by providing a means by which Omnia.9/XE can initiate the connection to an NfRemote client. This can be especially useful when working with Omnia Support as it allows communication outside of the local area network (LAN) without having to set up port forwarding.

Details for setting up a corresponding inbound connection are provided in Chapter 14, "Using the NfRemote Client Software".
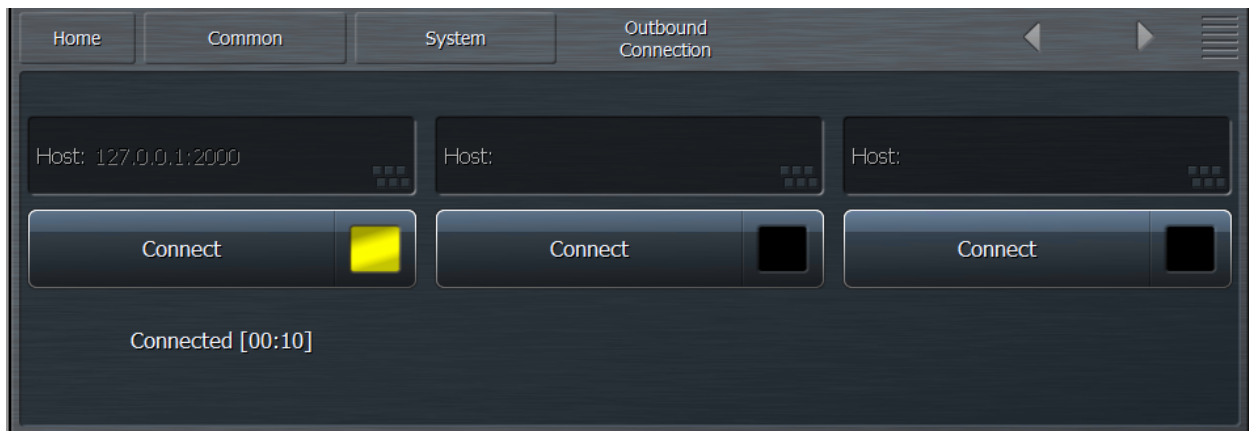
- Enter the IP address (or host name) and listening port of the computer running NfRemote using the following example as a guide: 127.0.0.1:2000 (the IP address and the listening port with a colon in between). The IP address and port should match those on the NfRemote end of the connection. Three connection slots are available.

- Enable the ability to connect by clicking the Connect button. Connection attempts are made every 5 seconds at first. After several unsuccessful attempts, the rate progressively slows down. As network usage is insignificant, Omnia.9/XE can be left to connect indefinitely without concern. Note that disabling this control will NOT terminate an active session.

- The session duration will be displayed at the bottom of the window.

HTTP Access Menu

The **HTTP Access** menu allows you to create a "white list" of allowable IP addresses belonging to network resources that need access to the Omnia.9/XE's HTTP server.
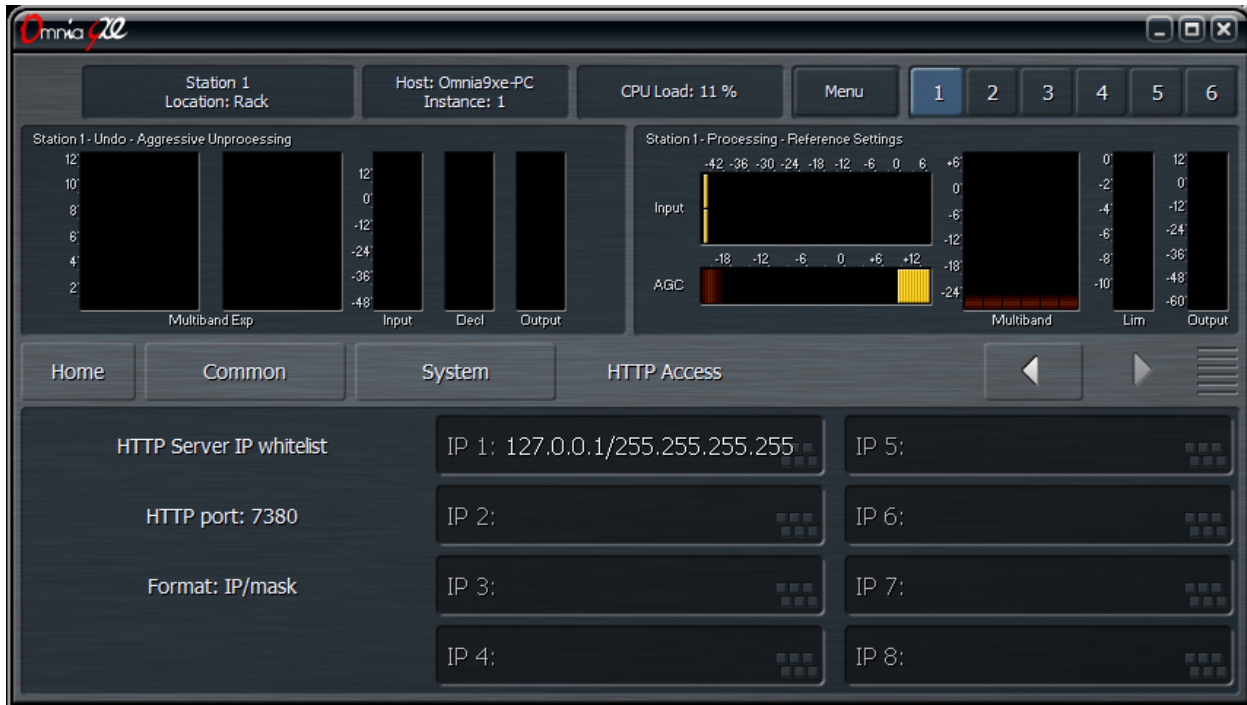
- Enter the IP address and subnet mask information using the following example as a guide: 192.168.1.0/255.255.255.0 (The beginning of the IP range followed by the subnet mask, separated by a forward slash). You may enter an IP address without a mask to allow access to only that specific IP address.

License Menu

The **License Menu** is a duplicate of that found in the Configuration App. It contains an overview of how many license are active and in which slot.

| Home | Common | License | | ◀ | ▶ | ☰ |

Overview	Slot #1	Slot #2	Slot #3
Slot #4	Slot #5	Slot #6	Slot #7	Slot #8

Chapter 8
The Station Menu
Input Menu

The Station Menu - Input Menu

The **Station Menu**(s) contains sub-menus to choose and adjust input sources, manage Undo settings, adjust the processing, provide server settings, set up and adjust the encoders, and customize the station name.
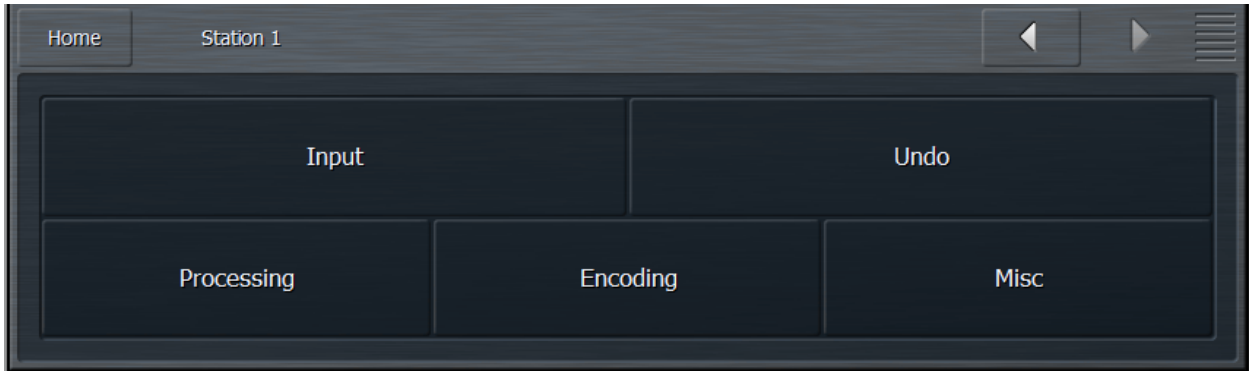
Input Menu

The **Input Menu** has two sub-menus for Source Selection and Display Settings.
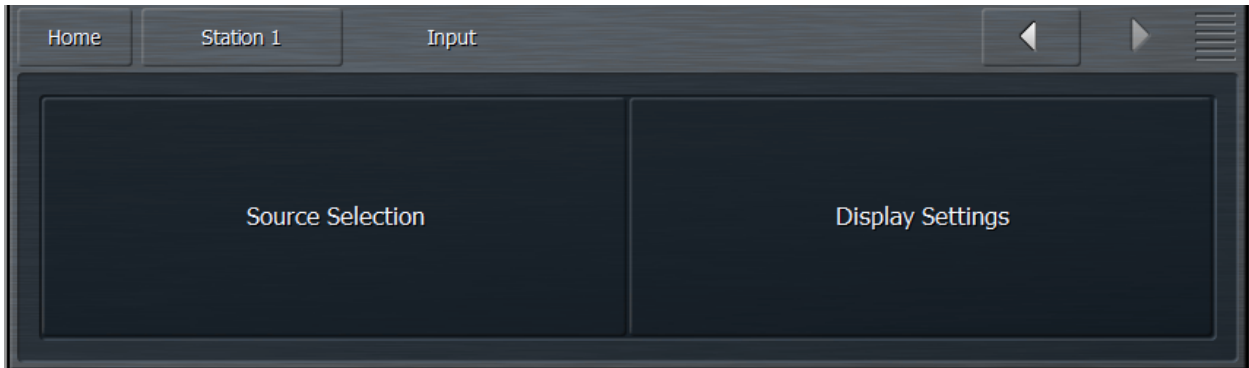
Source Selection Menu

The **Source Selection** menu provides adjustment for input gain, sets the input source, enables the clip warning feature, and selects the source channels.
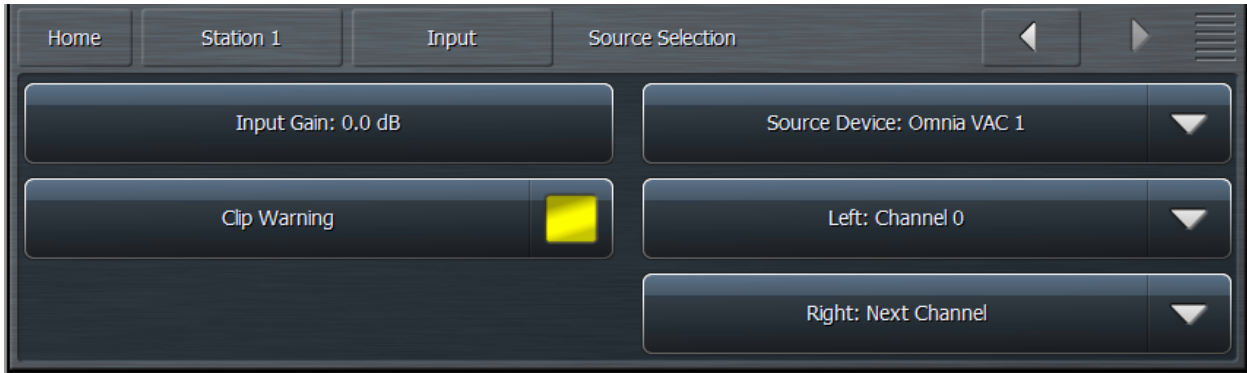
- The **Input Gain** control adjusts the level of the incoming audio from each source.

- When enabled, the **Clip Warning** control will cause the Input meters to flash red when the audio hits full scale.

- The **Source Device** dropdown menu provides a list of possible devices as established in the Audio Devices menu of the Configuration App.

- The **Left Channel** dropdown menu provides a list of possible audio channels. Available channels will be dependent upon the source device selected.

- The **Right Channel** dropdown menu is set to Next Channel by default. In this setting, the next channel from the one chosen in the Left Channel menu is selected, though any available channel may be used instead.

Display Settings Menu

The **Display Settings** Menu allows you to customize what is displayed in up to 8 different windows on-screen at the same time.

Each of the eight Display buttons allows you to choose which processing parameters or visualizations from one of Omnia.9/XE's valuable built-in diagnostic tools are viewable in each window. The six main groups are Undo Meters, Processing Meters, Loudness Meters, Oscilloscope, RTA, and FFT Spectrum Analyzer. You can also turn off any or all displays here.
Since Omnia.9/XE can display such a wide variety of information, it allows you to create and save up to six **Display Presets** per station which are always visible in the top right portion of the display. Any changes you make to the display settings are saved as you make them for that particular preset.

For example, you may wish to devote one preset to nothing but the Undo and Processing Meters to display these parameters in greater detail. Or, you could devote an entire preset to the oscilloscope and monitor the program input, the output of Undo, the processed output to see what the various stages within the processor are doing to the audio signal. The combinations are endless and yours to customize to suit your particular needs and preferences.

Undo Meters Menu

The **Undo Meters** button allows you to display the activity of the entire Undo section. Which specific Undo meters are displayed within this window is determined by settings in the Undo Metering portion of the main Undo menu. A full explanation of the Undo meters is given in the section dedicated to using Undo.

- The **Delay** control, which is also present in (and coupled to) the Processing Meters, Oscilloscope, RTA, and FFT Spectrum Analyzer menus, lets you synchronize Omnia.9/XE's meters with the audio to which you are listening. For example, if you are listening to a stream with a 10 second delay, it would be impossible to properly adjust the processing if that much time passed before you actually heard what you were seeing on the meters! By advancing the Delay control to 10,000ms, what your eyes see on the meters will coincide with what your ears are hearing. Keep in mind, however, that any changes you make to the processing would still take 10 seconds to be audible.

Processing Meters Menu

The **Processing Meters** menu allows you to display the levels and activity of Omnia.9/XE's various gain stages and processing sections.
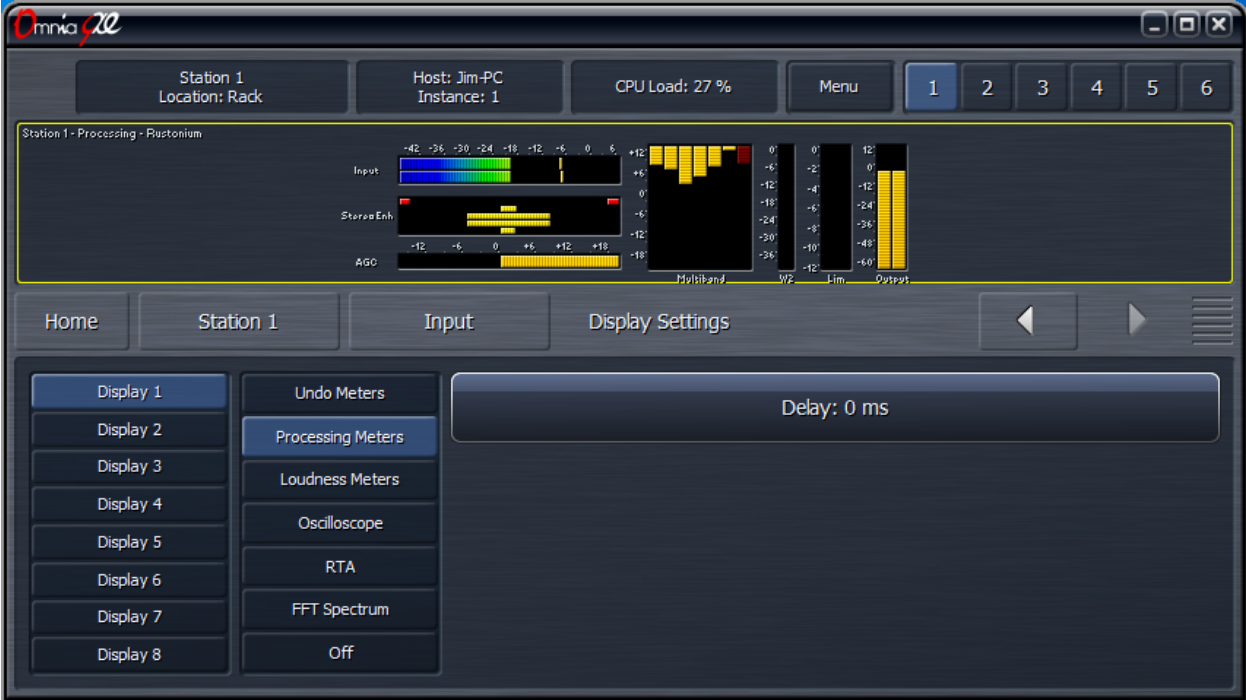
Loudness Meters Menu

The **Loudness Meters** menu provides separate ITU-R BS.1770-compliant loudness meters for Program Input and Output.

Oscilloscope Menu

The **Oscilloscope** menu allows you to monitor a very wide variety of input and output points from various stages of Omnia.9/XE's processing sections on the built-in oscilloscope.
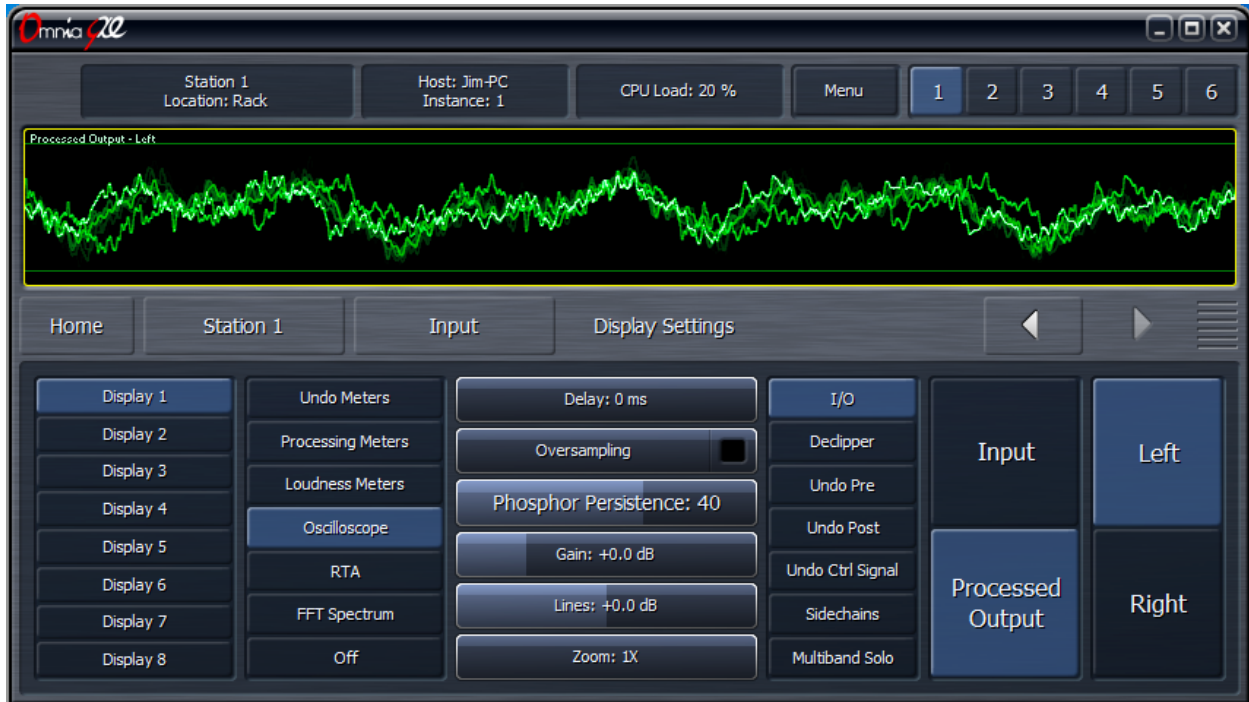
- The **Delay** control works the same here as it does previous and subsequent display sections.

- The **Oversampling** control enables or disables the oscilloscope's 4x oversampling function. With oversampling, the oscilloscope will more accurately display the sampled signal. However, since oversampling consumes more CPU resources, it is turned off by default.

- The **Phosphor Persistence** control adjusts how long latent images remain on the screen. A setting of 0 provides no image delay at all and offers the greatest amount of visual detail for the current information on the scope. Higher settings allow images of signal that has already passed through the scope to remain on the screen for progressively longer periods of time. Setting this control all the way to the right keeps the image from fading at all, useful for catching peaks.

- The **Gain** control adjusts the Y scale of the oscilloscope and moves the grid relative to the signal.

- The **Lines** control adjusts the offset of the reference lines as measured in dB.

- The **Zoom** control allows you to choose between a 1x, 2x, and 4x view of the X scale, similar to setting the sweep time of a traditional oscilloscope.

- The **I/O** (Input/Output) Menu allows you to view the audio signal of each channel at both the input and the processed output.

- From the **De-clipper** patch point, you can view the audio signal before or after de-clipping in either the left or right channels.

- The **Undo Pre** patch points allow you to view the audio signal from each of the 5 individual bands of the Undo section after the de-clipper but before Undo's multiband expansion in either the left or right channel.

- The **Undo Post** patch points allow you to view the audio signal from each individual band or the combined output of all bands of the Undo section after the de-clipper AND after Undo's multiband expansion in either the left or right channel.

- The **Undo Control** Signal patch points allows you to view the control signal from each of the individual bands in either the left or right channel.

- The **Sidechains** (SC) menu allows you to display left- and right-channel audio signals from both the sidechains and output sections of the Input AGC, Wideband AGC1, Wideband AGC2, and Wideband AGC3 compressors.

- From the **Multiband Solo** menu, audio from any one of the bands in the multiband compressor can be displayed. The number of available bands is determined by the number of bands employed in the current preset.

RTA Menu

The **RTA** menu allows you to monitor many of the same sources as those mentioned in the Oscilloscope menu section. The horizontal scale along the bottom of the RTA screen shows frequencies from 20 to 20,000Hz. The vertical scale along the left side shows the amplitude of the audio in decibels.

- The **Delay** control works the same here as it does previous and subsequent display sections.

- The **RTA Resolution** control lets you choose between a 1/3 and 1/6 octave display.

- The **Averages** control can be used to slow down or speed up the RTA's display response. A lower average setting (faster meter response) might be more useful for peak monitoring; while a higher

average setting (slower meter response) might be more useful for calibrating speakers. Higher averages are also useful for adjusting lower frequencies. The recommended setting for speaker calibration is 100. Regardless of how the Averages control is set, the RTA maintains a history "buffer" of 1,000 frames. Advancing the control to 1,000 will immediately display the average of the last 1,000 frames.

