Z/IPStream R/2

Telos Z/IPStream R/2

Z/IPStream® R/2 provides a multitude of options for broadcasters with the goal of maximizing streaming audio quality for listeners.

This second-generation Z/IPStream processor and encoder is essentially the hardware appliance version of the successful X/2 and 9X/2 software, allowing flexible, multi-format stream-encoding for up to eight audio programs in a single 1RU chassis and is ideal for high-density applications.

R/2 is available with 3-band Omnia processing or full Omnia.9 processing.

Notices and Cautions

CAUTION:

The installation and service instructions in this manual are for use by qualified personnel only. To avoid electric shock, do not perform any servicing other than that contained in the operating instructions unless you are qualified to do so. Refer all servicing to qualified personnel.

This instrument has an auto-ranging line voltage input. Ensure the power voltage is within the specified range of 100-240v. The ~ symbol, if used, indicates an alternating current supply.

 This symbol, wherever it appears, alerts you to the presence of uninsulated, dangerous voltage inside the enclosure – voltage which may be sufficient to constitute a risk of shock.

 This symbol, wherever it appears, alerts you to important operating and maintenance instructions. Read the manual.

CAUTION: DOUBLE POLE/NEUTRAL FUSING

The instrument power supply incorporates an internal fuse. Hazardous voltages may still be present on some of the primary parts even when the fuse has blown. If fuse replacement is required, replace fuse only with same type and value for continued protection against fire.

WARNING:

The product’s power cord is the primary disconnect device. The outlet should be located near the device and easily accessible. The unit should not be located such that access to the power cord is impaired. If the unit is incorporated into an equipment rack, an easily accessible safety disconnect device should be included in the rack design.

To reduce the risk of electrical shock, do not expose this product to rain or moisture. This unit is for indoor use only.

This equipment requires the free flow of air for adequate cooling. Do not block the ventilation openings in the top and sides of the unit. Failure to allow proper ventilation could damage the unit or create a fire hazard. Do not place the units on a carpet, bedding, or other materials that could interfere with any panel ventilation openings.

If the equipment is used in a manner not specified by the manufacturer, the protection provided by the equipment may be impaired.

WARNUNG:

Die Installations-und Serviceanleitung in diesem Handbuch ist für die Benutzung durch qualifiziertes Fachpersonal. Um Stromschläge zu vermeiden führen Sie keine andere Wartung durch als in dieser Betriebsanleitung aufgeführt, es sei denn Sie sind dafür qualifiziert. Überlassen Sie alle Reparaturarbeiten qualifiziertem Fachpersonal.

Dieses Gerät hat eine automatische Bereichseinstellung der Netzspannung.

Stellen sie sicher, dass die verwendete Netzspannung im Bereich von 100-240V liegt.

Das Symbol ~, falls verwendet, bezeichnet eine Wechselstromversorgung.

 Dieses Symbol, wo immer es auftaucht, macht Sie auf nicht isolierte, gefährliche elektrische Spannung (ausreichend um einen Stromschlag hervorzurufen) innerhalb des Gehäuses aufmerksam. Spannungen.

 Dieses Symbol, wo immer es auftaucht, weist Sie auf wichtige Bedienungs-und Wartungsanleitung hin. Lesen Sie die Bedienungsanleitung.

ACHTUNG: ZWEIPOLIGE ABSICHERUNG / NULLEITER ABSICHERUNG

Das Netzteil des Gerätes hat eine interne Sicherung eingebaut. Auch wenn die Sicherung durchgebrannt ist, können auf einigen primären Bauteilen noch gefährliche Spannungen vorhanden sein. Wenn ein Austausch der Sicherung erforderlich ist, ersetzen Sie die Sicherung nur mit gleicher Art und Wert für den kontinuierlichen Schutz gegen Feuer.

WARNUNG:

Das Gerätenetzkabel ist die Haupttrennvorrichtung. Die Steckdose sollte sich in der Nähe des Gerätes befinden und leicht zugänglich sein. Das Gerät sollte nicht so angeordnet sein, dass der Zugang zum

Netzkabel beeinträchtigt ist. Wird das Gerät in ein Rack eingebaut, sollte eine leicht zugängliche Sicherheitstrennvorrichtung in den Rack-Aufbau mit einbezogen werden.

Um die Gefahr von Stromschlägen zu verringern, darf dieses Produkt nicht Regen oder Feuchtigkeit ausgesetzt werden. Dieses Gerät ist nur für die Benützung im Innenbereich. Dieses Gerät erfordert freie Luftzirkulation für eine ausreichende Kühlung. Blockieren sie nicht die Lüftungsschlitze auf der Geräteoberseite und den Seiten des Gerätes. Unzureichende Belüftung kann das Gerät beschädigen oder Brandgefahr verursachen. Platzieren Sie das Gerät nicht auf einem Teppich, Poster oder andere Materialien welche die Lüftungsöffnungen beeinträchtigen könnten.

Wird das Gerät anders als in der, vom Hersteller angegebenen Weise verwendet, kann der, durch das Gerät gegebene Schutz beeinträchtigt werden.

USA CLASS A COMPUTING DEVICE INFORMATION TO USER. WARNING:

This equipment generates, uses, and can radiate radio-frequency energy. If it is not installed and used as directed by this manual, it may cause interference to radio communication. This equipment complies with the limits for a class a computing device, as specified by fcc rules, part 15, subpart j, which are designed to provide reasonable protection against such interference when this type of equipment is operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference. If it does, the user will be required to eliminate the interference at the user's expense. Note: objectionable interference to tv or radio reception can occur if other devices are connected to this device without the use of shielded interconnect cables. Fcc rules require the use of shielded cables.

CANADA WARNING:

"This digital apparatus does not exceed the Class A limits for radio noise emissions set out in the radio interference regulations of the Canadian department of communications."

"Le présent appareil numérique n'émet pas de bruits radioélectriques dépassant les limites applicables aux appareils numériques (de class a) prescrites dans le règlement sur le brouillage radioélectrique édicté par le ministère des communications du Canada."

CE CONFORMANCE INFORMATION:

This device complies with the requirements of the EEC council directives:

- 93/68/EEC (CE MARKING)
- 73/23/EEC (SAFETY – LOW VOLTAGE DIRECTIVE)
- 89/336/EEC (ELECTROMAGNETIC COMPATIBILITY)

Conformity is declared to those standards: EN50081-1, EN50082-1.

Z/IPStream R/2 MANUAL

TRADEMARKS

Telos, Omnia, Axia, Z/IPStream, and the Z/IPStream logo are trademarks of TLS Corp.. All other trademarks are the property of their respective holders.

NOTICE

All versions, claims of compatibility, trademarks, etc. of hardware and software products not made by Telos which are mentioned in this manual or accompanying material are informational only. Telos makes no endorsement of any particular product for any purpose, nor claims any responsibility for operation or accuracy. We reserve the right to make improvements or changes in the products described in this manual which may affect the product specifications, or to revise the manual without notice.

WARRANTY

This product is covered by a five year limited warranty, the full text of which is included in this manual.

UPDATES

The operation of Z/IPStream R/2 is determined largely by software. We routinely release new versions to add features and fix bugs. Check the Telos Alliance web site for the latest.

FEEDBACK

We welcome feedback on any aspect of Z/IPStream R/2, or this manual. In the past, many good ideas from users have made their way into software revisions or new products. Please contact us with your comments.

SERVICE

You must contact The Telos Alliance before returning any equipment for factory service. We will need your unit’s serial number, located on the back of the unit. The Telos Alliance will issue a return authorization number, which must be written on the exterior of your shipping container. Please do not include cables or accessories unless specifically requested by the Technical Support Engineer. Be sure to adequately insure your shipment for its replacement value. Packages without proper authorization may be refused. US customers, please contact Telos Technical Support at +1-216-622-0247. All other customers should contact local representative to make arrangements for service.

WE SUPPORT YOU...

BY PHONE / FAX:

- You may reach our 24/7 Support team anytime around the clock by calling +1-216-622-0247.

- For billing questions or other non-emergency technical questions, call +1-216-241-7225 between 9:30 am to 6:00 PM, USA Eastern time, Monday through Friday.
- Our Fax number is +1-216-241-4103.

BY E-MAIL:

- Technical support is available at support@telosalliance.com.
- All other questions, please email inquiry@telosalliance.com.

VIA WORLD WIDE WEB:

The Telos Alliance web site has a variety of information which may be useful for product selection and support. The URL is www.TelosAlliance.com.

REGISTER YOUR PRODUCT

Did you know that all Telos Alliance products come with a 5-Year Warranty? Take a moment to activate your coverage online at http://telosalliance.com/product-registration/ .

The Telos Alliance

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Quick Start Guide

Welcome to the Z/IPStream R/2

We know you're anxious to mount your new Telos Z/IPStream R/2 in the rack and get it running, but first, please take a moment to register your product online. Registration activates the warranty and is the best way to assure you will get quality support should you have questions regarding installation, operation, or troubleshooting.

Visit http://telosalliance.com/product-registration/ to register.

While visiting our website, you may also want to sign up for our Telos Alliance Newsletter so that you'll receive notification of new firmware upgrades available for your Z/IPStream R/2.

Visit https://www.telosalliance.com/newsletter to sign up.