• The **Target** control moves the dark red target window and/or the amplitude scale up and down within the display, a useful tool for establishing a target when performing speaker calibration.

• The **Range** control allows you to view a broader or narrower range of audio, indicated by the range scale along the left side of the display.

• The **Target Range** adjusts the height of the target window within a 6dB range in half-decibel increments. Lower settings are better suited to more precise measurements. Moving the Target Range slider all the way to the left hides the target window.

FFT Spectrum Analyzer Menu

The **FFT** menu allows you to monitor the same sources as those mentioned in the Oscilloscope menu section and can be used to identify poor source material. A sharp cutoff at 16kHz and only spurious peaks up to 20kHz is a tell-tale sign of MP3 source material which is to be avoided as re-encoding will significantly degrade audio quality. Lossless sources should be used if at all possible.

• The **Delay** control works the same here as it does previous and subsequent display sections.

• The **Resolution** control selects the number of frequency bands into which the audio is divided. 1x resolution gives you a refresh rate of approximately 48 frames per second and a frequency resolution sufficient for viewing the entire audio spectrum. 8x resolutions gives you a refresh rate of approximately 6 frames per second, but a much finer frequency resolution.

- The **Decay Rate** control adjusts how quickly the display will fall after a peak is displayed. Lower settings are the slowest, while the highest setting is instantaneous.

- The **Peak Hold** control enables or disables the ability to indicate and hold the highest peak achieved. Peak Hold is displayed in the dark green background layer.

- The **X-Center** control centers the display at the selected frequency.

- The **X-Width** control sets the displayed frequency range in kHz.

- The **Y-Center** control centers the display vertically around the selected level in dB.

- The **Y-Range** control sets the displayed level range in dB.

- The **dB Scale Position** control allows you to move the dB scale horizontally to a different place in the display. Advancing the control completely to the right places it in front of the graph at all times.

Chapter 9
The Station Menu
Undo Menu

The Station Menu - Undo Menu

What Exactly Is Undo?

In order to understand how to best use Undo, it is helpful to understand exactly what it is and how it works.

Undo is actually two separate processes - a de-clipper and a multiband expander – which work together to "undo" the poorly mastered, hyper-compressed and clipped audio found on so many modern CD's.

Take a look at the two waveforms below, captured using Omnia.9/XE's built-in oscilloscope. The top window shows the damaged source audio. The bottom window shows the same audio after it has passed through the de-clipper. Notice how the peaks that were clipped during mastering have been restored by the de-clipper.

- *Note:* *The oscilloscope gain was kept at the same level for comparison purposes as there is sufficient internal headroom, and the illustrations that follow do not suggest that the audio is being*
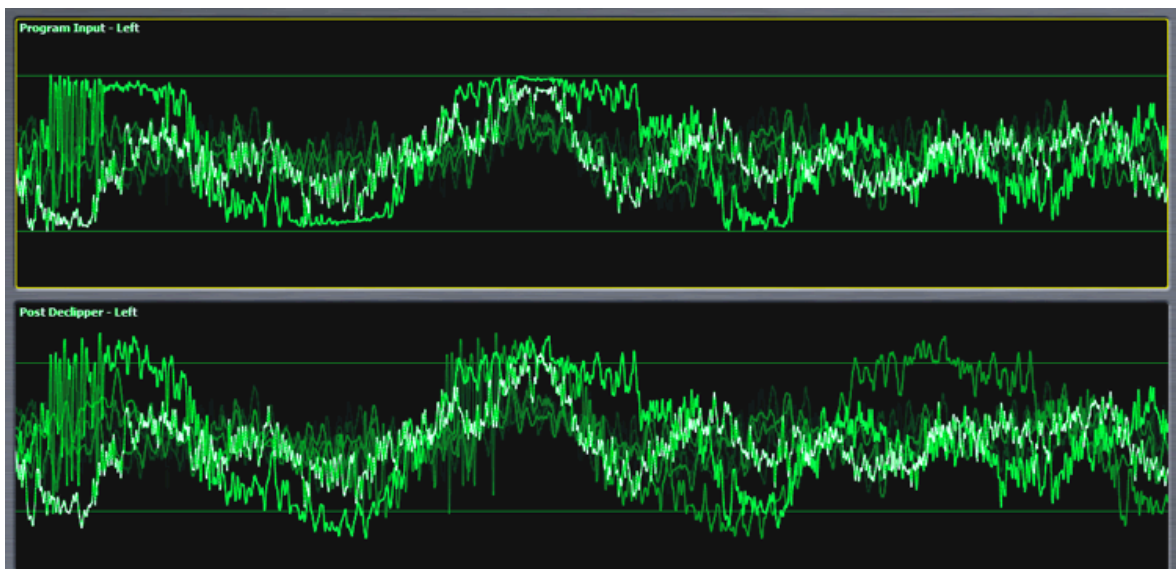
clipped again by any portion of "Undo".The second stage of Undo is a multi-band expander that is designed to restore dynamic range to audio that has been excessively compressed. Each of the five bands of the expander has a corresponding psychoacoustic dynamics detector and a peak detector that examine the peak levels of the incoming audio, and based upon that information plus the settings of the various controls, tells the expander how much dynamic range to restore to the audio.

The illustrations that follow show the damaged audio before de-clipping or expansion. The bottom window shows the same audio after it has been de-clipped and expanded.
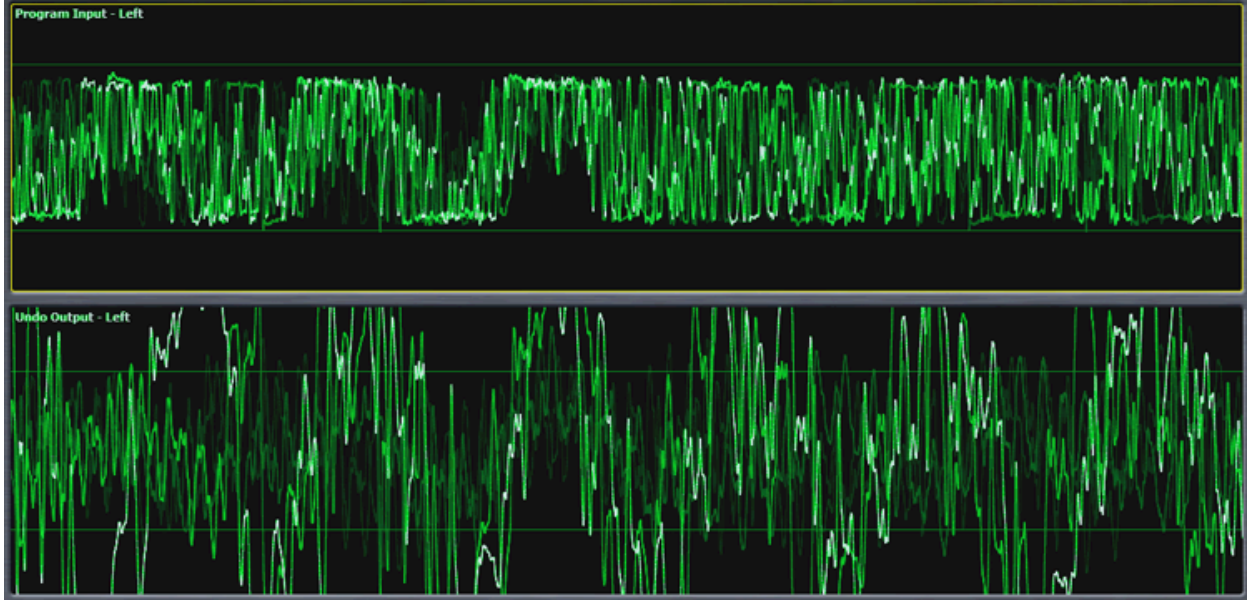
Undo Menu

Each station as a dedicated Undo section.

From the main **Undo** menu, you can load and save Undo presets,enable and disable the de-clipper or multiband expander, choose which functions are displayed in the Undo meter window, adjust the amount of de-clipping, adjust the dynamics detector's ratio, weighting, and peak settings, and set the expander's levels, rates, and ratios. A Display Settings menu is duplicated here for convenience.
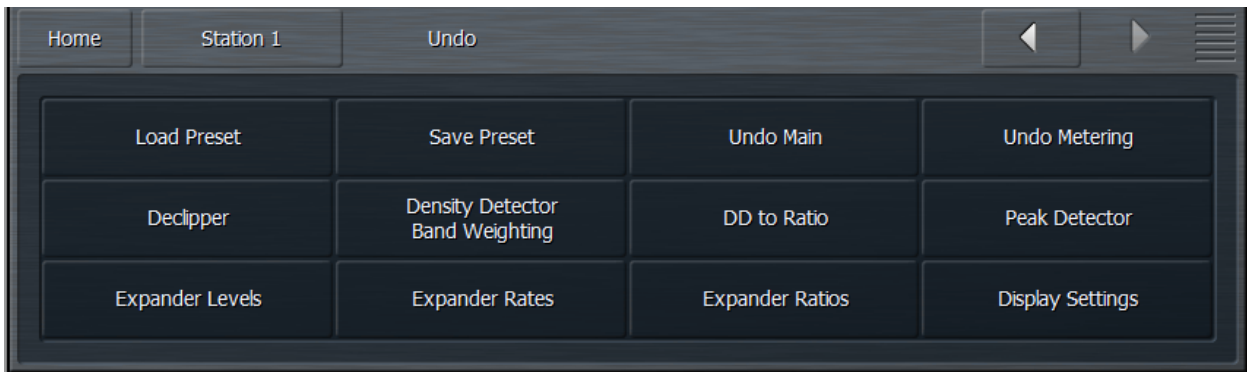
A word of caution about Undo! In keeping with the "spirit" of Omnia.9/XE, we have not blocked access to any of its controls, including those found in the Undo section. And while we would be the last ones to discourage you from making the most of everything Omnia.9/XE has to offer, we feel that outside of adjusting the level of de-clipping, most users will get the most out of this powerful tool by using one of the provided factory Undo presets. For the more curious and brave, read on!

Load Preset Menu

The **Load Preset** menu allows you view the Undo preset currently on the air, toggle back and forth between the current and backup presets, load a new preset, compare the settings of two presets, and delete user presets. The **Save Preset** menu allows you to edit the name of or save over a user preset.
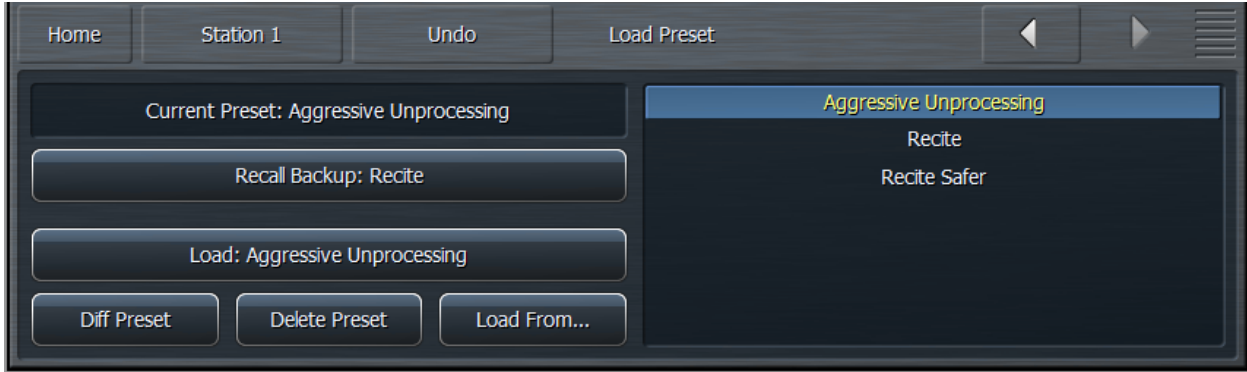
Highlighting a preset from the list places it in the Load button. Selecting the Load button puts that preset on the air and moves it up to the Current Preset position. The preset that was previously on the air will become the Backup Preset. You may also place a preset on the air from the list directly by double clicking it.

After changing any parameter of a preset, an (M) appears next to the Current Preset name and the text turns yellow to indicate the preset has been modified but not yet saved. The Backup Preset also becomes the former (and unmodified) Current Preset. This allows you to toggle back and forth to hear the difference between the original and modified presets on-air by simply clicking the backup button to swap the current and backup slots.

Changes to each parameter you adjusted to create the modified preset are also displayed in yellow on the specific control button in each section of the processing core, providing an instant visual comparison between the modified preset and the original preset upon which it is based.

You can also compare the differences between the Current Preset and the Backup Preset by selecting the Diff Preset button. Any control settings in the Current Preset that are different from those in the Backup Preset will be displayed in yellow in their respective sections. In addition, the backup value of each control is indicated by a small arrow on the bottom of the control itself.

Selecting the Delete Preset button allows you to permanently delete user-created presets. Factory presets cannot be deleted.

Save Preset Menu
The Save Preset menu allows you to edit the name of or save over a user preset.
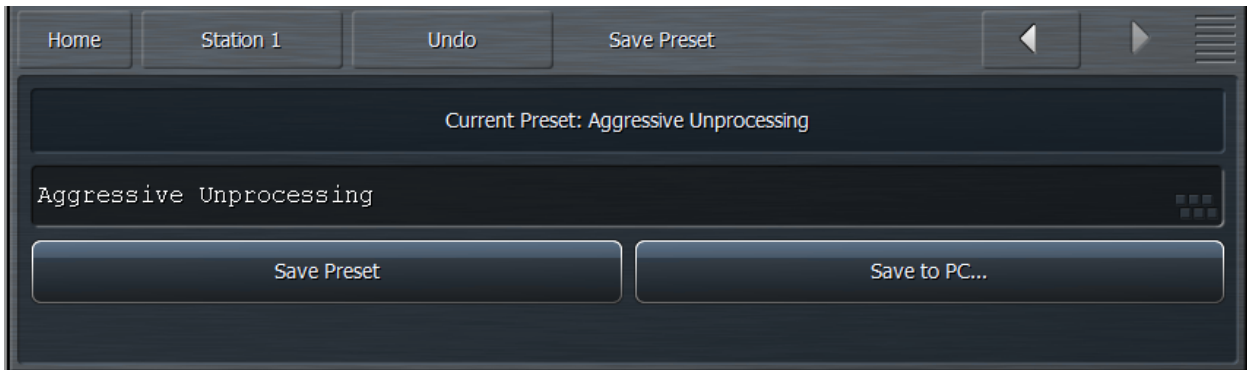
- You can rename your current preset, regardless of whether or not is has been modified, by selecting the **Edit Preset Name** button to bring up the on-screen keyboard. Type in the new name, select "OK", and select Save Preset. The new preset will now appear in the list.

- If your Current Preset has been modified, you also have the option to write over the existing preset without renaming it by selecting **Save Preset**.

That said, we strongly recommend that after you've made some adjustments and are ready to save those changes to a Custom Preset you give it a unique name and instead of saving directly over your current version. That way, should there come a time when you want to revert to a previous incarnation of a preset, you can do so. You cannot, however, if you continually save over the same preset throughout your adjustment process instead of giving it a unique name.

Keep in mind that you cannot give a Custom Preset the same name as an existing Factory Preset.

Undo Main Menu

The **Undo Main** menu contains the controls to enable and disable Undo, to bypass Undo, and to enable and disable the decliner and multiband expander independently.

- The **Power** button turns Undo completely on and off. Turning it off reduces latency and CPU load.

- The **Bypass** control removes both the de-clipper and expander from the audio path but the Undo process itself continues to operate. This makes it possible to monitor the effects of Undo on the oscilloscope regardless of whether or not it is affecting the output audio.

- The **Enable De-clipper** and **Enable Expanders** buttons turn the de-clipper and expander portions of Undo on and off independently and respectively.

Undo Metering Menu

The **Undo Metering** menu determines which of Undo's four metering functions are displayed in the Undo Meters window. A more comprehensive explanation of how to read and understand the Undo meters is provided later in this section when the various Undo controls are explained.

When enabled, the **Show Dynamics Detector** Meters, **Show Expander Ratio** Meters, **Show Speed Governor** Meters, and **Show Expander Threshold** Meters allow the associated meters for each respective section to be displayed in the Undo Meters window.

De-clipper Menu

The **De-clipper** menu contains a single control to set the level of de-clipping in 5 stages. Lower settings offer more conservative de-clipping action, while higher settings offer more de-clipping.
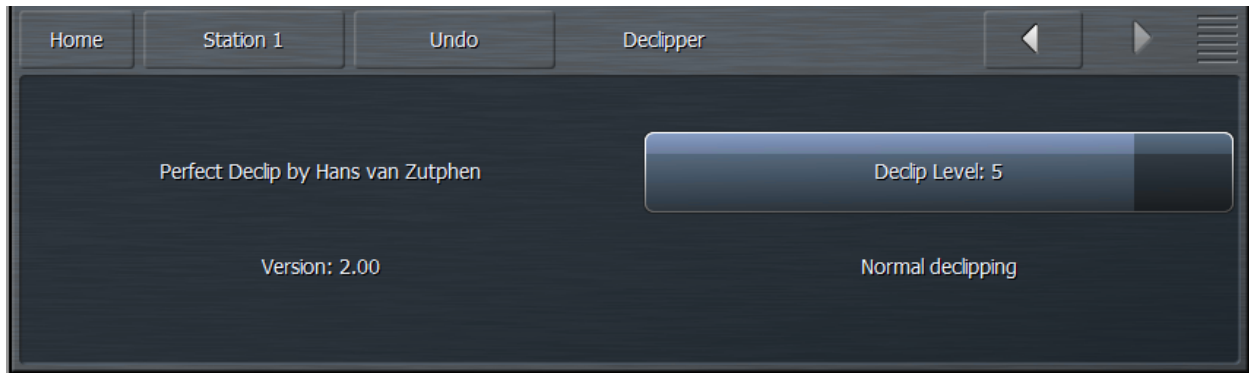
A setting of Level 5 the "normal" setting and recommended for most modern material which have been mastered with high levels of compression and clipping. If your station primarily plays source material that has been cleanly mastered, you may find lower settings are sufficient.

The most effective way to monitor exactly what the de-clipper is doing is to view the pre-de-clipper and post-de-clipper waveforms side-by-side using Omnia.9/XE's built-in oscilloscopes.

Density Detector Band Weighting

Each band of Undo's expander has a corresponding density detector. The **Density Detector Band Weighting** controls allow you to determine how much control each detector provides its own band as well as any other expander band.

Home	Station 1	Undo	Density Detector Band Weighting	◀	▶	☰
DD 1 Weight 1: 100	DD 2 Weight 1: 20	DD 3 Weight 1: 0	DD 4 Weight 1: 0	DD 5 Weight 1: 0		
DD 1 Weight 2: 19	DD 2 Weight 2: 100	DD 3 Weight 2: 20	DD 4 Weight 2: 10	DD 5 Weight 2: 12		
DD 1 Weight 3: 12	DD 2 Weight 3: 20	DD 3 Weight 3: 100	DD 4 Weight 3: 20	DD 5 Weight 3: 15		
DD 1 Weight 4: 0	DD 2 Weight 4: 12	DD 3 Weight 4: 20	DD 4 Weight 4: 100	DD 5 Weight 4: 35		
DD 1 Weight 5: 0	DD 2 Weight 5: 0	DD 3 Weight 5: 0	DD 4 Weight 5: 20	DD 5 Weight 5: 100		

To put this concept into traditional processing terms, it is similar to "band coupling" in a multiband compressor, where all bands may operate completely independently of one another or may influence one another by way of coupling to varying degrees.

The degree to which each dynamics detector band influences and controls each expander band is determined by a 5 x 5 slider matrix. Each column represents a dynamics detector band, while each row represents an expander band.
The default (and recommended) settings are set up so that each expander band controlled almost entirely by its corresponding dynamics detector band, but with some control provided by surrounding dynamics detector bands. Higher values mean a particular dynamics detector band has more influence over a particular expander band, and vice versa.

Please note that in order to provide a means by which to allow one dynamics detector band to provide 100% of the control over a single expander band, every control must be designed to go all the way to 100. When one control is set to 100 and other controls in that band are set to a value higher than 0, thereby causing a sum greater than 100, Undo interprets that information and adjusts the settings of each band to maintain the desired proportion of influence you have set for each band relative to 100.

Dynamics Detector to Ratio Menu

The **Dynamics Detector to Ratio** menu allows you to set time constants of the expansion ratio adjustment as well as the Expansion Start levels and Transition Width levels for each of the 5 bands of Undo's multiband expander.

| Home	Station 1	Undo	Density Detector Band Weighting	◀	▶	☰
DD 1 Weight 1: 100	DD 2 Weight 1: 20	DD 3 Weight 1: 0	DD 4 Weight 1: 0	DD 5 Weight 1: 0		
DD 1 Weight 2: 19	DD 2 Weight 2: 100	DD 3 Weight 2: 20	DD 4 Weight 2: 10	DD 5 Weight 2: 12		
DD 1 Weight 3: 12	DD 2 Weight 3: 20	DD 3 Weight 3: 100	DD 4 Weight 3: 20	DD 5 Weight 3: 15		
DD 1 Weight 4: 0	DD 2 Weight 4: 12	DD 3 Weight 4: 20	DD 4 Weight 4: 100	DD 5 Weight 4: 35		
DD 1 Weight 5: 0	DD 2 Weight 5: 0	DD 3 Weight 5: 0	DD 4 Weight 5: 20	DD 5 Weight 5: 100		

- The **Attack** and **Release** controls for each band determines how quickly the expansion ratio will track along with the dynamic detector settings. The default (and recommended) setting is 100, which means the expansion ratio will track instantaneously with no additional delay.

- When the dynamics detector falls below a certain level, expansion will begin, starting at a minimum ratio of 1:1 and increasing to a maximum ratio of 4:1. The point at which that expansion begins to occur is set by the **Expansion Start** control in each band.

- The relative point at which the expansion reaches its full 4:1 ratio is determined by the setting of the **Transition Width** control in each band.