Initial Configuration

Except for the initial network setup, the product setup and configuration will be done using a web browser. Z/IPStream R/2 ships with network addresses pre-configured as follows:

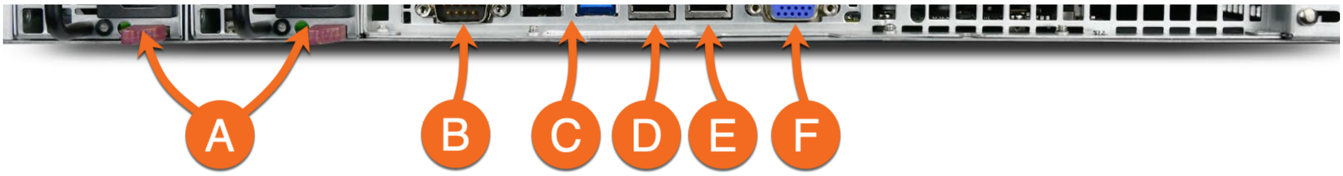
NET 1

- IP Address (static): 10.10.2.100
- Subnet Mask: 255.255.255.0
- Gateway: 10.10.2.1
- DNS: 10.10.2.1

NET 2

- IP Address (static): 192.168.2.100
- Subnet Mask: 255.255.255.0
- Gateway: None
- DNS: None

> ⓘ If these addresses fit your network scheme, you can connect to either network and proceed with the configuration. If not, proceed to the Alternate IP Address setup.



A. Power
B. RS232 (Serial)
C. USB Ports
D. NET 1
E. NET 2
F. VGA Connector
G. Intelligent Platform Management Interface (IPMI)
H. Audio 5 - 8
I. Audio 1 - 4

AC Power Connections

The Z/IP Stream R/2 does not have a power button; it will power on automatically once connected to AC Power.

The R/2 comes with dual power supplies (120V - 240V, 50 - 60Hz) for enhanced reliability. Please connect the power cords to both power connectors on the back using separate power sources.

> **Note -** If you are only connecting one of the power supplies, remove the other power supply (either fully or partially) to avoid a power supply alarm condition.

> **Note -** R/2 is fan-cooled. During boot-up, the fans run at full speed and can be quite loud. They will slow down (and quiet down) once boot-up is complete, but R/2 is designed to be installed in an area where fan noise will not be an issue.

Network Connections

The R/2 offers dual network adapters, NET 1 and NET 2 (see image above). The two adapters may provide a redundant network path to the Internet or separate the internal network (e.g., Livewire) from an outside network.

> **Important:** Do not connect both NICs to the same network. Doing so may create confusion as to which NIC to use to reach a specific destination.

Audio Connections

Telos Part # 1711-00368 (Lynx Audio Part AES1604) cables. Two required. Supplied with all new units.

The AES/EBU audio connections are provided through two 26-pin D-sub connectors. Connect the breakout cable(s) to the D-sub connectors. Use the thumbscrews to attach the cables in place securely.

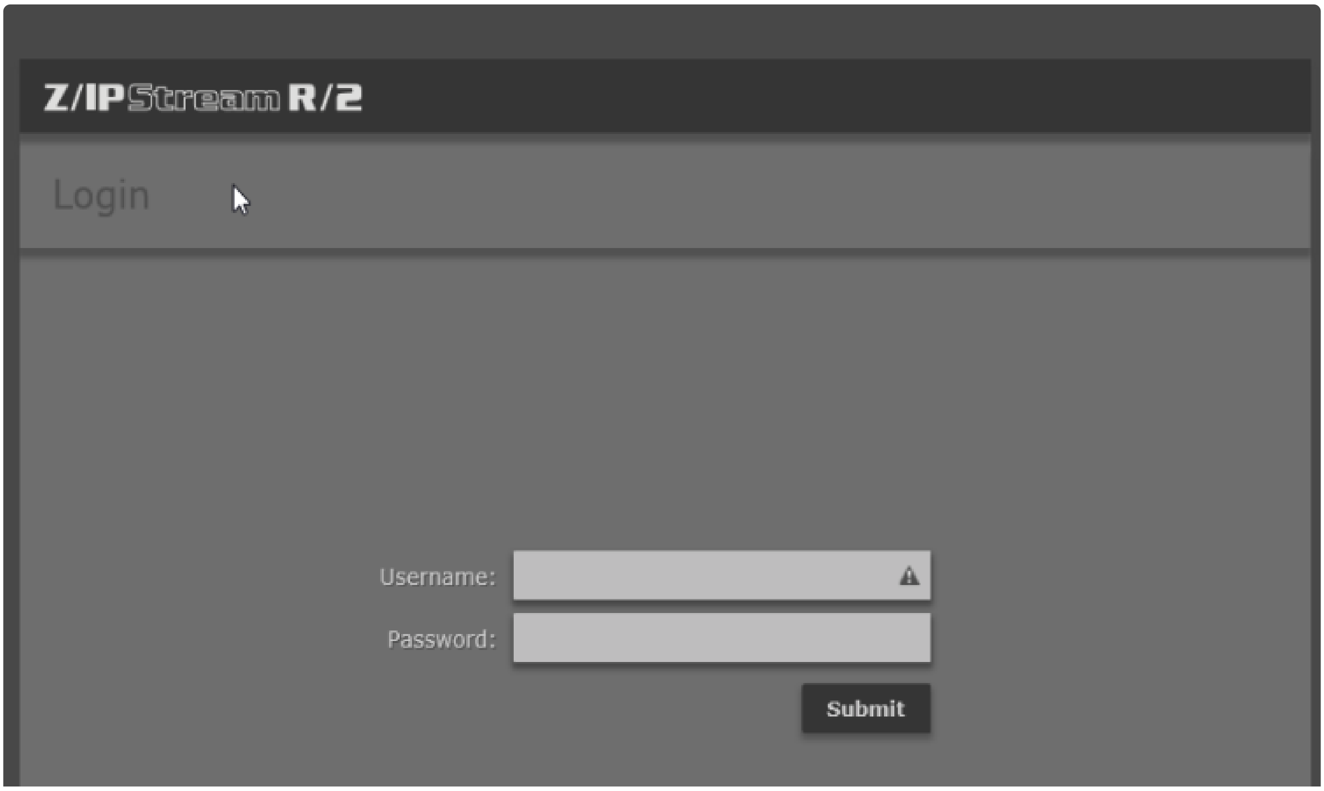
Continuing the Configuration Process

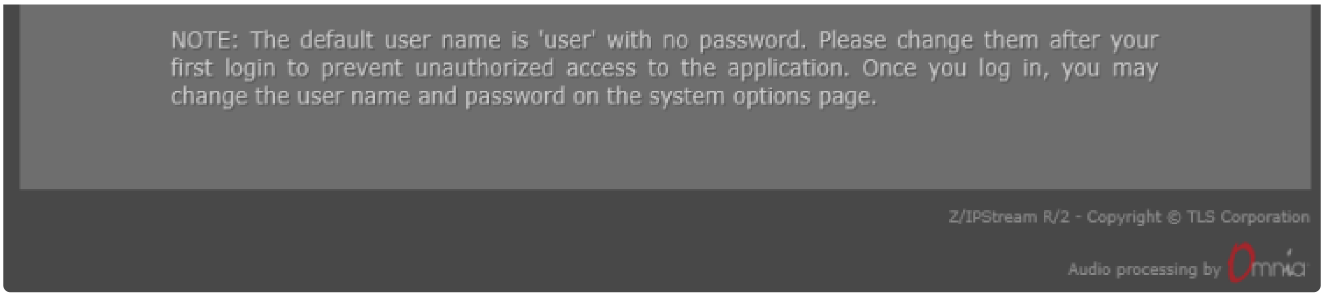
At this point, the initial hardware configuration is complete and the rest of the configuration is performed through a web browser from another computer.

Make a note of the static IP address indicated above. On a computer connected to the same network as the R/2, enter R/2's IP address in a web browser application.

- For NET 1 - Enter http://10.10.2.100 in your browser
- For NET 2 - Enter http://192.168.2.100 in your browser

The following login prompt will be displayed:





Z/IPStream R/2 Login Screen

Enter "**user**" as the username, leave the password field blank, then click the "Submit" button. You should now be logged in to the Z/IPStream R/2.

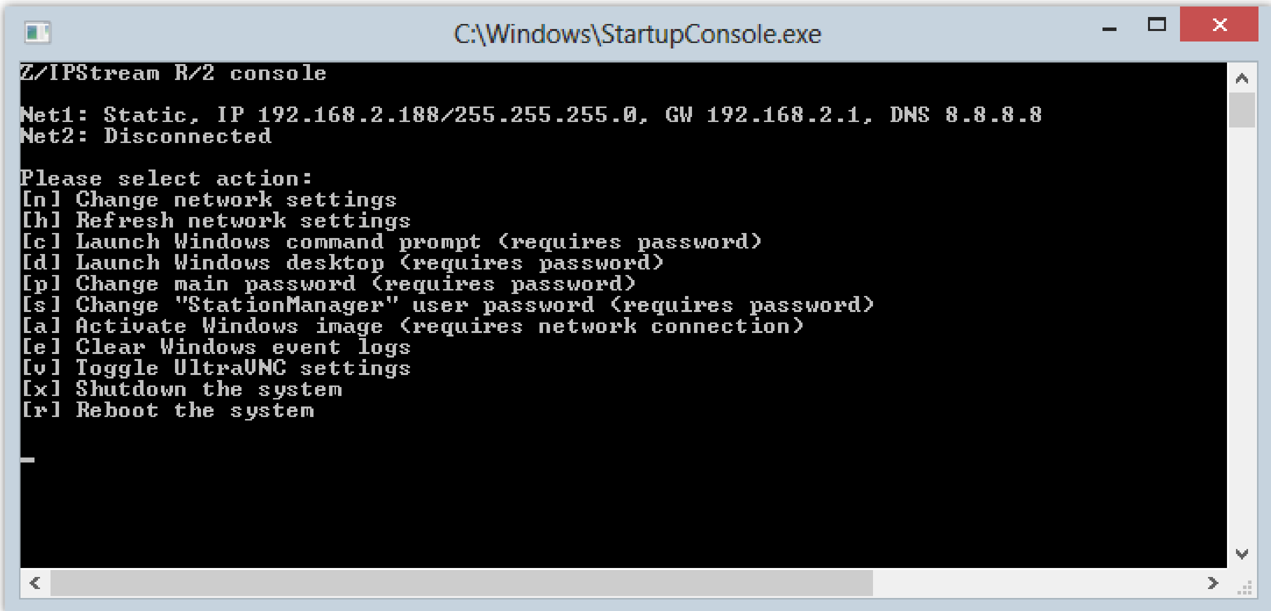
Please refer to the User Manual for full product configuration information.