The detected amount of dynamics in each band is indicated by the **Dynamics** meters. Taller bars indicate more inherent dynamics in the music, while shorter bars indicate less dynamics and are generally the inverse of the **Expansion Ratio** meters. The result of settings in the Dynamics Detector to Ratio menu can be observed on the Expansion Ratio meters in the Undo Meter Window.
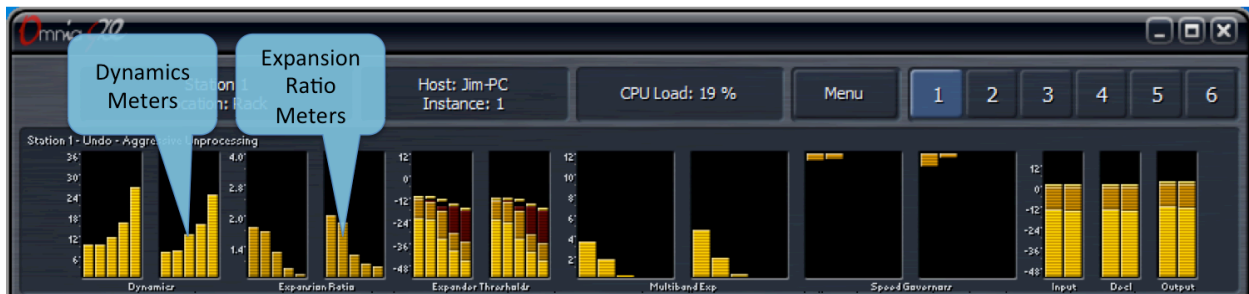
Peak Detector Menu

Undo measures audio peaks as a means by which to set the expander thresholds based upon the incoming program audio. The **Peak Detector** Menu controls the attack and release speed of each band of Undo's Peak Detectors.
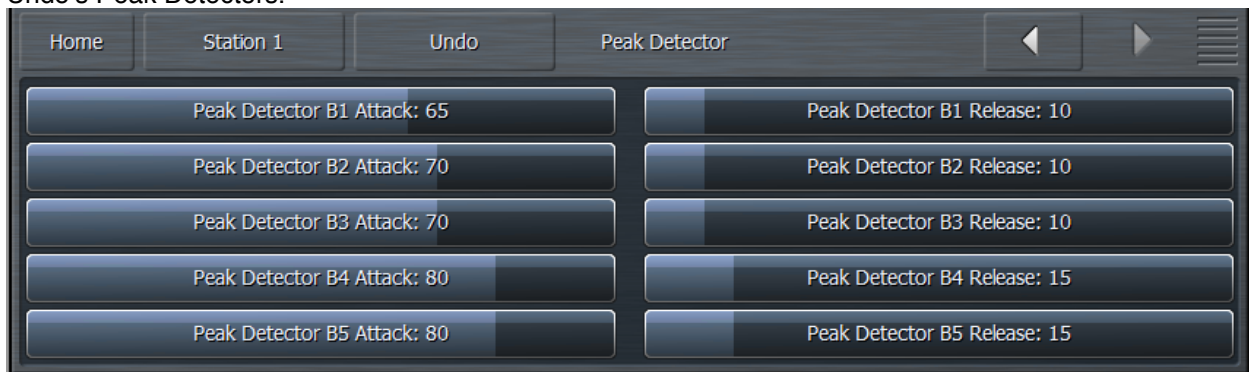
- The **Attack** speed controls determine how quickly Undo reacts to incoming audio peaks. Faster settings will make Undo more reactive to smaller incoming peaks by raising the threshold. Slower settings will effectively lower the threshold.

- The **Release** speed controls determine how quickly Undo allows the peak values to fall back toward the actual audio levels. Faster times will cause the peak levels to fall more quickly.

The effect of the Peak Detector settings is represented by the peak "blips" at the top of the dark orange peak level bars within the **Expander Threshold** meters in the Undo Meter window. Average levels are indicated by the bright orange bars.
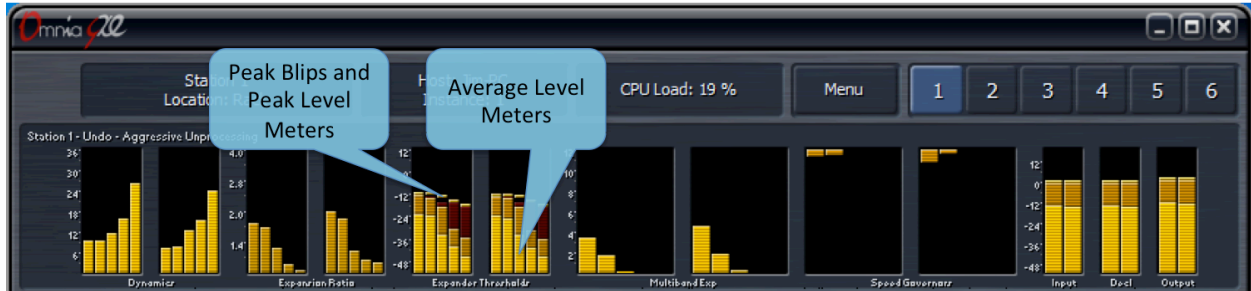
Expander Levels Menu

The **Expander Levels** menu gives you access to the Relative Threshold, Band<Wideband Distance Limit, and Maximum Gain controls of each band of Undo's expanders.

- The **Relative Threshold** control for each band determines how far below the currently detected peak the expander threshold sits. Lower thresholds (moving the slider to the right) result in a greater peak-to-threshold distance and therefore provide more expansion. Higher thresholds (moving the slider to the left) narrow the peak-to-threshold distance and provide less expansion.

- In addition to an adjustable per-band peak detector with adjustable attack and release speeds (whose function is described above in the Peak Detector menu section), each channel also has a wideband peak detector with fixed attack and release speeds. The **Band<WB Distance Limit** controls determine the greatest distance by which an individual band's peak detector can fall relative to the wideband level and keeps any one band from being expanded if there are insufficient levels to begin with. For example, if the current wideband peak is -6dB and this control is set to -10dB, the peak detector for that individual band will not fall below -16dB.

- The **Maximum Gain** control sets the absolute maximum amount of gain that can be applied to the audio in each expander band regardless of program material or any other control settings.

The **Distance Limit** is represented by the dark red bars in the **Expander Thresholds** meter in the Undo Meter window.
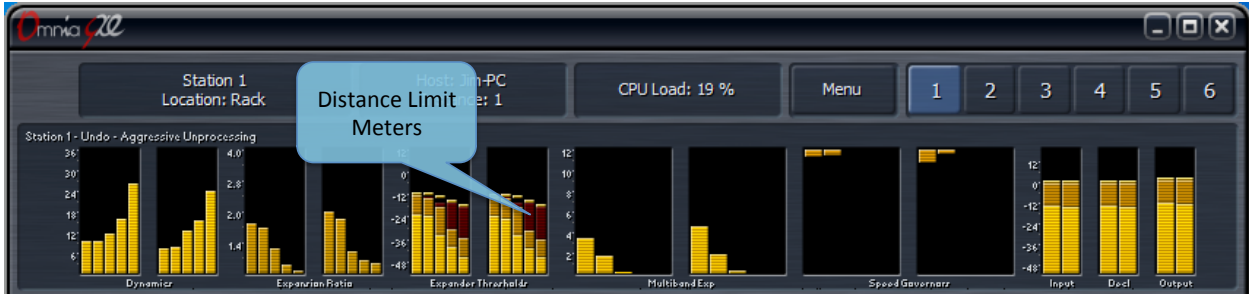
Expander Rates Menu

The **Expander Rates** menu provides controls for the attack and release speeds of the expanders for each band as well as attack and release speed controls for each band's speed governors.
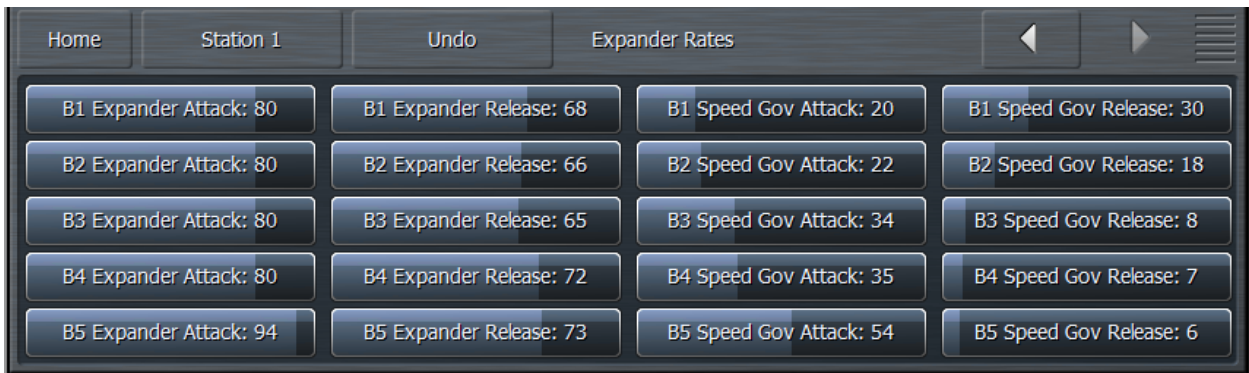
- The **Expander Attack** speed control determines how quickly each band's expander will rise when the audio is in need of expansion. Faster settings will expand the audio more quickly. Using the built-in oscilloscope to view the Undo Control Signal patch point is very helpful for adjusting attack and release speeds as they show the gain that will be applied to the audio in each band. A detailed explanation of how to monitor various patch points within Omnia.9/XE is provided in the Display Settings menu of this manual.

- The **Expander Release** speed control adjusts how quickly the expander levels will fall once the expansion is no longer called for. Faster settings will allow the levels to fall more quickly.

Undo also utilizes a second set of parallel expanders which look at the same input and automatically scale the attack and release characteristics of the main expanders. These Speed Governors detect when the main expanders have been operating at full speed for an extended period of time and will slow down the main expanders.

- The **Governor Attack** speed determines how long the speed governors wait before scaling back the speed of expanders. Faster attack speeds will cause the speed governors to slow down the expanders more quickly.

- The **Governor Release** speed setting determines how quickly the speed governors give up their influence on the main expanders.

The effect of the expander attack and release speeds are indicated by the **Multiband Expander** meters in the Undo Meter window. The levels of the speed governors are indicated in the **Speed Governors** window.
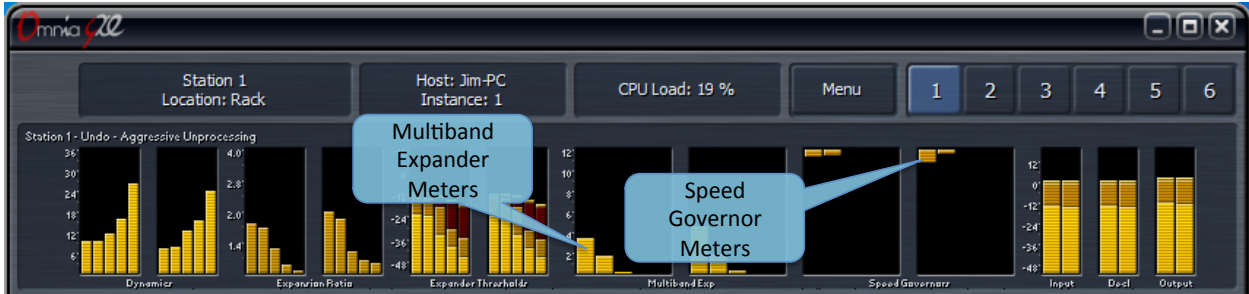
Expander Ratios Menu

Each Undo expander band can operate in a range of ratios between 1:1 and 4:1. The **Minimum Ratio** and **Maximum Ratio** determine the absolute minimum and maximum ratios, respectively, for each band.

There are three additional meters in the Undo metering window for audio levels at the input to Undo, after the de-clipper but before the multiband expanders, and at the output of Undo following the de-clipper and expanders. In each case, the bright orange bar indicates average levels, the darker orange bars indicate peak levels, and the floating orange blips serve as peak hold indicators.
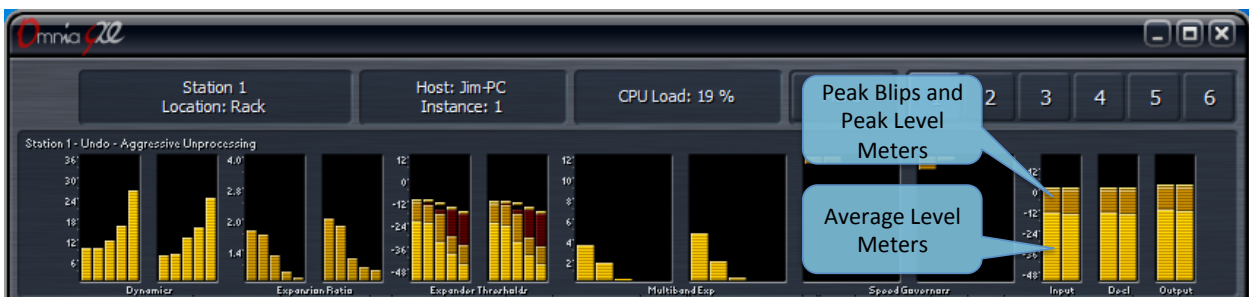
Chapter 10
The Station Menu
Processing Menu

The Station Menu - Processing Menu

The **Processing Menu** provides access to all sections of audio processing core for each station. From this menu, you can load and save presets, enable and configure Preset Protection. This is also where you can adjust audio parameters including the Downward Expanders, Input AGC, Wideband AGCs, Parametric Equalizer, Multiband Stereo Enhancer, Multiband AGC and Limiters, Band Mix, and Final Limiters.
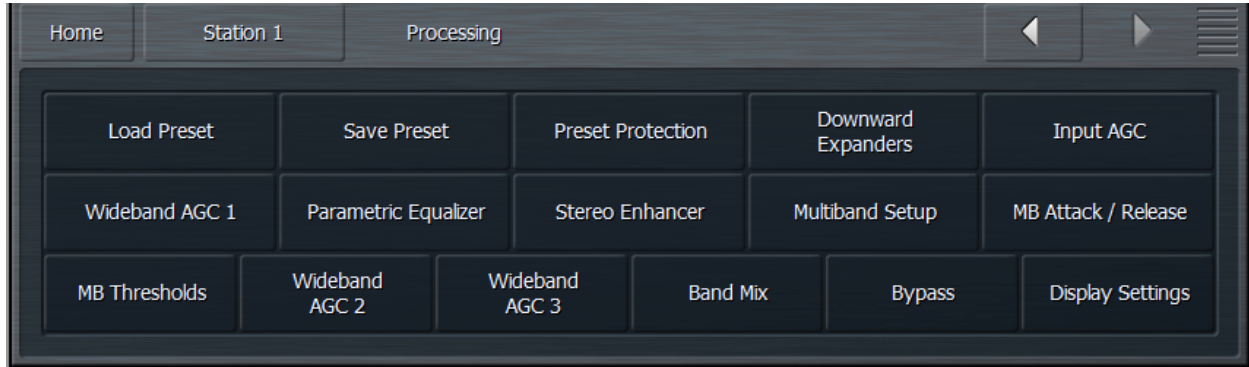
Load Preset and Save Preset Menus

The **Load Preset** controls found here work in the same manner as those in the Undo menu except that they load the processing presets. A detailed explanation of the individual controls can be found in the Undo section.

Preset Protection Menu

The **Preset Protection** Menu allows you to password protect a preset so that it cannot be modified or copied. In addition, the settings for this preset will be hidden from view as long as the preset is locked.
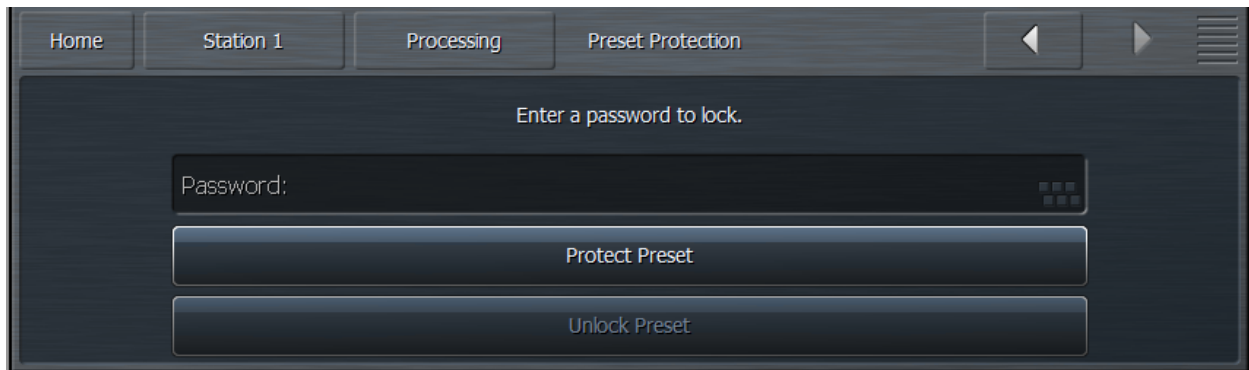
To password protect a preset, enter a password, click on "OK," and then on "Protect Preset." A warning dialogue box will appear. If your intention really is to protect the preset, select "Yes." The final step is saving the new, protected preset with a unique name in the Save Preset menu. Protected presets appear in green in the Load Preset menu. Remember not to accidentally leave an unencrypted version your preset in the Backup Preset slot!

To un-protect a preset, return to the Preset Protection menu, enter your password, click on "OK," and then on "Unlock Preset."

Downward Expanders Menu

The **Downward Expanders** menu allows you to control the maximum gain reduction (downward expansion) for all bands in the Downward Expanders section as well as the threshold, attack and release speeds for each individual band. The total number of bands available in the Downward Expanders section is determined by the number of bands of processing used in the Current Preset.
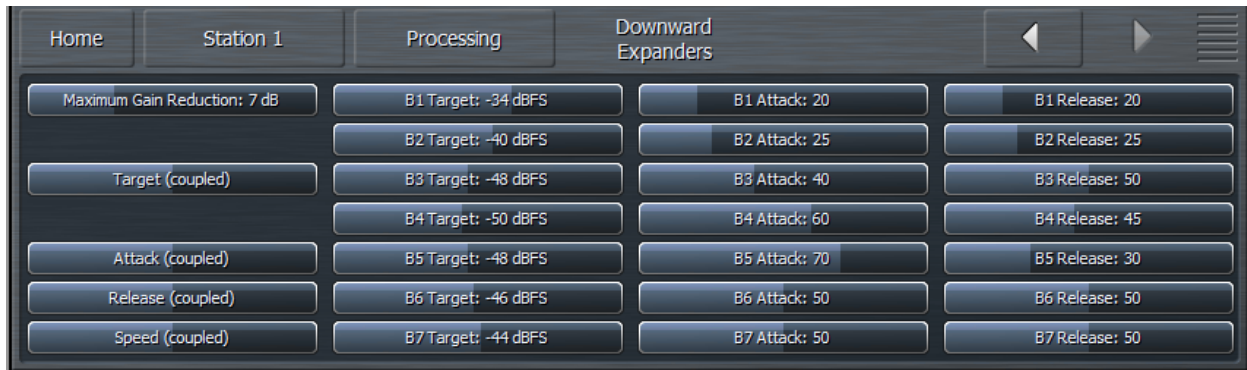
The Downward Expanders operates in a manner similar to a traditional "noise gate," and is useful for reducing unwanted background noise from less-than-perfect program material or from a noisy studio environment. The ratio of these expanders is 2:1.

- The **Maximum Gain Reduction** control determines the greatest amount of gain reduction for all available bands.

- The **Threshold** control for each band sets the point at which downward expansion begins. The range of this control is from -90 to 0dB. The **Threshold (Coupled)** control allows all bands to be adjusted simultaneously by an identical amount.

- The **Attack** control determines the speed with which the Downward Expanders act to reduce gain in each band when the program audio falls below the threshold. Lower settings provide slower attack speeds, while higher settings result in faster attack speeds.

- The **Release** control determines the speed with which the Downward Expanders act to return gain to each band when the program audio rises above threshold. Lower settings provide slower release speeds, while higher settings result in faster release speeds.

Input AGC Menu

The **Input AGC** Menu is used to set the ratio, maximum gain, attack rate, release rate, target, gate threshold, freeze threshold, and sidechain equalizer controls.

The Input AGC is the first gain control stage in Omnia.9/XE following Undo, and is designed to be used as a slower-acting leveler ahead of the Wideband AGC1 and multiband compressor sections that follow it.

It is worth noting that traditional processors only act upon audio above a particular threshold. They are driven into various amounts of gain reduction, but once the audio falls below the threshold, they "run out of room" or "top out," and are incapable of increasing the audio any further. They require some sort of make-up gain control later in the audio chain. The compressors in Omnia.9/XE operate above AND below threshold, controlling the dynamics over a much wider range and do not require makeup gain.

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Home	Station 1	Processing	Input AGC	◀ ▶
Ratio: Inf:1	Target: -12.00 dBFS	Type 1: Parametric EQ ▼	Type 2: Parametric EQ ▼	Type 3: Parametric EQ ▼
Max Gain: 20.00 dB	Gate Threshold: -30 dBFS	SC PEQ 1 Frq: 20 Hz	SC PEQ 2: 1163 Hz	SC PEQ 3: 20558 Hz
Attack: 30	Freeze Threshold: -42 dBFS	SC PEQ 1: 3.7 oct	SC PEQ 2: 1.5 oct	SC PEQ 3: 0.8 oct
Release: 30		SC PEQ 1: -20.0 dB	SC PEQ 2: -8.0 dB	SC PEQ 3: -2.0 dB
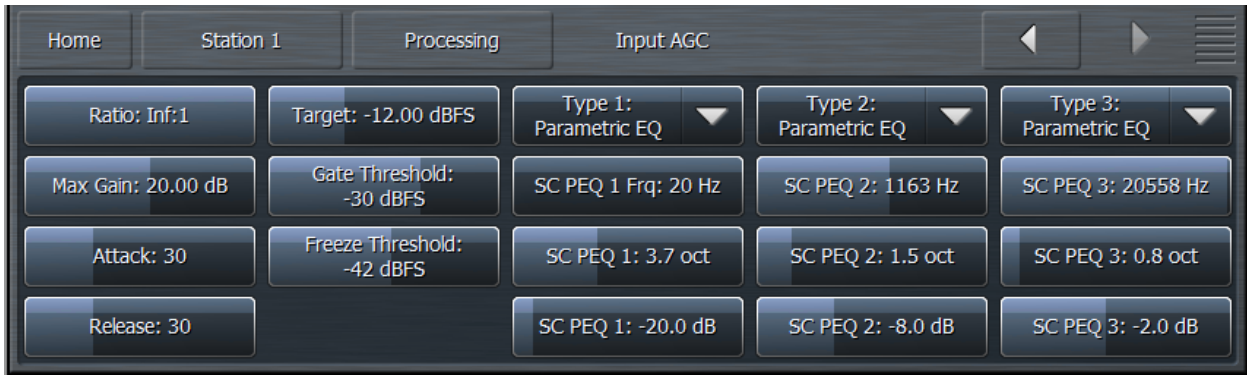
- The **Ratio** control determines how much the output audio will be increased or decreased in relationship to the input audio of the Input AGC section. For example, a ratio of 3:1 means for every 3dB of change in the level of the input audio, the output will be changed by 1dB. Lower (looser) settings provide less control of the dynamics in this section but provide a more open sound, while higher (tighter) settings provide more control at the expense of openness. The range of this control is 1.0:1 to Infinity:1.