Alternate IP Address Setup

If neither of the pre-configured IP addresses will work or you want to set them manually, you will need to connect a VGA monitor, USB keyboard, USB mouse and use the R/2's Startup Console.

After you connect the monitor, the Startup Console is shown.

> Many options are available from this console, most of which are for advanced use-cases and are beyond the scope of this quick start guide. We will use only the **[n] Change Network Settings** option here.
>
> **All remaining configuration is completed using a connected Web Browser.**



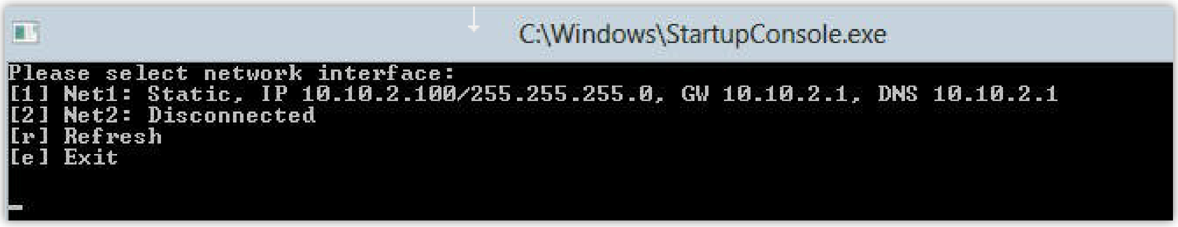
Z/IPStream R/2 Startup Console

- Use the mouse to click on the StartupConsole Window to make sure it has focus; the mouse will not be used again

- Press **[n] Change network settings**
- Choose **[1]** or **[2]** to change the network address settings.
- Choose the correct mode for this Network Interface.
  - [d] DHCP
  - [s] Static
  - [d] Disabled
- If you chose [s] Static, enter the following:
  - IP Address
  - Netmask
  - Gateway (optional, Enter to skip)
  - DNS (optional, enter to skip)

Repeat these steps to configure the second network interface.

Completing these steps will apply your new IP address settings immediately.



Once you are finished with the IP Address setup, return to the Initial Configuration section.

Please refer to the User Manual for additional information.

Installation Help

Contact the Telos Alliance Support Team via email at support@telosalliance.com, by phone at +1 (216) 241-7225, or online at https://success.telosalliance.com/support for assistance.

User Manual

Read Me. Please. (OK, at least read some of me.)

Read Me. Please. (OK, at least read some of me.)

If you're like many engineers or IT types, you've probably already put your brand new R/2 into the rack, connected power, connected the network, and perhaps even connected some audio. You may have even powered it up and figured out how to set an IP address to access the web GUI. If that's you…Great! You're a few steps ahead of the game and we hope this manual can provide some additional valuable information for you.

If that's totally not you and you're more of a "by the book" kind of person, that's great too. We hope you've at least gone through the Quick Start Guide section and found it helpful. This manual will cover some of the same basics as the Quick Start Guide, and go into greater detail about how to get your R/2 up and running.

The layout of this manual is pretty straightforward…It should be in the order you would normally perform installation and configuration steps for the most part (though there may be a few little rabbit trails). The author has tried to take a "what would someone trying to set this up for the first time need to know, and in what order" approach. Some of the chapters dealing with audio processing setup or other configuration details will be a bit longer than others, but overall this manual should be a pretty quick read.

We realize reading manuals isn't everyone's idea of a great time, but we'll try at least to keep it informative and somewhat entertaining—though the author has a habit of cracking some pretty bad jokes. If you aren't asleep by the end of it, and have your R/2 up and running, this manual has accomplished its goal.

There are probably a lot of things you could be doing with your time besides reading a manual, and we appreciate that you've taken the time to read this one. If you REALLY don't have time to read anything else, start with the Installation Section and read all the way through to the end of the Audio Sources section. The Processing and Encoding section is probably the longest in the manual but covers how to actually make R/2 start doing stuff. If you're in even more of a hurry, jump straight to the Frequently Asked Questions at the end of this manual.

If after reading this manual you have any suggestions for improvement (or if you see something we might have missed) send a note to inquiry@telosalliance.com.

As always, our support team is available 24/7 to assist by calling 216-622-0247 or e-mail support@telosalliance.com.

The world is listening. Stream like you mean it.

What is the Z/IP Stream R/2

A Bit of History

It started with the telephone…

The orignal Telos 10 phone hybrid

For over 30 years, Telos has led the industry in telephony and audio encoding technology for broadcast. When Steve Church developed the first digital telephone hybrid for broadcast in 1984, it was the beginning of a revolution in how the industry would get audio from the field on the air. Telephone calls had never sounded so good.

MP3 and ISDN

The Zephyr "classic"

This revolution continued with the introduction of the Telos Zephyr ISDN codec in 1993. It was the first codec to make use of the (then relatively new) MPEG Layer III codec to send full 20 kHz mono audio across a single 56 or 64 kbps data channel. Broadcasters could finally bring full-fidelity audio to their listeners from the field (or between facilities) without dedicated equalized program loops or RF links. The Zephyr quickly became the de-facto industry standard for ISDN codecs, used by studios around the globe for transporting audio.

Streaming…It begins.

Audio Active

While ISDN was becoming popular for broadcasters to transport audio from the field or between facilities, the next frontier was exploring new methods for getting audio to the listeners. The Internet was still in its infancy, but it had developed to the point where streaming audio (even over dialup) was becoming viable. In 1997, Telos introduced the very first dedicated hardware encoder for MP3 streaming over the Internet, simplifying what was a bit of a cumbersome process and dramatically improving the audio quality.

Omnia: A Legend in Audio Processing for Broadcast

Omnia Net

During this same period of time, Frank Foti was working tirelessly to improve the sound of audio processing for broadcast. His dial-dominating innovations in audio processing at Z100 were the foundation of what would ultimately become Omnia. The audio processing developments didn't stop at the FM dial, however… It quickly became apparent that there was considerable benefit to be gained from specialized audio processing for streaming as well.

Telos Encoding + Omnia Processing = Z/IPStream

The Telos Z/IPStream R/1

As streaming continued to gain in popularity, the demand for an "all in one" solution to handle both processing and encoding for streams grew tremendously. In 2009, Omnia launched the A/XE software package, combining the best of Telos encoding technology with 3-band Omnia audio processing tailored specifically for streaming. In 2012, Telos introduced R/1, a new hardware appliance designed specifically for stream processing and encoding without a dedicated PC.

The Z/IPStream family gained yet another powerful option with the availability of acclaimed Omnia.9 processing in 9X/2 streaming software. Z/IPStream represents the latest in encoding and audio processing developments from The Telos Alliance and supports a wide variety of streaming technologies including adaptive bitrate streaming for a seamless listener experience regardless of network conditions, or listening platform.

About Z/IPStream R/2

The Original Z/IPStream R/2

Up to 8 Channels of Z/IPStream, one box

The R/2 is a dedicated multichannel processor/encoder appliance for streaming, designed for applications demanding the ultimate in reliability, with the highest possible audio quality and channel density per rack space.

Features

Z/IPStream R/2 supports up to 8 simultaneous audio channels, including the latest generation of licensed MP3 and AAC codecs for streaming, with signature Omnia 3-band processing included or powerful (optional) Omnia.9 processing. Flexible configuration allows multiple processing and/or encoding instances per audio channel.

Audio I/O includes AES/EBU, Livewire Audio over IP, and multicast or unicast RTP. Supported server platforms include HTTP, Icecast, Shoutcast, Wowza, Adobe RTMP, and others.

Triton Digital Integration

R/2 includes direct support for the Triton Digital streaming platform, which allows users to take full advantage of their content distribution and dynamic ad replacement technologies while streamlining hardware and software implementation.