- The **Maximum Gain** control works in conjunction with the Ratio control to determine how much gain is available below threshold. If the Input AGC Ratio is set at Infinity:1 and the Input AGC Maximum Gain is set to 36dB, the Input AGC has 36dB of range below threshold. At a ratio of 2.0:1 and the same Maximum gain setting, the range is reduced by half to 18dB. The scale to the left of the Input AGC meter automatically adjusts as needed when changes are made to the Input AGC Maximum Gain or the Input AGC Ratio to accurately reflect how much range is available below threshold.

- The **Attack** control determines the speed with which the Input AGC acts to reduce audio above threshold. Lower settings represent slower attack speeds and allow more audio to pass unprocessed by the Input AGC into subsequent processing stages. Higher settings result in faster attack speeds and allow less unprocessed audio to enter subsequent sections. Because all of Omnia.9/XE's processing stages are to some extent program-dependent, putting actual measures of time on these controls would be pointless, and so the numbers on the various Attack and Release controls throughout are simply relative numbers.

- The **Release** control determines the speed with which the Input AGC increases audio below threshold. Lower settings provide slower release speeds, while higher settings result in faster release speeds.

- The **Target** control sets the target output level of the Input AGC. A lower setting results in a lower output level, while higher settings provide a higher output level. This is similar to a traditional "threshold" control when the levels are below the target.

- The **Gate Threshold** and **Freeze Threshold** controls work together to determine the points at which the release rate of the Input AGC slows by a factor of 3 (gate threshold) or freezes altogether (freeze threshold). The range of these controls is -90dB to 0dB. Lower settings means the audio must drop to a lower level before the release speed slows or freezes. Higher settings means the audio doesn't have to drop as much in level before the input AGC gain slows down or stops. Using higher settings when employing faster Input AGC release speeds can keep the audio from being increased too quickly or too much during quieter passages or pauses. If the display is sized and configured in such a way that the Input AGC meter is shown vertically, a Gate condition will be

indicated by a dim, dark red bar at the bottom of the meter. A Freeze condition will be indicated by a brighter dark red bar.

- The Input AGC features a fully-adjustable, 3-band **Sidechain Equalizer**, which allows you to make it more or less sensitive to particular frequencies. When the controls are not set to cut or boost any frequency, the Input AGC reacts to the full audio spectrum. When set to cut or boost a particular range of frequencies, the Input AGC becomes less sensitive (cut) or more sensitive (boost).

   In addition, an assortment of pre-configured filters are available, including a Low Pass Filter, a High Pass Filter, a Band Pass Filter, a Notch Filter, a Low Shelf EQ, and a High Shelf EQ.

   - The **Type** drop down menu determines what type of EQ or filter is employed.

   - The **Frequency** slider is used to set the center frequency for each band. The range of this control is 20 to 22,050Hz.

   - The **Width** slider determines how much audio above and below the center frequency will also be affected by any boosts or cuts in gain. The range of this control is 0.0 to 10.0 octaves in one-tenth octave increments. Lower values provide a narrower (sharper) boost or cut, while higher values provide a wider (gentler) boost or cut.

   - The **Gain** slider determines how much the audio selected with a combination of the Frequency and Width sliders is boosted or cut. Each band can be boosted or cut by 12dB in one-quarter dB increments for a total range of 24db per band.

      - ***Note:*** *You can see a visual representation of the sidechain EQ curve in a display window using Omnia.9/XE's built-in RTA.*

Wideband AGC1 Menu

The **Wideband AGC1** menu provides access to the sidechain delay, maximum gain, maximum gain reduction, ratio, attack, release, progressive release, target, gate threshold, freeze threshold, and three-band sidechain parametric equalizer controls.

- The **Wideband AGC1 Enable** button enables this section, which follows the Input AGC section and provides additional wideband compression as determined by its various controls. Disabling the Wideband AGC1 also makes this patch point unavailable in an oscilloscope or RTA display window.

- The **Bypass** button removes the Wideband AGC1 compressor from the audio path, but its patch point remains an available option for viewing on the oscilloscope or RTA via the Display Settings menu.

- The **Sidechain Delay** feature is useful for both adding punch and managing the amount of low frequency power (while increasing bass punch).

- The **Maximum Gain**, **Ratio**, **Attack** rate **Release** rate, **Target**, **Gate Threshold**, and **Freeze Threshold** controls work in the same manner as their counterparts in the other sections of the Omnia.9/XE. However, the Maximum Gain control in the Wideband AGC1 section has a range of 24dB.

- The **Maximum Gain Reduction** control sets the maximum amount of gain reduction possible in the Wideband AGC1 compressor, and is adjustable from 0 to 24dB in one-quarter dB increments.

- The **Progressive Release** control determines the degree to which the Wideband AGC1 compressor releases its gain at a faster rate as the audio is driven further toward or into gain reduction. At a setting of 0, the control has no effect and the Release speed control fully determines the rate of release. Increasing the setting progressively makes the release speed of the audio faster as gain decreases.

- The 3-band **Sidechain Equalizer** can be used to make the Wideband AGC1 more or less sensitive to the frequencies determined by the Frequency, Width, and Gain controls, which function exactly like their counterparts in the Input AGC section above. A PEQ preview patch point similar to the one described in the Input AGC section is also available here.

Parametric Equalizer Menu

The **Parametric Equalizer** menu allows you to set up the 6-band phase-linear parametric equalizer, which is located just ahead of the multiband section of the processing core. In addition, an assortment of pre-configured filters are available, including a Low Pass Filter, a High Pass Filter, a Band Pass Filter, a Notch Filter, a Low Shelf EQ, and a High Shelf EQ.
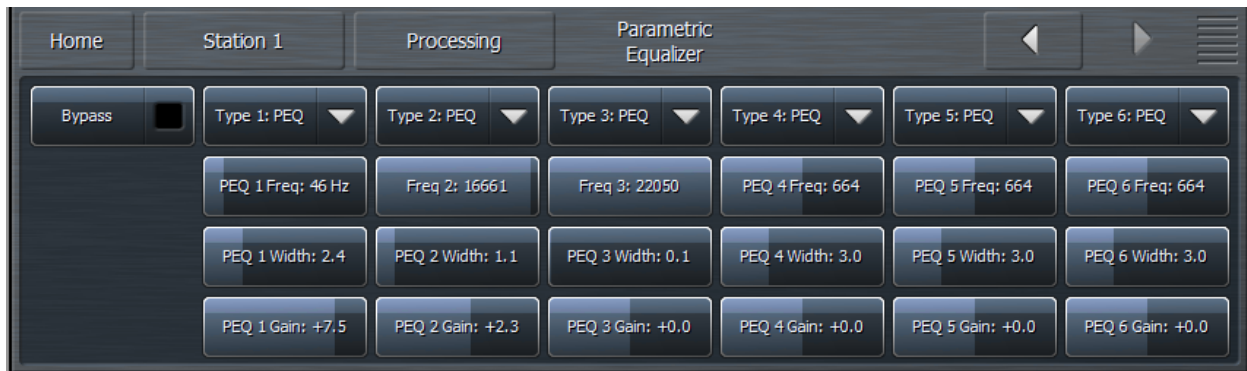
- The **Bypass** button removes the equalizer from the audio path.

- The **Type** drop down menu determines what type of EQ or filter is employed.

- The **Frequency** slider is used to set the center frequency for each band. The range of this control is 20 to 22,050Hz.

- The **Width** slider determines how much audio above and below the center frequency will also be affected by any boosts or cuts in gain. The range of this control is 0.0 to 10.0 octaves in one-tenth octave increments. Lower values provide a narrower (sharper) boost or cut, while higher values provide a wider (gentler) boost or cut.

- The **Gain** slider determines how much the audio selected with a combination of the Frequency and Width sliders is boosted or cut. Each band can be boosted or cut by 12dB in one-quarter dB increments for a total range of 24db per band.

Although changes made in the parametric equalizer section are offset somewhat by the action of the multiband compressors that follow, this does not occur to the degree you might expect based upon your experience with other processors. The parametric equalizer in Omnia.9/XE is a very versatile and powerful tool for creating your on-air sound. A visual representation of the effects of the PEQ using the built-in real time analyzer can be seen in the RTA portion of the Display Settings section of this manual.

Stereo Enhancer Menu

Omnia.9/XE offers a unique multi-band Stereo Enhancer, whose total number of bands is determined by the number of bands of processing used in the Current Preset. Regardless, it never works on bass, which is why Band 1 is never represented and Band 2 may be grayed out. This approach significantly reduces the chance that quieter, hard-panned stereo sounds in a recording with a strong centered mono sound will be shifted out of phase and offers much greater control over the stereo enhancement available in various portions of the spectrum.

Home	Station 1	Processing	Stereo Enhancer	◀	▶	≡
Enable ▢	Tgt Width (cpl)	Max Gain (coupled)	Max G/R (coupled)	Attack (coupled)	Release (coupled)	
	B2 Tgt Width: 75%	B2 Max Gain: 6 dB	B2 Max G/R: 6 dB	B2 Attack: 50	B2 Release: 50	
	B3 Tgt Width: 50%	B3 Max Gain: 4 dB	B3 Max G/R: 6 dB	B3 Attack: 50	B3 Release: 25	
	B4 Tgt Width: 69%	B4 Max Gain: 7 dB	B4 Max G/R: 6 dB	B4 Attack: 30	B4 Release: 15	
	B5 Tgt Width: 80%	B5 Max Gain: 11 dB	B5 Max G/R: 6 dB	B5 Attack: 70	B5 Release: 80	
	B6 Tgt Width: 73%	B6 Max Gain: 10 dB	B6 Max G/R: 6 dB	B6 Attack: 55	B6 Release: 65	
	B7 Tgt Width: 65%	B7 Max Gain: 9 dB	B7 Max G/R: 6 dB	B7 Attack: 30	B7 Release: 35	

- The **Stereo Enhancer** menu gives you access to the Target, Maximum Gain, Maximum Gain Reduction, Attack speed, and Release speed of each of its bands.

- The **Enable** control turns the Stereo Enhancer on or off.

- The **Target Width** control determines the ratio of L+R to L-R. Higher settings provide more stereo enhancement. Adjust this control carefully to avoid turning the stereo image "inside out" by allowing L-R to overpower L+R which ruins mono compatibility and increases multipath distortion.

- The **Maximum Gain** control determines how much the Stereo Enhancer can increase L/R separation in an effort to achieve the Target Width in program material that has a narrow stereo image. The range is between 0 and 18dB.

- The **Maximum Attenuation** control determines how much the Stereo Enhancer can reduce L/R separation in an effort to achieve the Target Width in program material that already has a wide stereo image. The range is between 0 and 18dB.

- The **Attack** control determines the speed at which the stereo image is narrowed. The **Release** control determines the speed at which the stereo image is widened.

- The **Target Width** (coupled), **Maximum Gain** (coupled), **Maximum Attenuation** (coupled), **Attack** (coupled) and **Release** (coupled) controls allow you to adjust all the bands simultaneously by an equal amount in their corresponding sections.

Multiband Setup Menu

The **Multiband Setup** menu provides control over the number of processing bands employed as well as the maximum gain, progressive release, gate threshold, freeze threshold, and gate delay controls. Also found here are the enable buttons for gain reduction override and controls for the gain reduction ratio.

Home	Station 1	Processing	Multiband Setup	◀	▶	☰

|		
Bands: 7	B1 Gain Reduction Ratio Override ■	B1 Gain Reduction Ratio: Inf: 1
Maximum Gain: 16.00 dB	B2 Gain Reduction Ratio Override ■	B2 Gain Reduction Ratio: Inf: 1
Ratio: 5.6:1	B3 Gain Reduction Ratio Override ■	B3 Gain Reduction Ratio: Inf: 1
Progressive Release: 100	B4 Gain Reduction Ratio Override ■	B4 Gain Reduction Ratio: Inf: 1
Gate Threshold: -30 dBFS	B5 Gain Reduction Ratio Override ■	B5 Gain Reduction Ratio: Inf: 1
Freeze Threshold: -42 dBFS	B6 Gain Reduction Ratio Override ■	B6 Gain Reduction Ratio: Inf: 1
Gate Delay: 190 ms	B7 Gain Reduction Ratio Override ■	B7 Gain Reduction Ratio: Inf: 1

- The **Band** slider determines the number of bands in the multiband processing section and ranges from 2 to 7.

- The **Maximum Gain**, **Ratio**, **Gate Threshold**, and **Freeze Threshold** controls work in the same manner as their counterparts in other sections of the Omnia.9/XE.

- The **Gate Delay** control determines how long the Gate Threshold and Freeze Threshold controls wait before they begin working. The range of this control is between 0 and 255ms. Setting the control to "0" means that as soon as audio falls below the threshold as determined by the settings of the Gate and Freeze controls, it immediately slows or stops, respectively. Higher settings mean it will take longer for the release of the audio to slow or stop. A Gate condition will be indicated by a dim, dark red bar at the bottom of the multiband meters. A Freeze condition will be indicated by a brighter dark red bar.

  - *Note: Setting the Gate Delay much lower than the default setting of 79ms will cause the gate to take effect during the brief pauses in dry speech, resulting in a much lower volume level from an announcer mic, for instance, as compared to music. Used creatively, this is actually a very useful tool for controlling announcer/music level balance.*

- The **Progressive Release** control determines the degree to which the Multiband compressor releases its gain at a faster rate when the audio is driven further toward or into gain reduction. At a setting of 0, the control has no effect and the Release speed control fully determines the rate of release both below and above threshold. Increasing the setting progressively makes the release speed of the audio faster as gain decreases.

- The **Gate Threshold** and **Freeze Threshold** controls work in the same manner as their counterparts in the other sections of the Omnia.9/XE.

- The **Gain Reduction Ratio Override** and **Gain Reduction Ratio** controls work together to let you set a different ratio for each band when audio in that band crosses above threshold – that is, when it is driven into gain reduction. The ratio for audio below threshold is always determined by the Ratio control for all bands. Specifically, the Gain Reduction Ratio Override control enables or disables the Gain Reduction Ratio controls, which are sliders that let you set the ratio of audio above threshold from 1:1 to Inf:1.

It is worth mentioning again here the importance of a concept unique to Omnia.9/XE Most (if not all) other processors "top out at 0" – that is, they constantly operate in a state of gain reduction, and once the audio falls below threshold, they can no longer increase it any further. To make up for the fact that they are capable only of reducing gain, they rely upon a "makeup gain" control somewhere downstream in the audio chain to get the levels back up. Omnia.9/XE is not only capable of gain reduction – that is, driving audio levels above threshold as other processors do – but is also capable of increasing gain below threshold, which makes it possible to employ powerful controls like Progressive Release and Gain Reduction Ratio Override.

Multiband Attack/Release Menu

The **Multiband Attack/Release** section lets you control the Attack Speed and Release Speed of each band in the multiband AGC section.
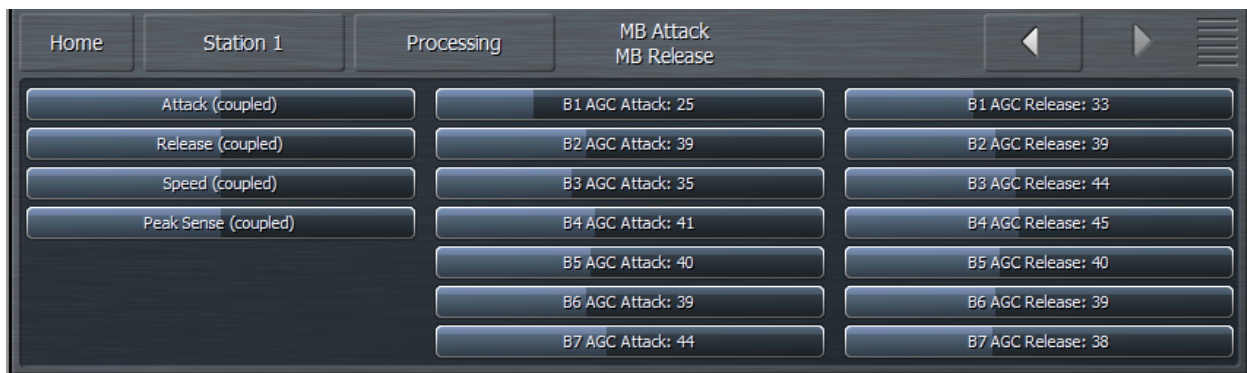

- The **Attack** speed and **Release** speed controls work in the same manner as their counterparts in the other sections of the Omnia.9/XE. However, the behavior of the multiband AGC compressors is also program-dependent.

- The **Attack (coupled)** control allows you to adjust the attack speed of all of the bands simultaneously by an equal amount.

- The **Release (coupled)** control allows you to adjust the release speed of all the bands simultaneously by and equal amount.

- The **Speed (coupled)** control allows you to adjust both the Attack speed and Release speed of all the bands simultaneously by an equal amount.

- The **Peak Sense (Coupled)** control simultaneously adjusts the attack and release rates but in opposition to one another. Sliding the control to the right increases the attack rate and slows the release rate, making it more peak sensitive. Sliding the control to the left decreases the attack rate and speeds up the release rate, making it less peak sensitive.

The attack and release speeds of the multiband limiters are program-dependent and not adjustable.

Multiband Thresholds Menu

The **Multiband Thresholds** menu allows you to set the target for each of the Multiband AGC bands as well as the threshold for each band of the Multiband Limiters. The total number of bands available in the Multiband Thresholds section is determined by the number of bands of processing used in the Current Preset.
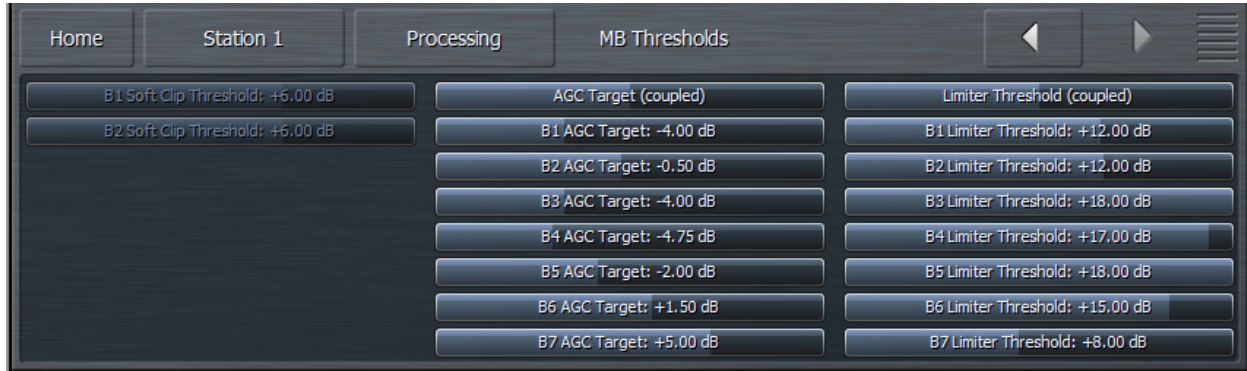
- The **AGC Target** controls set the target output level of each band of the Multiband AGC. A lower setting provides a lower output level, while a higher setting provides a higher output level. These controls have a range between +12 and -12dB in one-tenth dB increments.

- The **Limiter Threshold** controls determine at which point the Multiband Limiter acts upon the incoming audio for its particular band relative to its corresponding AGC Target. For example, a setting of +6dB means that any peaks of less 6dB above the AGC Target level will not be processed by the limiter. These controls have a range between +18 and 0dB in one-tenth dB increments.

Wideband AGC2 Menu

The Omnia.9/XE's Wideband AGC2 control allows you to insert one additional AGC processing stage into the chain as outlined in detail below.

The **Wideband AGC2** menu provides access to the transient enhance, maximum gain, maximum gain reduction, ratio, attack, release, progressive release, target, gate threshold, freeze threshold, and three-band sidechain parametric equalizer controls.

- The **Bypass** button removes the Wideband AGC1 compressor from the audio path, but its patch point remains an available option for viewing on the oscilloscope or RTA via the Display Settings menu.

- The **Transient Enhance**, **Maximum Gain**, **Ratio**, **Attack** speed, **Release** speed, **Target**, **Gate Threshold,** and **Freeze Threshold** controls work in the same manner as their counterparts in the Wideband AGC1 section.

- The **Wideband AGC2** dropdown control enables or disables the Wideband AGC2 section and allows you to choose whether it is situated before or after the Multiband AGC section or used as a dedicated Bass Compressor.

- If you use the AGC2 as a **Bass Compressor**, it will be situated after the Multiband section but will affect only the lower bands, and allow you to push the bass a bit harder without over-driving the final clipper or using excessively fast attack and release speeds in the lower bands of the Multiband AGC.

- **Bass Onl**y ("BO") employs a sidechain filter that allows only the audio from the lower bands to affect gain, so only the lower frequencies are compressed above threshold.

- **Bass Wideband** ("BW") also employs a sidechain filter, but one that contains the entire audio spectrum, so the bass becomes more compressed when the entire mix is above threshold. This mode is most useful when loudness is your primary processing goal, as it could allow full-scale bass audio in circumstances when there is no mid-range or treble audio present. However, there will be less bass present in situations when there are other frequencies present.