A few words about Livewire+ and AES67

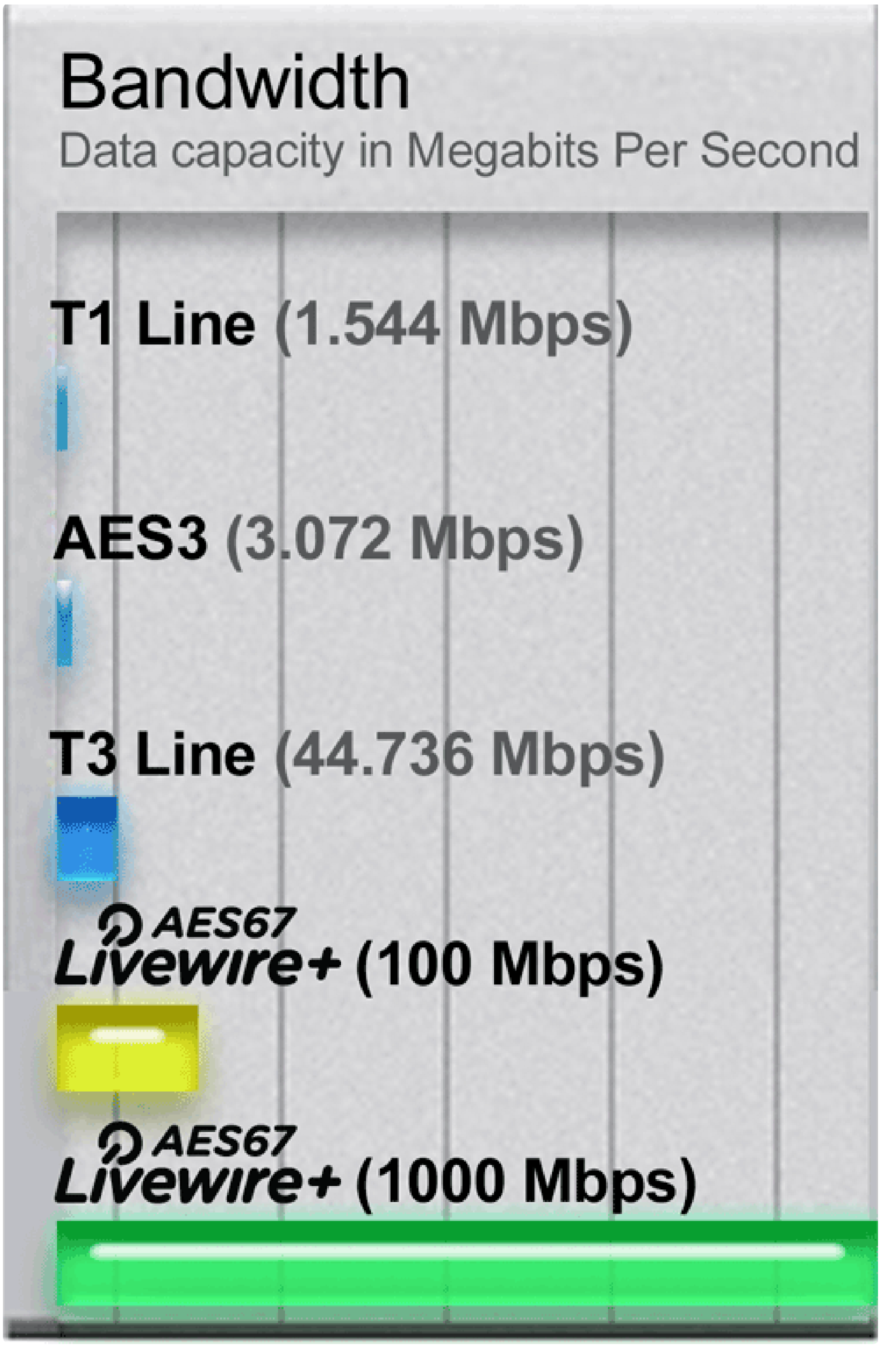
What is Livewire+™?

Livewire+ is the pioneering technology invented by the Telos Alliance to convey low-delay, high-reliability audio over switched Ethernet.

Introduced as Livewire™ in 2003, today's Livewire+ is AES67-compliant. That means that it complies fully with the AES67-2013 Interoperability Standard, allowing AES67 devices to connect directly to Livewire+ networks and exchange audio streams. Livewire+ is also extensible, able to incorporate future AES standards when they are ratified. Livewire+ is also backward-compatible with the RAVENNA™ networking protocol.

With Livewire+, a single Ethernet cable carries real-time uncompressed digital audio, device control messages, program associated data, and even routine network traffic. An entire facility can be wired in hours, instead of weeks. All Axia Audio studio products, and most products from other Telos Alliance brands, utilize Livewire+ networking technology. Expanding or modifying your system is simple thanks to the

Livewire+ offers a revolutionary change in how studios can be built. But at the same time, it's a natural continuation of general trends and what you already know.



How Livewire+ works

Livewire+ has an audio advertising system. Every source has a text name and numeric ID. These are transmitted from source devices to the network. Devices that play audio build lists of all available sources from which users can select.

Using xNode audio interfaces, you enter the names of your input sources via any PC with a web browser. With playout PCs attached to the network, you open a configuration window.

Livewire+ networks employ two types of audio streams. Livestreams have small, frequent packets optimized for live audio that requires very low (circa1 ms.) delay, for microphones and headphone audio. Standard Streams are also real-time streams, but with bigger packets, and are used for audio streams that don't require super-low latency - like audio from CD players, or that exchanged with automation system PCs. Devices that connect to Axia networks can transmit and receive both stream types; the user selects which type to generate when a device is initially configured.

A sophisticated phase-locked loop clocking system allows Livewire+ to use very small buffers for the least latency and ensures that audio channels remain time-aligned (as needed for multiple mics in a studio or for TV surround-sound mixing.)

Converged Networks

An Ethernet network used for Livewire+ audio can also be shared with other data transmissions, such as file transfers and web browsing. An Ethernet system with a switch at the center may have a mix of audio nodes and normal servers, PCs, etc. because the Ethernet switch directs traffic only to where it is needed.

Even on a single link, traffic can be mixed because we use modern Ethernet's priority mechanism to be sure audio packets have first priority on the link's bandwidth. A studio audio delivery system can use this capability to download an audio file from a server, for example, while simultaneously playing another audio file live.

Livewire+ maximizes the benefits of converged networking in the broadcast facility. Many stations using Livewire+ have computer data, telephone, audio, and control on a single network that uses computer industry-standard wiring, spurring cost-efficiencies throughout the plant.