Wideband AGC3 Menu

The **Wideband AGC3** menu operates in the same manner as Wideband AGC2, with all of the same controls, but with one difference: It cannot be used as a wideband compressor before the multiband section, only after. It can, however, be used in the Bass Only or Bass Wideband mode just like Wideband AGC2.

Band Mix Menu

The **Band Mix** Menu allows you to adjust the final output of each band after all Wideband and Multiband processing has been applied. It can be used very effectively to tailor the overall spectral balance of your sound, but keep in mind that this is the final point of adjustment before the audio reaches the Final Limiter. In other words, levels increased in the Band Mix section can only be controlled by final limiting, which may result in unexpected or unwanted density on some material, so care is required when making adjustments here.

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| Home | Station 1 | Processing | Band Mix | ◀ | ▶ | ≡ |

| Band Mix (coupled) | B1 Level: -0.25 dB | Bass Clipper ⬜ | Final Limiter Drive: -1.50 dB |
	B2 Level: +0.75 dB	Bass Clipper Slope: 1	Output Level: 0.00 dBFS
	B3 Level: +1.00 dB	Bass Clipper Threshold: -3.3 dBFS	Multiband Limiters ⬜
	B4 Level: +1.50 dB		
	B5 Level: +1.50 dB		
	B6 Level: +2.00 dB		
	B7 Level: +0.00 dB		

- Each **Band Level** control has a range of -12 to +12dB in one-quarter dB increments.

- The **Band Mix (coupled)** control allows you to adjust the output of all bands in the Band Mix section simultaneously and by an equal amount.

- The **Bass Clipper** can be enabled or disabled as desired.

- The **Bass Clipper Slope** control determines the slope characteristics of the bass clipper.

  - **Slope 1** is filtered at a very low frequency so that the low bass stays "clean" even when clipped hard. However, some of the mid-bass will pass through the clipper, which may result in more of the final waveform being taken up by the bass. For lighter processing settings, Slope 1 offers the cleanest and punchiest bass sound.

  - **Slope 2** has a higher cutoff frequency than Slope 1, offering more control over the mid-bass while creating a "dirtier" bass sound with more harmonics. On certain formats, or when listening on a smaller radio, this may be a desirable effect.

  - **Slope 1+2** is simply Slope 1 followed by Slope 2. Most of the low bass will be controlled by Slope 1 and stay nice and clean, while the remaining bass will be taken care of by Slope 2 without affecting the low bass. If you are using heavier overall processing, this option will give you the cleanest sound.

- The **Bass Clipper Threshold** control sets the threshold of the Bass Clipper. Raising the threshold (moving the slider to the left) reduces the amount of clipping performed by the Bass Clipper, but place a greater burden on the Final Limiter. Conversely, lowering the threshold (moving the slider to the right) will yield more bass clipping, which takes some of the load off the Final Limiter, but may result in low frequency distortion if set too low.

- Peak control is provide by the final limiter and can be adjusted with the **Final Limiter Drive** control Reducing limiter drive (sliding the control to the left) reduces the amount of limiting and gives you a more open sound, but at the expense of overall loudness. Conversely, increasing limiter drive (moving the slider to the right) will result in more limiting and a louder output, but at the expense of punch.

- When the **Multiband Limiter** control is enabled, each band of the multiband AGC section has a dedicated limiter associated with it that controls peaks per band. These limiters are bypassed when this control is disabled, leaving only the Final Limiters to control peaks.

- The **Output Level** slider sets the final overall output level of the processed audio.

Bypass Menu

The **Bypass** Menu provides a means by which to completely bypass process and set bypass gain.
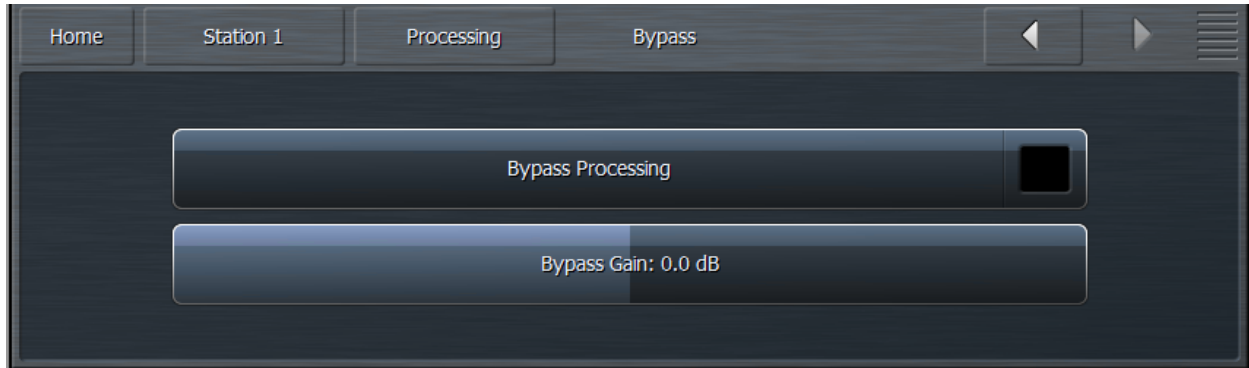
- The **Bypass Processing** control enables and disables all dynamics processing, but does **not** bypass Undo.

- The **Bypass Gain** slider sets the level of output audio when the processing is bypassed and should be adjusted so that processed and unprocessed levels match as closely as possible to eliminate audible gain shifts when comparing processed and unprocessed audio on-air.

Chapter 11
The Station Menu
Encoding Menu

The Station Menu - Encoding Menu

The **Encoding Menu** contains the controls necessary to set up and configure each of the enabled encoders and is where details of the servers can be entered. The number of encoders of each type displayed here is dependent upon the initial settings in the Configuration App. In the examples below, we have set up two encoders for each supported format - AAC, MP3, MP2, and WMA.
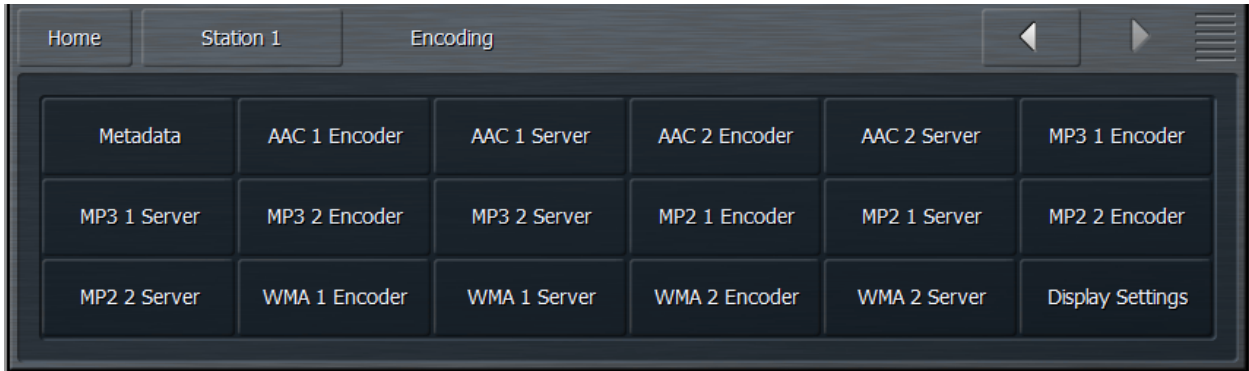
Metadata Menu

Omnia 9/XE's built-in http server can be used to feed information such as the song title and artist information to the stream through the **Metadata Menu.**

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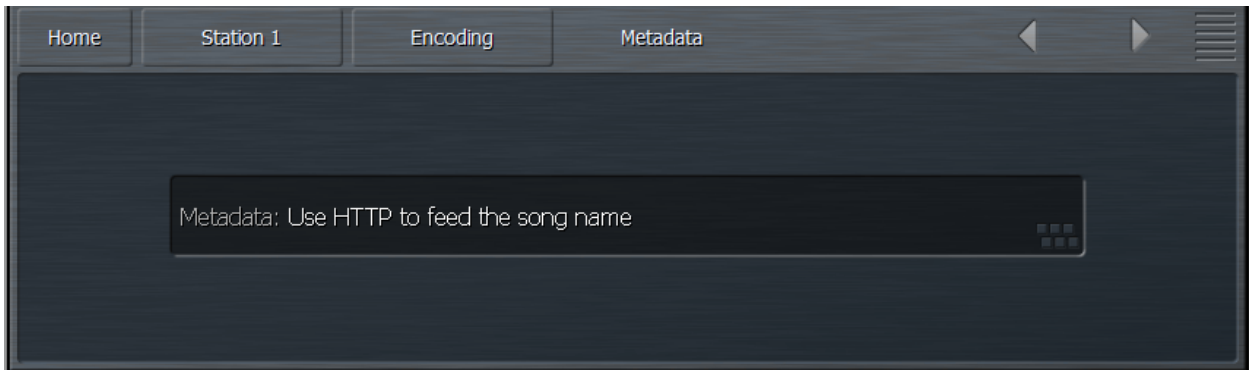
AAC Encoder Menu

The **AAC Encoder** menu allows you to enable or disable the stream, adjust its output gain, enter the stream's title, and select the SBR Mode, channel mode, sample rate, bit rate, and audio bandwidth.

- The **Title** field allows you to enter the stream title displayed to listeners and on public listings, such as shoutcast.com.

- The **Enable** control enables or disables the stream. When enabled, the encoder will continually try to connect to the specified server.

- The **SBR (Spectral Band Replication) Mode** toggles between Off, On, and Oversampled. We recommend setting this control to Off (AAC) for bit rates of 192 kbps or higher, On (AAC+) for bit rates of 96 kbps or lower, and Oversampled for bitrates between 112 and 160 kbps.

- The **Channel Mode** menu provides a choice between Mono, Stereo, Stereo Independent, Parametric Stereo, and Dual Channel. We recommend setting this control to Stereo for bit rates over 40kbps and to Parametric Stereo for bit rates of 40kbps and lower. Mono should be used only if the program material is mono. Because the Stereo Independent and Dual Channel settings force the encoder to divide bits equally between the left and right channels instead of allowing it to allocate the bits as needed, we recommend not using these settings unless you have a specific reason to do so.

- The **Bit Rate** menu allows you to choose encoding bit rates between 16kbps and 320 kbps. Your choice of bit rate settings will significantly influence the audio quality of your stream. However, you will also need to take other factors into consideration, such as the total amount of bandwidth available for the stream, the anticipated number of users, your maximum upload speed, and the download speed limits of your intended users.

- The **Audio Bandwidth** menu allows you to set the high frequency range of the stream between 8 and 20kHz. Unless you have a specific reason to set the bandwidth manually, we strongly recommend leaving this control set to Encoder Default.

- The **Sample Rate** menu will either maintain the global sample rate as set in the Configuration App or implement a high-quality re-sample to 32, 44.1, or 96kHz.

- The setting of the **Gain** control is critical. Normally, this control can be left at 0.0. However, there are times when you will want to reduce the input level into the encoder. For example, if you are employing a lot of peak limiting and using AAC+ encoding at a low bit rate, there will be peak overshoots created as a side effect of the bit rate reduction.

- Unless the input level into the encoder is reduced, it will be up to the listener's decoder to deal with the excessive levels, which will most certainly degrade the quality of the audio. You may also consider backing off the Final Limiter Drive.

AAC Server Menu

The **AAC Server** menu allows you to input and set up your server settings.

- The **Title** field should contain the name of your audio stream.

- The **Server** field should contain the URL of the server, including the port, in the following format: "streamingserver.example.com:8000"

- The **Mount Point** field (for Icecast 2 servers) should contain the mount point on the server, in the following format: "/example1.mp3" following the server name.

- Enabling the **Allow Public Listing** control sets the "public listing allowed" flag in the stream, as determined by the server.

- The **Server Type** control allows you to choose between Shoutcast and Icecast2.

- The **Mime Type** control specifies the HTTP Mime Type of the stream and determines whether Audio/AAC or Audio/AACP settings are used to provide for player compatibility.

- The **URL** field is a metadata field whose information is sent to the server to be (potentially) displayed in the listings and player. The URL is typically the station's official home page

- The **User Name** field contains the user (log-in) name for the server. This is usually "source" for Icecast servers.

- The **Password** field contains the password for the server.

- The **Genre** field is also a metadata field. The type or style of content found on the stream is typically entered here.

- The **Apply Immediately** control determines whether any changes you make to the server settings take effect immediately or when you manually reconnect.

MP3 Encoder Menu

The **MP3 Encoder** menu allows you to enable or disable the stream, adjust its output gain, enter the stream's title, and select the channel mode, sample rate, bit rate, audio bandwidth, and encoding quality.
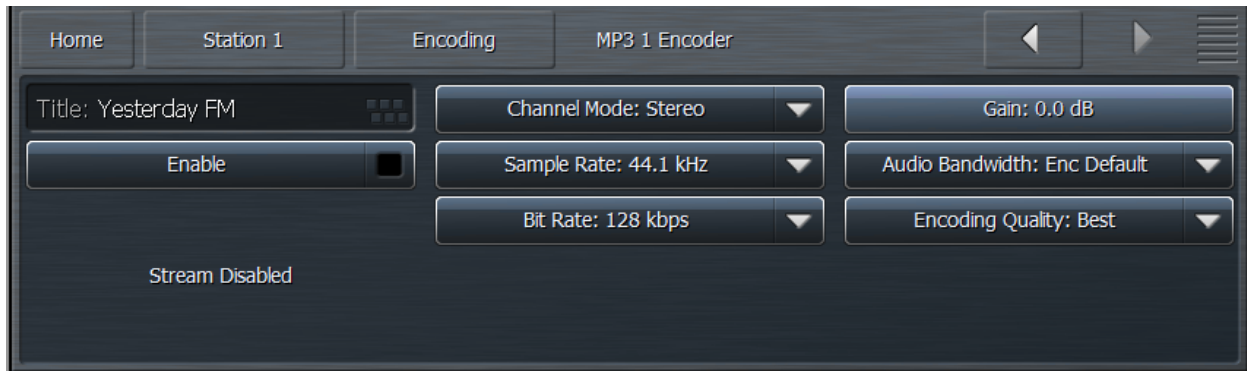
- The **Title** field allows you to enter the stream title displayed to listeners and on public listings, such as shoutcast.com.

- The **Enable** control enables or disables the stream. When enabled, the encoder will continually try to connect to the specified server.

- The **Channel Mode** menu provides a choice between Mono, Stereo, and Stereo Independent. Because the Stereo Independent settings forces the encoder to divide bits equally between the left and right channels instead of allowing it to allocate the bits as needed, we recommend not using this settings unless you have a specific reason to do so.

- The **Sample Rate** menu will either maintain the global sample rate as set in the Configuration App or implement a high-quality re-sample to 8, 11.025, 12, 16, 22.05, 24, 32kHz.

- The **Bit Rate** menu allows you to choose encoding bit rates between 16kbps and 320 kbps. Your choice of bit rate settings will significantly influence the audio quality of your stream. However, you will also need to take other factors into consideration, such as the total amount of bandwidth available for the stream, the anticipated number of users, your maximum upload speed, and the download speed limits of your intended users.

- The **Audio Bandwidth** menu allows you to set the high frequency range of the stream between 3 and 20kHZ. Unless you have a specific reason to set the bandwidth manually, we strongly recommend leaving this control set to Encoder Default.

- The **Encoding Quality** menu provides a choice between Best, Normal, and Low CPU which allows you to balance the tradeoff between higher audio quality and lower CPU usage.

- The **Gain** control determines the final output level into the encoder. If you are encoding to a low bit or sample rate, you will almost certainly need to turn this down to prevent harsh crackling sounds resulting from clipping in the receiving decoder.

MP3 Server Menu

The **MP3 Server** menu allows you to input and set up your server settings.

- The **Title** field allows you to enter the name of the encoder.

- The **Server** field should contain the URL of the server, including the port, in the following format: "streamingserver.example.com:8000"

- The **Mount Point** field (for Icecast 2 servers) should contain the mount point on the server, in the following format: "/example1.mp3" following the server name.

- Enabling the **Allow Public Listing** control sets the "public listing allowed" flag in the stream, as determined by the server.

- The **Server Type** control allows you to choose between Shoutcast and Icecast2.

- The **URL** field is a metadata field whose information is sent to the server to be (potentially) displayed in the listings and player. The URL is typically the station's official home page.

- The **User Name** field contains the user (log-in) name for the server.

- The **Password** field contains the password for the server.

- The **Genre** field is also a metadata field. The type or style of content found on the stream is typically entered here.

- The **Apply Immediately** control determines whether any changes you make to the server settings take effect immediately or when you manually reconnect.

MP2 Encoder Menu

The **MP2 Encoder** menu allows you to enable or disable the stream, adjust its output gain, enter the stream's title, and select the channel mode, sample rate, and bit rate.
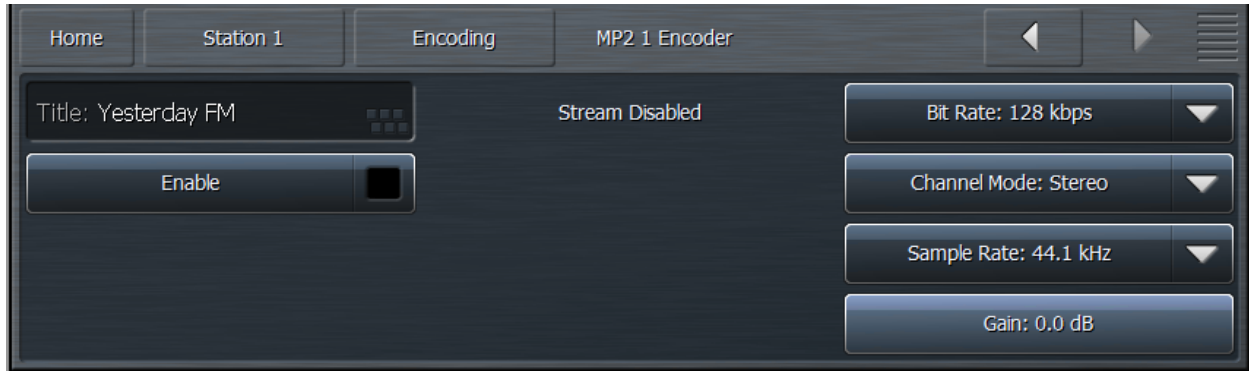
- The **Title** field allows you to enter the stream title displayed to listeners and on public listings, such as shoutcast.com.

- The **Enable** control enables or disables the stream. When enabled, the encoder will continually try to connect to the specified server.

- The **Channel Mode** menu provides a choice between Mono, Stereo, Joint Stereo, and Dual Channel.

- The **Bit Rate** menu allows you to choose encoding bit rates between 64kbps and 384kbps. Your choice of bit rate settings will significantly influence the audio quality of your stream. However, you will also need to take other factors into consideration, such as the total amount of bandwidth available for the stream, the anticipated number of users, your maximum upload speed, and the download speed limits of your intended users.

- The **Sample Rate** menu will either maintain the global sample rate as set in the Configuration App or implement a high-quality re-sample to 8, 11.025, 12, 16, 22.05, 24, 32kHz.

- The **Gain** control determines the final output level into the encoder.

MP2 Server Menu

The **MP2 Server** menu allows you to input and set up your server settings.
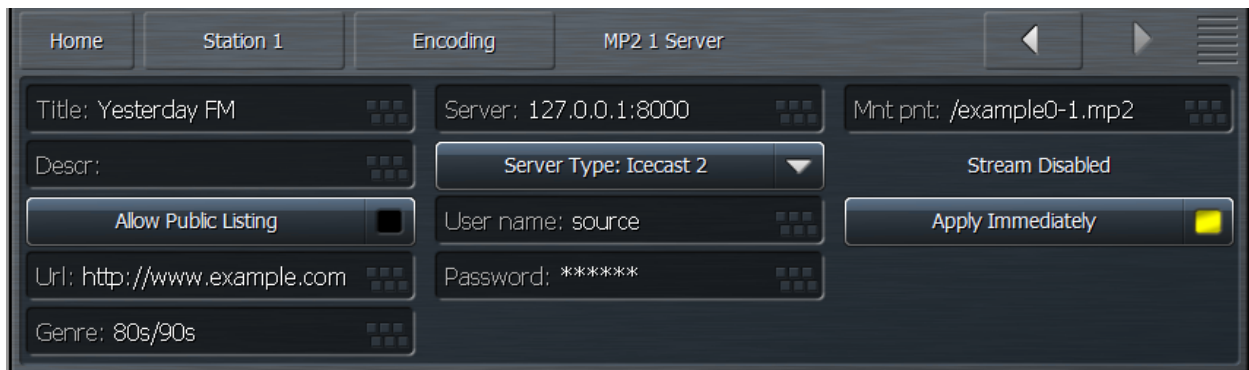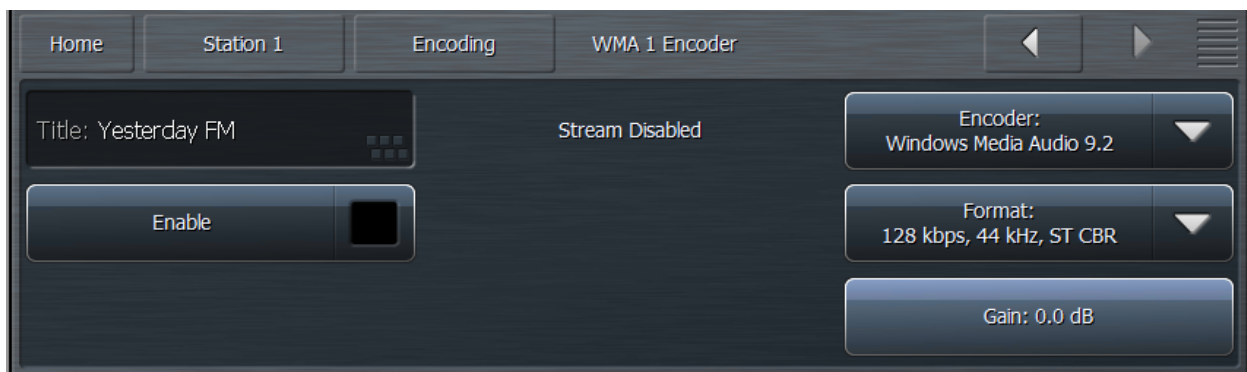
- The **Title** field allows you to enter the stream title displayed to listeners and on public listings, such as shoutcast.com.

- The **Server** field should contain the URL of the server, including the port, in the following format: "streamingserver.example.com:8000"

- The **Mount Point** field (for Icecast 2 servers) should contain the mount point on the server, in the following format: "/example1.mp3" following the server name.

- Enabling the **Allow Public Listing** control sets the "public listing allowed" flag in the stream, as determined by the server.

- The **Server Type** control allows you to choose between Shoutcast and Icecast2.

- The **URL** field is a metadata field whose information is sent to the server to be (potentially) displayed in the listings and player. The URL is typically the station's official home page.

- The **User Name** field contains the user (log-in) name for the server.

- The **Password** field contains the password for the server.

- The **Genre** field is also a metadata field. The type or style of content found on the stream is typically entered here.

- The **Apply Immediately** control determines whether any changes you make to the server settings take effect immediately or when you manually reconnect.

WMA Encoder Menu

The **WMA (Windows Media Audio) Encoder** menu allows you to enable or disable the stream, adjust its output gain, enter the stream's title, and select the encoder type and format.
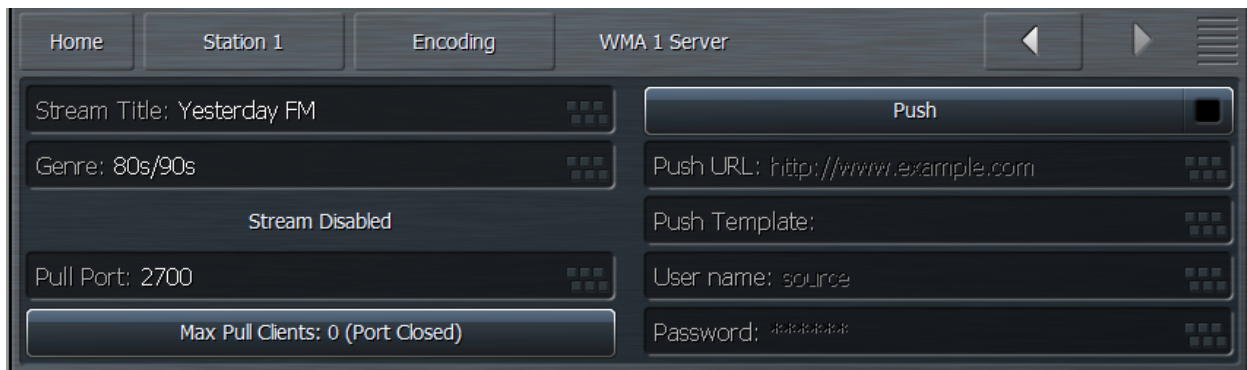

- The **Title** field allows you to enter the name of the encoder.

- The **Enable** control enables or disables the stream. When enabled, the encoder will continually try to connect to the specified server.

- The **Encoder** menu lets you choose between Windows Media Audio 9 and Windows Media Audio 9 Professional

- The **Format** menu allows you to choose a combination of bit rates and sample rates to determine the quality of the audio stream.

- The **Gain** control determines the final output level of the encoder.

WMA Server Menu

The **WMA Serve**r menu allows you to input and set up your server settings.

- The **Title** field allows you to enter the name of your audio stream.

- The **Genre** field is also a metadata field. The type or style of content found on the stream is typically entered here.

- The **Pull Port** field allows you to enter the server's port number.

- The **Max Pull Clients** control lets you set the maximum number of pull clients. We strongly recommend against letting listeners through directly without a server in between.

- The **Push** control enables or disables push capabilities outward toward the server, much like Icecast or Shoutcast. When enabled, fields for providing the Push URL, Template, User Name, and Password are accessible.

Chapter 12
The Station Menu
Miscellaneous Menu

The Station Menu - Miscellaneous Menu

The **Miscellaneous Menu** contains sub-menus that allow a customized station name to be entered as well as a Display Settings menu.
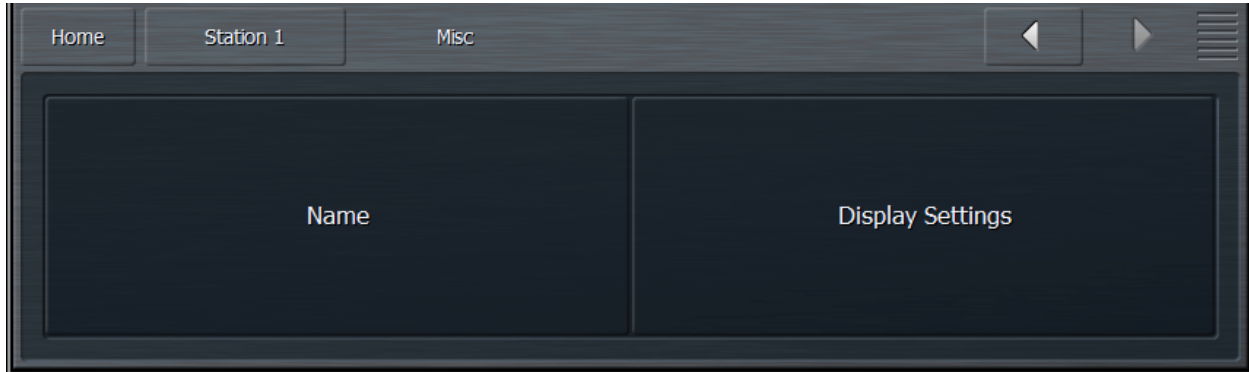
Name Menu

The **Name** menu provides five separate fields for entering the station name in different customized lengths.

Because Omnia.9/XE's user interface is scaleable and re-sizeable, it may not always be possible to display the full name in its entirety. (This is easily observed in the various menu and control descriptions as you resize the GUI window).

As a rule-of-thumb, we suggest keeping the "Shortest" to a length of 4-5 characters and add a few characters more for the next longest name.
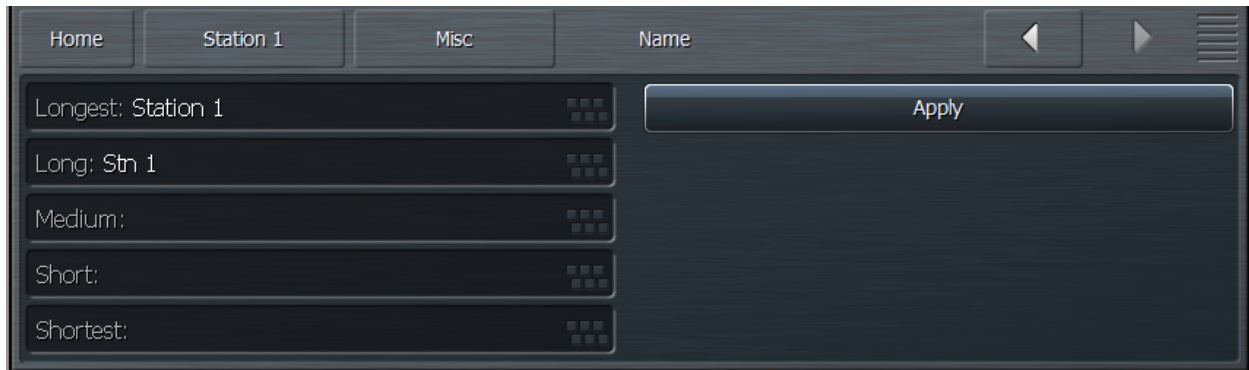
Chapter 13
Factory Presets

Factory Presets

Omnia.9/XE contains a good assortment of factory presets from the Omnia.9 hardware processor created by a handful of folks who are deeply passionate about audio processing and have extensive experience with the product.

Creating a preset takes countless hours of critical listening to a wide variety of music, including hand-picked "problem songs" that contain passages that are known to be difficult for a particular section of any processor to handle. Each preset has been created with a particular sonic goal in mind. Some are relatively simple in structure, while others exploit Omnia.9/XE's unique processing functions and utilize a few "tricks" to achieve a particular sound.

We have deliberately avoided naming presets by format or genre (with the occasional exception) so you'll find no "Urban," "Open Country," "Rock," or "Smooth Jazz" badges here. With conventional naming, many presets that would work quite nicely on your station will be overlooked simply because no one would ever think of trying something called "Hot Country" if they were running a Light AC format that needed some extra punch.

Instead, we encourage you to audition each preset, preferably "on the bench" and not on the air at first, to get a feel for each. Listen for a spectral balance that appeals to you first, and then for texture, without much consideration for loudness. Most of the presets (minus the ones specifically created for maximum loudness) have plenty of "room to grow" in terms of loudness, so if you find something you like but it doesn't seem quite loud enough, don't summarily dismiss it; instead, try it out and use the suggestions in the "Getting the Sound You Want" chapter of this manual to up the loudness.

Finally, we are often asked how best to create a custom preset "from scratch". Since all custom presets are initially based upon a factory preset, we recommend building off of "Reference Settings" since it is the most neutral-sounding factory preset and one upon which many of the factory presets themselves were built.

1981A/O92 – by Jesse Graffam

Designed to capture the spirit of one of the most famous analog processing setups of the 1980's, "1981A/O92" is a 7-band preset that uses tight ratios over threshold in the lowest band for strong consistent bass regardless of source material with looser ratios in the mid-bands to keep vocals and drums open. It also takes advantage of the WB AGC2 in the "Bass Only" mode to deliver an especially solid low end and uses the parametric EQ to "pre-load" the highs which then receive a good dose of fast compression and limiting to deliver a familiar-sounding top end. An excellent, easy-to-listen-to preset for 70's and 80's formats.

Barbeque Sound – by Rob Morsink

If you like a lot of "sizzle" with your music, "Barbeque Sound" brings it on and sends the highs out front to be noticed. The multi-band AGC's use an Infinite:1 ratio under threshold and looser ratios over threshold, providing higher density in lower-level audio and more openness in louder sounds. Fast multi-band release times offer up density and spectral consistency while the WB AGC-2 operating in the "Bass Only" mode gives low-end transients like kick-drums some extra punch.

Big Bottom– by Leif Claesson

As the name implies, a thundering and broad low-end is the hallmark of this 6-band preset. The strong bass is created by a combination of the parametric EQ, the WB AGC-2 in "Bass Only" mode, and a final boost in the Band Mix. Vocals are pushed a bit farther back in the mix in favor of the strong low end, and Infinite:1 ratios in all bands of the multi-band AGC at all times creates a dense and consistent output, conjuring up the general sound of another un-named but ubiquitous processor known for its bass texture and consistency. If there's bass to bring forward in the music, "Big Bottom" will find it.

Big Iron – by Jesse Graffam

"Big Iron" is a 4-band preset dedicated to the glory days of radio and emulates the characteristics of such notable compressors as the ART VLA and the Dorrough DAP 310. It provides a warm, open, vocal sound and won't attempt to aggressively re-equalize the original spectral balance of the source material. It makes use of Omnia.9/XE's multi-stage AGC sections prior to the multi-band compressors as well as the Gain Reduction Override feature in the multi-band section. It's competitively loud but still punchy.

Celeste – by LeeXS

A fast-acting WB AGC and Infinite:1 ratios over threshold in the multi-band AGC's give "Celeste" a consistent sound on a wide variety of source materials. It's strong, warm, and round bass texture and smooth midrange make it a good choice for smooth R&B or light AC stations who value long-term listening and an easy sound over all-out loudness.

Classical/Jazz – by Leif Claesson

"Classical/Jazz" is a 3-band preset designed to provide gentle wideband and multi-band dynamics control without interfering with the original spectral balance of the music. It is not designed for loudness, as it utilizes a modest amount of slow multi-band gain riding and minimal final clipping. It has a much wider gain range than corresponding presets in most other processors, yielding excellent consistency and listenability at lower volumes (such as an office setting) or in noisy environments. This preset works equally well with classical and jazz programming.

Echoes - by Jim Kuzman

This preset was created with openness and dynamics in mind but despite its mission remains competitively loud on average. Very slow multiband attack and release rates, high multiband AGC target settings, and higher-than-normal Gate and Freeze thresholds prevent soft passages from increasing too much and allow short-term dynamics to pass through with plenty of punch. This makes it a good choice for stations who play World or New Age music or for any format where long-term listening is a priority.

Empire State of Processing – by Leif Claesson

Built to answer the question "How loud can a 3-band preset be?", "Empire State of Processing" relies more on the final clipper to deliver competitive loudness while employing slow attack and release rates and low ratios in the multi-band section for a more open feel. Vocals are warm and full, balanced by a crisp high end.

Eruption 2.0 – by Leif Claesson and Johnny Lundholm

Deep, always-present bass compliments of fast attack and release rates in the lowest band and a touch of intentional wideband "breathing" make the dynamic and punchy preset a great choice for techno, electronic, or highly-produced urban music. "Eruption 2.0" leans heavily on the final limiter to deliver plenty of loudness on-air.

French Kiss – by Leif Claesson

A 6-band preset reminiscent of the type of processing favored in France a decade or two ago, "French Kiss" offers up lots of clean, distortion-free bass and uses the WB AGC2 compressor to intentionally and noticeably pump the audio when the bass kicks in. While less suited for typical commercial formats, it pairs very well with electronic dance music.

Helix – by Jesse Graffam

"Helix" is a 7-band preset that sports a smiley-face EQ curve to deliver a solid low end and sparkling highs, but still provides a nice open mid-range sound. This preset uses the maximum amount of multi-band range, an Infinite:1 multi-band ratio, and a fair amount of high-frequency limiting to deeply re-equalize and re-balance virtually any input.

Imagine Hot Mid – by Rob Morsink

This bright and loud 3-band preset puts and emphasis on mids and highs and employs a fast release time in the WB AGC-1 compressor to keep levels consistent going into the multi-band AGC section which uses light ratios over threshold. A great preset for making vocals shimmer and for bringing out all the details in the upper frequencies.

Jill FM – by Jesse Graffam

A 7-band preset inspired by the eclectic programming of the "Jack" formats, "Jill FM" provides excellent vocal and lyrical clarity and a super-smooth mid-range. This preset uses a tight ratio on the AGC but loose multi-band ratios and heavier multi-band limiting. Carefully designed not to over-process the bass or high end, this is a great choice for stations with a high female demographic or with long-term listening as a goal who still need to maintain a competitive degree of loudness.

Motor City – by Jesse Graffam

A 7-band preset that unapologetically delivers an in-your-face rock sound, "Motor City" gets a little edgy in the mid-range but is never harsh. This preset makes use of Omnia.9/XE's Wideband AGC2 after the multi-band section, and while it uses a gentle multi-band ratio below threshold, it makes extensive use of tighter ratios in all bands via the Gain Reduction Override feature.

New York – by Jesse Graffam

One of Omnia.9/XE's loudest factory presets, "New York" uses 6-bands of fast multi-band release times, lots of multi-band limiting, low multi-band AGC thresholds, and a generous amount of clipping to burn a hole in the dial with a very dense, compressed sound. A very generous bass boost from the parametric EQ creates a loud, harmonic-rich bass but never forces its way into the mids or highs. If you want maximum loudness, this is a good starting preset that can be backed down in several ways as needed to trade off a bit of that loudness for a cleaner sound.

Northridge HD – by Jim Kuzman and Leif Claesson

"Northridge HD" is a 7-band preset that makes use of the Wideband AGC2 compressor in "Bass Only" mode to deliver lots of bass punch. Loudness is built relying more upon the dynamics section rather than the final limiter, specifically through the use of the Wideband AGC1 compressor and the multiband limiters. It is also one of the brightest and "coolest" factory presets in terms of spectral balance and can be made louder if needed by increasing the final limiter drive.

NYC Attack of the Camclones – by Jesse Graffam

Slightly louder than "New York," "Attack of the Camclones" provides a huge, tight, dense bass texture, uses fast attack and release times to build density, and has dip in the lower mids to keep male vocals from becoming muddy. It also foregoes any stereo enhancement. A great choice for the CHR format or anyone who wants an ear-catching, prominent sound.

Orlando – by Jim Kuzman

Smooth and laid back but far from being too quiet, this 7-band preset provides gentle re-equalization from the multi-band AGC while still maintaining some of the character of the original audio. "Orlando" is competitively loud when fed more contemporary music, but very suitable for jazz and classical stations that desire better spectral balance than the preset designed specifically for those types of music. There's plenty of room to build loudness as needed with the final limiter without losing the overall feel of this preset.

Passive Aggressor – by Jesse Graffam

A 5-band preset with a split personality, "Passive Aggressor" is relaxed and dynamic in feel, yet it provides a great degree of spectral control and overall loudness by utilizing Omnia.9/XE's "Gain Reduction Override" settings. Looser ratios below threshold and very tight ratios above in the low and high bands coupled with very slow multi-band attack and release times make this preset a great easy-to-listen-to choice for stations that place an emphasis on TSL.

Plutonium HD – by Leif Claesson

"Plutonium HD," the loudest factory preset and the most aggressive of the "atomic" presets, was initially designed to satisfy the specific needs of a large European broadcast group who not only wanted large quantities of bass, but plenty of bass punch and minimal distortion as well. When processing for loudness, there's little (or no) room left for bass. Heavy bass clipping makes some room, but at the expense of punch. "Plutonium HD" solves this dilemma by employing the WB AGC2 compressor in the "Bass Only" mode and incorporating the sidechain delay feature. If your needs call for lots of bass and loudness, "Plutonium HD" delivers.

Point Blank – by LeeXS

Slightly more dense and loud than "Jill FM," "Point Blank" is a 7-band preset that leans more on the multi-band limiters than the final clipper to build loudness. Bright but always clean and well-balanced, this preset delivers good source-to-source consistency without sounding overly compressed or processed. An excellent choice for a light AC station that wants long-term listening but doesn't want to get too quiet, or for a CHR station that doesn't mind trading a bit of loudness for dynamics.

Quintessence - by Sam Sergi

This 5-band preset features full, consistent bass and a sparkling high end. Mids are pushed further back in the mix and loudness is made a priority over openness and punch. Fast multi-band AGC settings ensure spectral consistency regardless of the balance of the source material, making it a good choice for formats where the music spans multiple decades.

Radium – by Leif Claesson

Derived from "Uranium-238", this member of the "atomic" family of presets dials back the limiter drive and loosens up the multiband ratios a bit. The result is a slight loss of loudness but a considerably more open and dynamic sound. That said, this is still not a quiet preset. If you find yourself needing just a bit of extra loudness but don't want to lose dynamics, "Radium" is a great place to start.

Reference Settings - by Leif Claesson

"Reference Settings" is the most neutral-sounding factory preset but that does not diminish its suitability for a wide variety of formats. It is the default preset for Omnia.9/XE It has been used with only slight modifications on active rock stations in very "loud" markets with great success. If you have your eyes set on creating your very own custom preset for your station, this is the preset with which we recommend you begin your masterpiece as it is straightforward and doesn't include any little tuning "tricks" to create a particular effect.

Rustonium – by Leif Claesson and Jesse Graffam

"Rustonium" is a hybrid of two past presets – Jesse's "Rusticity" and Leif's "Plutonium". The bass is always present but punchy, while the high end is bright and prominent but never shrill or edgy. The multi-band attack and release times are relatively slow, but this 7-band preset makes extensive use of Omnia. 9/XE's progressive release control to allow plenty of gain control in each band without the fear of any band getting "stuck" with too little gain. This preset also places the Wideband AGC2 after the multi-band section to build some extra loudness just before the final limiter. A great "as-is" out-of-the-box preset for nearly every contemporary format.

Stacked Boxes – by Jim Kuzman

Remember back when the dial-dominating air chain was made up of a rack full of analog gear? Say, one box for compression and leveling, another pair of boxes with lots of LED's providing 4-band compression, and a program/peak limiter (maybe from the same folks as the compressor/leveler) – all driving a final limiter/clipper with edge meters and a key-locked front panel? We do too. "Stacked Boxes" pays homage to the not-so-distant past of analog processing.

Stacked Boxes Plus! - by Jim Kuzman

Offspring of "Stacked Boxes," the "Plus!" version uses six bands of multiband compression and limiting instead of four, providing a similar overall feel as the original but with more refined mids and better source-to-source spectral control.

The Blitz – by Jim Kuzman

Living proof that loudness and high-quality audio are no longer mutually exclusive qualities, this 6-band preset hands over tight, solid bass, intelligible mid-range, an airy high end, and a wide stereo image. Designed to provide the maximum degree of loudness but never at the expense of audio quality, "The Blitz" is a great choice for classic and active rock formats with plenty of room for an extra dB or two on the clipper drive for those who need more loudness. (Thanks, Bill!)

Tokyo – by Jesse Graffam

Born of "Passive Aggressor" and created in homage to the "classic" sound of previous Omnia processors, the 7-band "Tokyo" delivers detailed transients and percussion without sacrificing average loudness. The mid-range is full and round with increased vocal intelligibility, a sound that translates well to smaller speakers and earbuds as well as higher-quality systems. A tightly-controlled low end ensures the bass always shows up for the party.

Trinity – by Leif Claesson

Inspired by the sound of the classic Omnia.3 FM Turbo, this 3-band preset is dynamic and punchy with an open mid-range and a more classic "Omnia" sound. "Trinity" doesn't use any parametric EQ or stereo enhancement, runs with a very light ratio in the multi-band section, and presents itself very transparently on the air, making it suitable for nearly any format.

Upsidasium – by Leif Claesson

The smoothest and most open of the "atomic" presets, "Upsidasium" is still relatively loud but thanks to very loose multiband ratios, is also extremely open-sounding. Drums and other transient sounds cut through the mix, vocals are dynamic, and the upper high end delivers that "lighter than air" sound (here's looking at you, Denny!). If your situation permits giving up the last dB of loudness in favor of a sound that's easy to listen to long-term but you don't want to sound underpowered, "Upsidasium" offers the perfect compromise.

Uranium-238 – by Leif Claesson

The second loudest of the "atomic" presets, "Uranium-238" is slightly less radioactive version of "Plutonium FM". It has a similar sound to its parent preset, but does not utilize the Wideband 3 compressor after the multiband section and employs less final clipping. A good choice if loudness is still your goal but "Plutonium" feels too over-the-top.

Zenith – by Jesse Graffam

How much processing can you get away with before things start to sound processed? The 7-band "Zenith" was created to answer that question. Always open and natural sounding in exchange for overall spectral and loudness consistency, this preset lets the music be what it was intended to be – with just a little push in the right direction.

Chapter 14
Using the NfRemote Client Software

Using the NfRemote Client Software

Understanding the NfRemote Connection Window

A basic explanation of installing the NfRemote software and making an initial inbound connection to Omnia.9/XE was provided earlier, but there are additional features and menu choices available to make remote connections as easy as possible. As you can see, it is possible to connect to any number of Omnia.9/XE host PC's (or Omnia.9 processors) individually or concurrently.
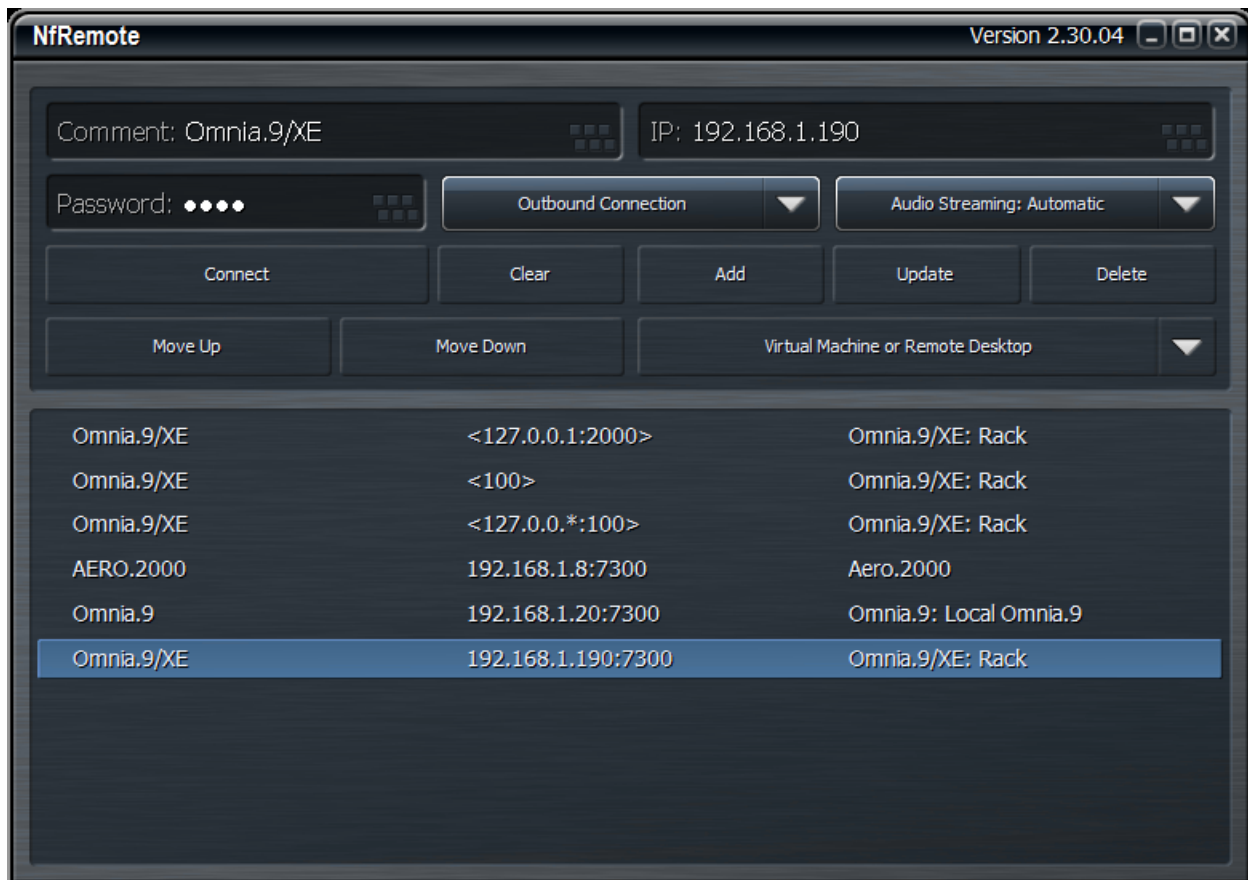
While the client software itself is not a terribly resource-intensive application, it does place some demands upon your local network. Because the client allows you to make "live" adjustments to Omnia.9/XE and observe its meters (as well as other displays such as the oscilloscope) in real time, and on networks with sufficient bandwidth play audio back through the remote computer, it is important to have a robust Ethernet or Wi-Fi connection to the between the client and the host PC.

- The **Comment** field allows you to enter a "friendly" name for each device to which you plan to connect.

- The **IP Address** field should contain the IP address of the Omnia.9/XE host computer.

- The **Password** field should contain the same password established in the Omnia.9/XE application.

• The **Connection Type** dropdown menu determines whether the connection is **Outbound** (NfRemote initiates connection to Omnia.9/XE) or **Inbound** (Omnia.9/XE listens for an incoming connection request from an instance of NfRemote). Note that the IP addresses of inbound connections are enclosed by chevrons.

• It is possible to stream audio from the Omnia.9/XE host PC to a remote PC using NfRemote. The **Audio Streaming** dropdown menu enables this feature and determines the type of data compression used to deliver the audio as well as the oscilloscope, RTA, and FFT display information.

  • **Automatic** chooses the best audio format based upon connection speed.

  • **Off** completely disables audio streaming and the client-side oscilloscope, RTA, and FFT displays.

  • **Force Lossy** enables lossy streaming even on connections that are on the border of being too slow and forces lossy streaming even on fast connections.

  • **Force Lossless** enables lossless streaming even on connections that are too slow. This choice is best reserved for local networks with sufficient bandwidth to support it.

• Clicking on the **Connect** button will initiate a connection to Omnia.9/XE whose IP address is currently displayed.

• The **Clear** button removes all information from the Comment, IP Address, and Password fields.

• The **Add** button carries the current Comment, IP Address, and Password information and carries it to the bottom of the window where it is saved for easy recall later.

• The **Update** button refreshes the list of saved connections.

• Clicking on the **Delete** button will remove the selected connection from the list.

• The **Move Up** and **Move Down** buttons will move the selected connection up or down respectively in the list so that you can sort them in any order.

• The **User Interface Pointing Device** dropdown menu offers several choices for optimizing NfRemote depending upon the type of device you are using.

  • **Local Mouse or Touchpad** should be selected if you are running NfRemote on a local PC or laptop.

  • **Virtual Machine or Remote Desktop** is the correct choice if you are connecting to the computer running NfRemote from another computer. This includes software like Parallels, VMware, Team Viewer, GoToMyPC, Remote Desktop Connection, or a VNC client. Wacom Digitizer users should also selection this option.

  • **Local Mouse (Touch Simulation)** allows you to use the finger-friendly layout of the Touch Screen option but with a mouse. It can also be used instead of the Virtual Machine or Remote Desktop option.

  • **Touch Screen** provides a finger-friendly layout optimized for touch screen devices like tablets or all-in-one PC's with a touch interface.

- **Touch Screen Presentation** operates just like Touch Screen but with a large cursor, making it ideal for situations when you are using a projector for demonstration or instructing a group on how to use NfRemote.

Initiating an Outbound Connection from Omnia.9/XE

Normally, an instance of NfRemote running on a remote computer initiates the connection to the host PC running Omnia.9/XE ("inbound" connection). However, there are some situations where it is more appropriate for Omnia.9/XE to initiate the connection to an instance of NfRemote on a remote computer ("outbound" connection).
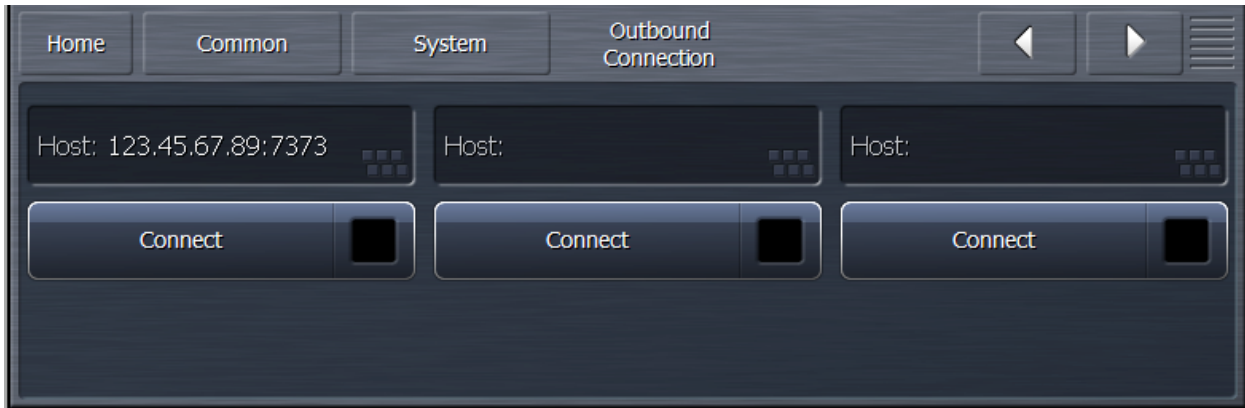
One example of how this can be used is for technical support. By creating an outbound connection, you are connecting to the support engineer's NfRemote computer which in turn allows them to control your Omnia.9/XE.

In such a situation, the support engineer would provide an IP address (or host name) and a port number separated by a colon (123.45.67.89:7373 for example). Because IP addresses are commonly dynamically allocated, using a host name is the preferred convention.

Client-Only Controls

As mentioned earlier, the remote interface looks nearly identical to Omnia.9/XE's app window display. If you have already familiarized yourself with its controls, you should feel right at home operating it remotely. Therefore, only those controls or features unique to the client software will be explained here.

Perhaps one of the most valuable capabilities of the client software is speaker calibration. Since your Omnia.9/XE already contains a built-in pink noise generator and an RTA, adding your own calibrated microphone to the setup will allow you to perform a calibration of any speaker system.

You may wonder why we consider this to be such an important exercise when listeners aren't going to be hearing your stream on calibrated monitors. Great question! Yes, listening to your station's processing on speakers of different qualities in a wide variety of listening environments is important to get a "real world" feel of how things sound – but we feel strongly that having at least one set of calibrated "reference" monitors is absolutely critical to making educated decisions as you adjust your processing. If you make adjustments based upon what you hear on un-calibrated speakers every decision you make will be influenced by the inaccuracies of those speakers in that room. Details of how to perform such a calibration are found later in this chapter.

Joining the Home screen is an additional **Client Audio** menu which contains additional menus for Client Audio Output, Client Microphone Input, and Client File Recorder.

Additionally, the **Save Preset** and **Load Preset** menus in the client software have the additional capabilities of being able to save a preset to the location of your choice on the remote PC itself, or load a preset from the remote PC to the Omnia9/XE host PC. This includes flash drives and external USB drives, which provides an excellent way to back up any custom presets or easily duplicate them for use on another computer running the Omnia.9/XE software.

When using the **Load Preset** function, be aware that the preset will be loaded to the **Current Preset** position and will therefore *immediately go on the air*. Additionally, in order for the preset to appear in the list, you must save it using the **Save Preset** function.

Client Audio Output Menu

The **Client Audio Output Menu** contains additional controls similar to those in the Speaker Output menu of the Omnia.9/XE if you specified a WDM Speaker output device in the Configuration App. There is also a TCP Link Status window to provide information about the remote connection.
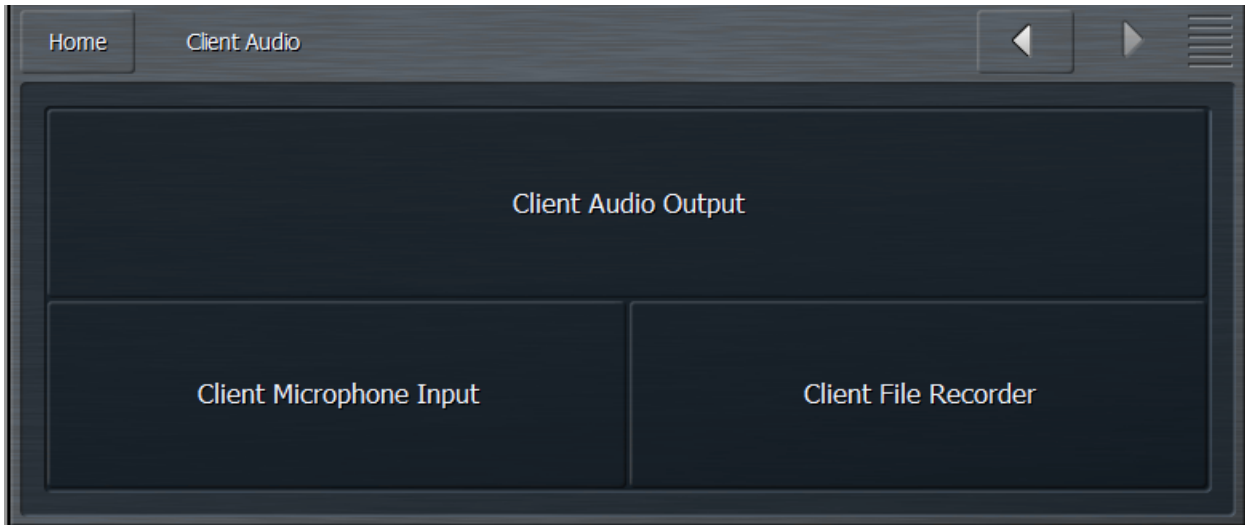
Patch Point Menu

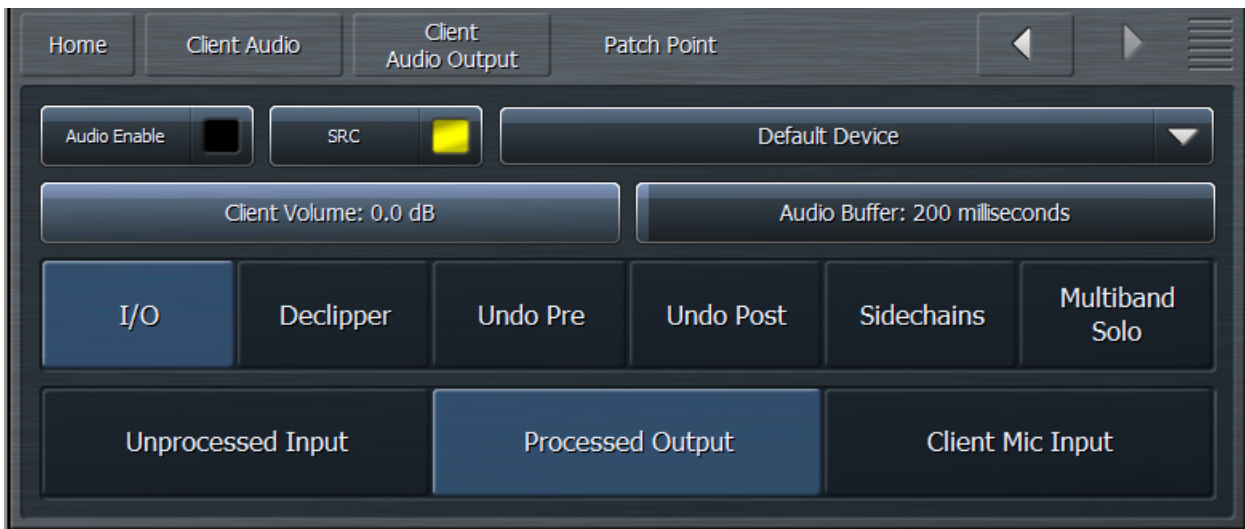The **Patch Point Menu** provides additional controls for enabling Client Audio and SRC, choosing an Output Device, adjusting Client Volume, and setting the Audio Buffer. Various patch points from within the Omnia.9/XE processing core for the selected station are available along with a Client Mic Input.

- When selected, the **Audio Enable** feature allows processed audio to from various patch points within the Omnia.9/XE to be streamed directly to the client PC. Keep in mind that audio streams require bandwidth, as does displaying patch points on the oscilloscope, FFT spectrum analyzer, or RTA. During handshaking, the remote control software measures the available network bandwidth. If you are on a local network, you will get PCM (NICAM) streaming. If you connect over the internet with sufficient bandwidth, you will get data-reduced streaming.

- Enabling the **SRC** control lets the client software dynamically adjust the sample rate of the client playback. If you are experiencing interruptions or dropouts to the client audio stream, enabling SRC will result in a steadier connection, though at a slight cost to CPU usage. Regardless, the client software will automatically determine whether to stream client audio in a linear (uncompressed) PCM format or in a data-reduced format, which, while compressed, can still provide excellent audio quality.

- The client software recognizes any audio output devices (speakers, headphones, sound cards) installed on the remote control PC, and makes them available in the **Output Device** drop-down menu.

- The **Client Volume** slider controls the volume of the selected output devices on the remote PC. This control is duplicated on the top menu of the client software where it is available at all times.

- Depending upon the bandwidth available on your local network, it may be necessary to buffer incoming audio to the client to avoid dropouts by adjusting the **Audio Buffer** control. The range of this control is between 0 and 10,000 milliseconds (10 seconds). The meters on the remote PC will automatically adjust so that they are always synchronized with the client audio as long as client audio is enabled. In addition, you can use the meter delay control in the display section to delay meters further to compensate for any delay occurring after the client PC (such as when using Bluetooth speakers).

- The **TCP Link Status** window provides information about the connection status between the remote PC and Omnia/9XE, including Current Ping time in milliseconds, Link Bandwidth in KB/sec, the Audio Format (PCM or Vorbis) of the stream, and Send and Receive rates.

Client Microphone Input Menu

The **Client Microphone Input Menu** allows you to pick which microphone you'd like to use, a parametric equalizer to create custom EQ curves for individual mics, and the ability to save and recall custom mic presets. There is also a duplicate button to the TCP Link Status window and the Display Settings menu for convenience.

Input Device Menu

The **Input Device Menu** provides a list of available microphones, including built-in mics and any outboard microphones you may have plugged into the PC running the remote interface software. If you plug in an outboard mic while this window is open, you may need to tap the "Refresh List" button in order for the software to recognize the addition.

Load Preset Menu

This **Load Preset Menu** operates in a manner similar to the **Load Preset** menu in other sections of Omnia.9/XE. However, the client software offers an additional **Load From** button that allows you to call up a preset file from the remote PC, including from a flash drive or external USB drive.

Save Preset Menu

Just like the **Load Preset** menu, this operates like its counterparts elsewhere in Omnia.9/XE. The **Save To** button allows you to save your presets to a location on the remote PC, including a flash drive or external USB drive.

Client File Recorder

Just as it is possible to record audio from various patch points within the processing chain on the actual Omnia.9/XE itself, you can also record audio streamed to the client PC. The recorders in the client software work in exactly the same way.

Speaker Calibration

As mentioned earlier, taking the time to calibrate at least one set of speakers to act as your reference monitors is time well spent. In order to perform a speaker calibration, you need four tools:

- A calibrated microphone to capture the audio.

- A pink noise generator to provide audio for measurement (pink noise having an equal amount of "noise power" in each octave).

- A real-time audio analyzer (RTA) to visually display what the microphone is hearing.

- Some means by which to correct for speaker and/or room inaccuracies (such as a parametric equalizer).

Omnia.9/XE provides three out of the four tools – the pink noise generator, the RTA, and the EQ. All you need to add is the calibrated mic.

Choosing a Calibrated Mic

It is important to understand that you *must* use a *calibrated* microphone designed for sound measurement purposes to perform a proper calibration. Studio or recording mics – even the high-end, high-dollar ones – are of no value when calibrating a system.

You can spend thousands of dollars on a scientific-grade, high-precision calibrated microphone, but the good news is – you don't have to. Even the least expensive models will provide good quality results and are worthy of consideration.

While we are not in the business of selling microphones, nor do we have any interest in promoting any brand or model over another, we can tell you that Behringer ECM8000 and DBX RTA-M work quite well and can be purchased at local music stores and from various on-line vendors for around $100. If you prefer to connect your mic via USB or Firewire, several manufacturers (M-Audio, E-Mu, and Blue come to mind) offer XLR to USB and XLR to Firewire converters starting at under $50).

Setting Up the Mic

Place the microphone as close as possible to the position you will occupy when monitoring so that it hears what you will hear. In cases where you are calibrating two of the same speakers, and both are placed in similar locations in the room, you can calibrate both the left and right channels simultaneously. If you find yourself in a situation where the speakers will be affected differently by their individual placement in the room – one is suspending from the ceiling in a corner and the other is sitting on a shelf, for example – or one is significantly closer to your monitoring position than the other – we recommend performing an individual calibration for each speaker.

Setting Up the Client Software for Calibration

For the example below, let's assume that you are calibrating your studio monitors, which are being fed audio from the Speaker Output portion of Omnia.9/XE's Monitor Outputs. We will also assume that you are in a situation where you can use the same calibration settings for both speakers. By default, the parametric EQ section is set up in the "common" mode, which means changes made to the left channel will be mirrored in the right channel.

Connect the Client PC and the Calibrated Mic

The first step to calibration is making sure you have a remote connection to your Omnia.9/XE.

Once connected, plug your calibrated mic into the remote PC, install any necessary software or drivers, and establish that it is working properly in Windows.

Set Up RTA and Oscilloscope Display Windows

After microphone installation and setup are complete, you will need to make sure you have the RTA and the oscilloscope set up in a Display Window.

If an RTA is not already visible, select any of the current display windows to bring up a Display Settings menu and set up one of the windows to monitor the Client Mic input (for example, Display 3, RTA, I/O, Client Mic Input).

1/3 octave resolution will provide more of a "big picture" overview, while the 1/6 octave will identify more specific peaks and valleys. We suggest an Average setting of 40 and a Target Range of 2.0dB. You will need to adjust the Target and Range controls for your individual situation once you have pink noise running through the speakers at a reasonable volume.

We also recommend setting up an Oscilloscope menu in the same manner so that you can watch for any clipping that may occur should you need to make any significant boots to a particular part of the audio spectrum during calibration. If you see (or hear) any clipping during the process, turn down the **Master Gain** control in the Main 1 menu.

Feeding Pink Noise To the Speakers

From the Home menu, select **Client Audio**, **Client Microphone Input**, then **Input Device** from the **Home** menu of the Omnia.9/XE client software on the remote PC. Your calibrated mic should be available in the device menu. You may need to refresh the list to make it visible.

Return to the **Home** menu, select **Monitor Outputs**, then **Speaker Output**, then **Main 1**. Turn down the Speaker Volume to a fairly low level, then enable Pink Noise for the left channel. Turn up the Speaker Volume enough to bring the level of pink noise well above the noise floor of the room, and to a level that approximates the volume at which you would normally listen to program material when adjusting your processing. As you adjust the overall volume, you should see the RTA display respond.

Final Adjustments to the RTA

Once you have established an overall pink noise level, adjust the **Range** control so that you can see the entire spectrum easily on the display, normally in the vicinity of 70 – 80dB.

Adjust the **Target** control so that the band with the lowest level (not counting very low or very high frequencies that are clearly beyond the ability of the speaker to reproduce) is situated at the top of the bright red target window. This will allow you to reduce, rather than boost, most of the frequencies with the parametric EQ.

When reading the RTA, it is important to know what peaks and dips are actual representations of "problem areas" with the speaker/room combination and which are microphone placement anomalies. If you move the microphone just slightly and the problem area magically corrects itself, you're seeing such a placement-skewed reading. A little experimentation will quickly reveal this. Once you have finalized the mic placement, it's time to begin correcting things with the PEQ.

Using the Parametric EQ to Smooth Things Out

Methods of equalizing speakers in their rooms vary, but generally speaking, using the fewest number of bands of EQ to achieve the desired result is preferred. Cutting (reducing) frequencies is usually preferred to boosting (increasing) them, although sometimes boosting a certain range is not easily avoided.

Remember that there are a total of 12 available bands of EQ available to you – 6 in the **EQ1** menu and 6 more in the **EQ2** menu.

We assume that if you've gotten this far, you have a general idea of how the controls of a parametric equalizer work, but as a quick review:

- The **Gain** control determines how much the specified frequency range is amplified or attenuated, expressed in decibels.

- The **Bandwidth** control determines how wide or narrow a range of frequencies will be affected, measured in octaves.

- The **Center Frequency** control determines what frequency will lie in the middle of the adjustment curve, expressed in Hz.

You will be able to identify where your speakers start to roll off in terms of frequency response (especially on the low end) pretty quickly. You can't overcome pure physics with an equalizer, and small speakers will never reproduce bass with the same authority as larger ones, so there's no point in trying to give a 15dB boost at 30Hz to small monitors in the hopes that you'll get the equivalent of a free subwoofer. (What's frequently more effective with bass-shy speakers is taming the often over-boosted mid-bass that manufacturers dial in in the hopes that you think you have more low end than you do).

As you study the RTA, we recommend identifying and correcting any glaring peaks or valleys over a relatively narrow ranges of frequencies first to take care of any "hot spots." Then, work your way out to making broader changes to the overall curve.

Chasing Your Tail

Speaker calibration can be a frustrating process, especially if this is your first go-around with it. Sometimes it's like squeezing a balloon: When you squish one end, the other end puffs out. So it goes with calibration. Make a boost to one frequency, and the surrounding frequencies go up, too.

Experimenting with the center frequency and bandwidth controls will help, as will a good dose of patience and allotting enough time to do it up right. Your efforts will be rewarded when you play music through your calibrated setup for the first time and wonder if someone swapped out your tired old monitors with something new and much more expensive!

Saving Your Calibration Settings

You can save your calibration settings in the form of a preset by going to the **Save Preset** menu and typing in a friendly name for your file, preferably one that uniquely identifies the speakers and location ("JBL 4312 FM Air").

Tweaking After Calibration

You may notice that while your speakers sound significantly better than they did before you calibrated them, and they are technically much more accurate, they sound a little thin in the low end and a little dull in the highs because of the way the human ear perceives sound at the far ends of our hearing range.

We suggest that instead of making changes to the EQ sections, you use the **Loudness 1** and **Loudness 2** menus to make subjective adjustments. These menus each give you three additional bands of fully parametric equalization.

If desired, they also allow you to adjust at what point on the Speaker Volume control the changes take affect (since our ears are less sensitive to certain frequencies at lower volume levels).

- The **Minimum** slider on each EQ band sets the point on the volume control where the EQ has no effect on the sound.

- The **Maximum** slider sets the point on the volume control where the EQ in that band is in full effect (boosted or cut to whatever gain level you have set).

For example, if the **Minimum** slider is set to -20dB, the **Maximum** is set to -10dB, and the **Gain** is set to minus 5.0dB, equalization will begin to take effect when the volume control is set to -20dB and will be attenuated by 5dB by the time the control is set to -10dB.

If the minimum and maximum values are the same, the loudness controls function as a normal parametric equalizer independent of the volume setting.

Chapter 15
Getting the Sound You Want

Getting the Sound You Want

Achieving a High Quality Sound

When all-out loudness isn't your primary processing goal (and it shouldn't be with streaming audio), here are some suggestions to help you achieve that goal:

- Reduce the Final Limiter Drive in 0.25dB steps until you reach the point where the quality no longer dramatically improves but you start to lose loudness - then go back up a notch. It's surprising how much a small change can open up your sound.

- Reduce the level of the Band Mix section, especially if you have increased the output of one or two particular bands to achieve a particular sound.

- Raise the threshold of the Multiband Limiters to keep the audio from hitting the limiters as hard or as often. (You may need to lower the Band Mix level to keep from driving the final limiter too hard when making this adjustment).

- Lower the targets of the Multiband AGC compressors

- Use slower attack and release speeds in the Input AGC, Wideband AGC, and Multiband AGC sections.

- Use "looser" ratios in the Input AGC, Wideband AGC, and Multiband AGC sections.

Spectral Consistency & Equalization

Stations often want to create their own "signature sound" for their streams that present a spectrally consistent output without regard to the frequency balance of the original source material.

There are basically three ways to alter the spectral balance of your sound with Omnia.9/XE, and each provides a different end result: The **Parametric Equalizer**, the **Multiband AGC Targets**, and the **Band Mix** control. Adjustments to each section can produce somewhat similar or very different outcomes depending upon a variety of factors.

- The **Parametric EQ** is a very versatile and powerful means by which to shape your sound. You can set up a shelving EQ to create low bass "slam," set up a broad, gentle cut to the high-mids to smooth out horns or female vocalists, or add some "air" or "sparkle" to your sound by adding a bit of a boost to the very top of the spectrum.

  Because the Parametric EQ is located fairly early in the processing chain and ahead of the Multiband AGC and Limiters, these processing sections are likely to make some attempt to compensate for large changes in the spectrum. However, because the crossover slopes of the multiband processors are relatively broad, the Parametric EQ can have more impact than you might expect.

  Finally, remember that EQ is a *static* process. That is, it increases or reduces gain to a particular part of the spectrum irrespective of program content. In other words, if you apply 8dB of bass boost it is applied to *all* incoming audio whether it is bass shy or bass heavy.

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- Adjusting the **Multiband AGC Targets** can have a very significant effect on the spectral balance of your sound as well but there are important factors to consider when adjusting these controls. For example:

  - Raising the target of a particular band can make the audio within that band louder (providing increased gain) but this can also slow down the release rate if Progressive Release is employed, as Progressive Release is driven by the current amount of gain in that band.

    The opposite is true when lowering a target. Doing so can make a particular band softer (providing decreased gain) but can result in an accelerated release rate when Progressive Release is used.

  - Since raising the target increases gain, doing so means that audio in a particular band may not be increased sufficiently if the source material is lacking in that part of the spectrum. For example, if the source material lacks brightness and the target of the top bands is raised too much, there may not be enough "room" in those bands to sufficiently increase their output.

    The result may be spectral inconsistencies from source-to-source. If creating a consistent spectral signature is important, this may be problematic. If, on the other hand, your goals lean more toward preserving the spectral balance of the original material, this could be advantageous.

- Making adjustments to the **Band Mix** will also influence the spectral balance, but since this is the final point in the processing chain before the Final Limiter, any peaks that result from adjustments made here will be dealt with only by this aggressive processing stage. While it is not always a bad idea to increase the output level of a particular band, depending upon the sound you are trying to create, you may have better results by decreasing the output of the rest of the bands by that amount instead. Or, you may instead wish to make the adjustments to suit your tastes and then reduce the overall Band Mix level.

The Multiband AGC Target and Band Mix controls can be used together to help fine tune your sound.

For example, let's say you are using a 6-band preset and you want to create a more dense-sounding mid-range. By lowering the targets of bands 3 and 4, you will drive those bands further toward a state of gain reduction (and increase their release rate when Progressive Release is utilized) but this can also cause them to become softer in the mix if Progressive Release is not used or the source material doesn't have significant content in those bands. You can compensate for this loss by increasing the gain of those bands in the Band Mix.

Or, let's say you want an open and dynamic low end, and have raised the targets of Bands 1 and 2 to achieve less compression but now find that the bass seems too prominent in the mix. You can decrease the gain of these bands in the Band Mix section to restore a more balanced sound.

Chapter 16
The Importance of
High Quality Source Audio

The Importance of High Quality Source Audio

There is a saying in the technology world (and in the broadcast plant when it comes to audio) that has never been truer than it is when applied to streaming audio: "Garbage in, garbage out".

We have all heard audio streams that sound "swishy" or "underwater". This is especially true of low-bit rate encoded streams, but it's not only the bit rate of the stream that contributes to this sound: It is often made worse by the source material itself.

Imagine making a copy of an original document on an office copier; the duplicate looks pretty much like the original, but not quite. Now imagine making a copy from the copy, then another copy from THAT copy, and so on. Pretty soon you have a document that doesn't resemble the original at all.

Source audio that has been encoded multiple times (or with poor quality encoders) similarly degrades with each subsequent encoding. This is called "cascading" data compression.

To put this in an audio perspective, consider that an MP3 file encoded at 128 kbps stereo is usually about 10% of the size of the original, uncompressed, linear audio file. The MP3 files gets to be that small because the encoder discards an enormous amount of information from the original file. Ideally, the promise of so-called "lossy" encoders is that it discards only information it determines we won't miss. The extent to which this is true is the subject of much ongoing debate, with audio purists on one end who claim they can hear the tiniest differences (on their $10,000 speakers) and teenagers with iPhones and cheap earbuds on the other who wouldn't know good audio if they heard it.

The reality for most listeners is somewhere in the middle, which is why we give recommendations elsewhere in this manual about the minimum settings for what we consider "broadcast quality" audio.

But one thing is certain: Since streaming audio uses bit-reduced, lossy encoders for the sake of efficiency in the same way we use these encoders to make manageable music files, **using uncompressed source material is a MUST in our opinion.** Neglecting to do so will either result in audio that has degraded to the point that even the average listening will notice it or being forced to encode at excessively high bit rates which requires more bandwidth (for you on the server side and your listeners on receiving side).

Once upon a time, the case could be made for storing a digital music library in a compressed format because most hard drives were relatively small and the big drives were expensive. At the time this manual was written, a quality 2 Terabyte external hard drive can be purchased for under $100. What can you do with 2TB of space? You can store approximately 3,320 hours of 16 bit 44.1kHz linear PCM stereo audio on it. If you like your information in bite-sized (byte sized?) pieces, that's nearly 57,000 three-and-a-half minute songs!

Chapter 17

The Effects of Audio Processing on Encoded Audio

The Effects of Audio Processing on Encoded Audio

Streaming Isn't FM

Processing for streaming audio is very different than processing for FM. Unfortunately, many broadcasters fail to realize just *how* different it is, and the results can easily be observed in arguably the majority of streams delivered by traditional terrestrial broadcasters.

Unlike FM transmission where loudness is realized at least in part by clipping, the input to a streaming encoder must NEVER be clipped.

At first glance, this might seem like an easy enough goal to achieve: Simply set the output of the streaming processor at such a level as to avoid clipping the input of the encoder. But upon closer examination and with an understanding of how bit rate-reducing (lossy) codecs to their job, this simply isn't enough.

What Encoders Like

Lossy encoders (including the MP2, MP3, AAC, and WMA encoders in Omnia.9/XE) perform best when the incoming audio has short term dynamics, aka "peaks". This is where they tend to hide any audible encoding artifacts. When the content lacks short term dynamics, the artifacts are more readily revealed.

Therefore, it is not only unnecessary to make sacrifices in your processing for the sake of loudness, it is unwise as well.

With all processing - including FM - there is a point of diminishing return where a small amount of increased loudness will come at the price of a large drop in openness and quality. Depending upon the market, it it sometimes necessary for an FM station to walk right on that fine line to stay competitively loud. Thankfully it is possible to step back from it with streaming audio deliver a product that is plenty loud and significantly more enjoyable to listen to than FM radio.

How Listeners Listen

We are not suggesting that you feed a quiet stream. Not at all! But unlike the habits of FM radio listeners who are able to quickly punch up stations with presets on their car stereos, changing streams requires more time. By the time the new stream buffers and loads, any comparative loudness opportunities have been lost (assuming the new stream is within a couple of dB's of the previous one).

The Benefits of Undo

The very widespread problem of source material that has been dynamically over-processed and even clipped during the mastering process is well known.

We have already established that content with short term dynamics is better received by streaming encoders than content with little peak content. We can control the amount of additional processing that we apply, but what can be done about the source material?

Omnia.9/XE's solution uses the same "Undo" process found in the Omnia.9 hardware processor. It is a two-stage process involving de-clipping and multiband expansion. The audio is first examined for clipping. If it is present, the de-clipper intelligently recreates a real peak in place of the flat-topped signal. Next, a multiband expander restores some dynamic range to the signal.

The result is that the dynamics portion of the processor is presented with clean, dynamic audio which can withstand the rigors of normal processing.

There is another significant advantage that Undo brings to the table for streaming: The resulting audio is inherently "encoder friendly". Because the de-clipper portion removes distortion, the encoder can spend its limited bits encoding real audio rather than distortion. And because the multiband expander portion adds short term dynamics, it gives the encoder more flexibility to mask encoding artifacts.

Chapter 18

Choosing an Encoding Format and Recommended Settings

Choosing an Encoding Format and Recommended Settings

Omnia.9/XE provides a choice of MP2, MP3, AAC, and WMA encoding formats. Here we endeavor to explain the merits and pitfalls of each and provide some recommendations regarding bit rate, sample rate and setting output levels in the processing section of Omnia.9/XE.

MP2
Technical Details

The full name for the format we simply call MP2 is actually MPEG-1 Audio Layer II. Though MP2 is a perceptual audio encoder (also known as a "lossy" encoder because it removes information from the original audio signal) it is also a sub-band audio encoder, so compression takes place in the time domain.

Advantages

MP2 is more error-resilient than other formats and more tolerant of "cascading" compression. It delivers exceptional audio quality at very high bit rates (320 or 384 kbps).

Disadvantages

It is not a commonly used format, and there are better choices at lower bitrates for general streaming.

Applications/Summary

The MP2 format the best choice if you are sending a single stream to a service provider who will then re-encode it into other formats or bit rates for distribution.

MP3
Technical Details

MP3 - technically MPEG-1 Audio Layer III or MPEG-2 Audio Layer III - is also a "lossy" encoder, but it is a transform audio encoder. This means that compression takes place in a hybrid form in the frequency domain after it also occurs in the time domain.

Advantages

The main advantage to MP3 is that it is a "legacy" format and can be decoded by the widest variety of players. Beyond that, MP3 offers few if any inherent advantages.

Disadvantages

Compared to AAC, MP3 doesn't make very efficient use of bandwidth. In other words, it requires encoding at a higher bit rate to deliver a similar level of quality.

Recommended Settings

In our view, the minimum encoding bit rate to yield "broadcast quality" audio is 128 kbps stereo. 192 kbps sounds better still but marks the "point of diminishing return", and there is progressively less sonic improvement available in exchange for significantly more bandwidth beyond this point.

For bit rates of 128 kbps stereo or higher, the Final Output Level can be left at 0.00 dBFS and the sample rate can be set at 44.1kHz.

For a bit rate of 112 kbps stereo, the sample rate should be set to 32kHz with a Final Output level of -0.5.

At a bit rate of 96 kbps stereo, the sample rate should be set to 32kHz with a Final Output Level of -1.0.

For bit rates of 64 kbps stereo and lower, we recommend changing the sample rate to 22.05kHz and setting the Final Output Level to -2.0 to prevent any chance of decoder-side clipping.

If you need to provide a very low bit rate MP3 stream, we suggest using 32 kbps *mono* at a sample rate of 22.05kHz with a Final Output Level of -2.0.

AAC

Technical Details

AAC - an abbreviation for Advanced Audio Coding - is a form of MPEG-4 encoding usually presented in an MP4 wrapper and commonly (if not incorrectly) referred to as just "MP4". It is the most modern "lossy" encoder available today. It converts the original audio signal from time-domain to frequency-domain like MP3, but in a more direct method. It offers a modular approach to encoding including the so-called "AAC +" or "aacPlus" variations, properly referred to as HE-AAC (high-efficiency AAC).

There are different modes available within AAC+ itself. Omnia.9/XE provides AAC+ from within its AAC encoders whenever its SBR (Spectral Band Replication) control is set to "On" or "Oversampled". When this control is set to "Off", encoding is plain AAC.

While these modes and AAC+ are technically still part of the AAC encoding process, they merit their own sections and will be covered in detail separately.

Advantages

AAC is the most efficient encoder currently available. It provides better quality audio at a given bit rate when compared to MP3. It has gained acceptance and popularity in recent years and can be decoded by most players.

Disadvantages

Decoding for AAC is not quite as universal as MP3, but it is nearly so due mainly to its use by Apple in its iTunes player and iPhone/iPad/iPod portable devices.

Recommended Settings

"Broadcast quality" audio can be achieved at a bit rate of 96 kbps stereo at a 44.1kHz sample rate, though 128 kbps stereo is recommended for the best quality and fewest audible encoding artifacts. 160 and 192 kbps stereo sound better still, but 192 kbps stereo marks the point of diminishing return where significantly more bandwidth will yield comparatively small improvements to audio quality.

At 128 kbps stereo, no adjustment to the Final Output Level is necessary. It can be left at 0.0dBFS.

At 96 kbps stereo, we recommend a sample rate of 32kHz and setting the Final Output Level to -1.0.
If you find the need to use a bit rate below 96 kbps stereo, we recommend one of the SBR modes described in detail below.

HE-AAC (AAC+) in SBR Mode
Technical Details
SBR mode recreates the high end from a frequency-doubled version of the original frequencies. The base AAC encoding is performed at one-half the specified sample rate while the SBR encoding is done at the full sample rate.

Advantages
Standard SBR mode offers the most "bang for the buck" for lower bit rate encoding (64 kbps stereo and below), but standard AAC or AAC+ in Oversampled SBR mode are better choices for higher bit rates. When an AAC+ stream is presented to an encoder that can decode standard AAC but not AAC+, it will still play the base AAC audio eliminating compatibility issues.

Disadvantages
Using SBR causes a lack of tight peak control in material that has been heavily processed and often requires the Final Output Level to be reduced to compensate.

While the compatibility with players than can decode AAC but not AAC+ can be seen as an advantage (at least they will play *something*) the base AAC encoding is done at a lower sample rate, which yields audio with a narrower frequency range and less high end.

Finally, because of the way in which the harmonics created in the SBR process correlate to their corresponding fundamental frequencies, the audio can take on a "metallic" sound not found in true AAC.

Recommended Settings
A bit rate of 64 kbps stereo and a sample rate of 44.1kHz can deliver "broadcast quality" audio, but the Final Output Level should be set to -1.0dBFS to prevent decoder-side clipping.

48 kbps stereo can be adequate for reproduction on smaller devices (mobile phones and tablets, for instance) with a Final Output Level setting of -1.5.

At bit rate of 32 kbps, we recommend setting the Channel Mode to Parametric Stereo. This creates a filtered mono mix that won't be as pleasant as true stereo but represents the best possible use of the available bits and is a good compromise. The Final Output level should be set to -1.5.

HE-AAC (AAC+) in SBR Oversampled Mode
Technical Details
Like in the standard SBR mode, the base AAC encoding is performed at one-half the sample rate of the SBR encoding. However, in the Oversampled mode, base AAC encoding is performed at the specified sample rate while SBR encoding is accomplished at two times (double) the specified sample rate, thus the term "oversampled".

Advantages
Oversampled SBR can yield a frequency response all the way out to 20kHz at a sample rate of 128 kbps. In comparison, a standard AAC stream encoded at the same sample rate would be frequency limited to 16kHz.

This is a benefit in and of itself, there is an additional advantage: AAC players that cannot decode the SBR portion of the stream and pass only the base AAC portion will yield audio all the way out to 16kHz.

Disadvantages

See "Disadvantages" for AAC+ in SBR mode.

Recommended Settings

Like with standard SBR, Oversampled SBR can provide "broadcast quality" results with a 64 kbps stereo bit rate and is really the ideal setting. The Final Output Level should be set to -1.5dBFS.

Oversampled SBR is the preferred mode for 128 kbps stereo encoding and requires no Final Output Level attenuation.

For bit rates beyond 128 kbps, we recommend standard AAC encoding be used.

WMA

Technical Details

WMA (Windows Media Audio) is a "lossy" compression technology developed by Microsoft as a part of the Windows Media framework. Computers running the Microsoft Windows operating system are commonly equipped with with the Standard Windows Media Audio codec (version 9.2 at the time of this writing), the WMA10 Pro codec, or both. Which codecs are included in your particular version of Windows will determine which encoders are available when using Omnia.9/XE. Like AAC, it is a pure transform coder.

Advantages

At one time, AAC was not supported in Flash players and WMA offered a potential compatibility advantage. Flash players now support AAC, leaving no real advantage today to using WMA for streaming audio.

Disadvantages

WMA is among the least efficient encoding methods. Encoding artifacts are still audible at comparatively high bit rates.

Recommended Settings

192 kbps stereo is the minimum bit rate for "broadcast quality" audio. No reduction in the Final Output Level is necessary; it can be left at 0.0dBFS.

Encoding artifacts are clearly audible at rates as high as 128 kbps.

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