Audio Quality

A Livewire+ network is a controlled, high-speed environment, with no risk of audio drop-outs from network problems and plenty of bandwidth for many channels of high-quality uncompressed audio. We use studio-grade 48kHz/24-bit PCM encoding. Axia digital xNode audio adapters deliver 138dB of dynamic range, with less than 0.0002% THD. Even analog xNodes have 100dB dynamic range, < 0.005% THD, and headroom to +24dBu.

Livewire+ is standards-based

Since the very beginning, The Telos Alliance has based its AoIP networking technologies on standards. IP (Internet Protocol), the networking standard that is the underpinning of nearly all critical business networks (and of the Internet itself) is the basis for Livewire+ AoIP.

As charter members of the AES X.192 Working Group, we helped define the AES67 standard — and became the first broadcast manufacturer to become AES67 compliant.

Livewire+ is so standards-based, in fact, that your audio can even be played by PC media players that support standard protocols and uncompressed PCM audio. The Internet's IP standard for streaming media, called RTP/IP, is used for standard audio streams. RTP stands for Real-Time Protocol. RTP is the Internet's standard way to transport streaming audio and video, just as TCP/IP is the standard for general data.

The Gold Standard

In the decade since the introduction of Livewire+, broadcasters have adopted it at an exponential rate; AoIP has become the new standard for broadcast facilities.

Consider these facts:

- There are over 6,000 studios worldwide built with Livewire+.
- More than 5,500 networked Axia consoles are at work daily.
- Over 80 Livewire+ Partner companies provide compatible hardware and software products.
- There are more than 60,000 individual Livewire+ devices in the field.
- Livewire+ is fully compliant with the AES67 Interoperability Standard.

Impressive, no? But there are even more exciting things in the future. The Telos Alliance, with one of the largest R&D groups in broadcast, is fully committed to AoIP interoperability. We've been proponents of open standards since Day 1, freely sharing our technology with interested Hardware and Software Partners. We were charter, supporting members of the X.192 Working Group that defined the AES67 standard, and as founding members of the Media Networking Allliance are actively engaged in work to promote and enhance standards-based AoIP networking

Installation

3 Installation

Physical Considerations

Rack Mounting

R/2 includes a pair of rack rails designed to secure the unit to the rear of the equipment rack. It is highly recommended that you install these rails if possible due to the depth of the unit. Mounting by the front panel alone may cause excessive strain on the chassis and is not recommended. The supplied rails should fit in most standard equipment racks, but if installing the rails is not possible, you may wish to consider placing an upside-down rack shelf underneath the unit for additional physical support.

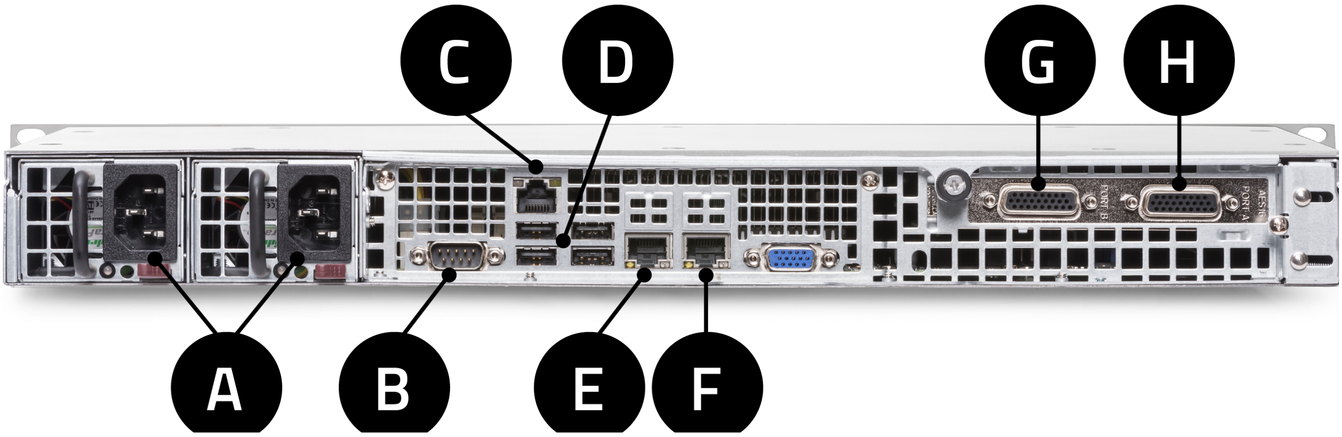
Airflow

As with most rack-mount server ch assis, the airflow on R/2 is from front to back. While it is not absolutely necessary to leave an additional rack space above and below the unit, it certainly is not a bad idea if the space is available. Do not block the front panel of R/2 behind a solid rack door.

Wiring Management

Since airflow is from front to back, it is important to keep the rear of the unit free from obstructions to allow for hot air exhaust. Install the wiring in such a way that it allows for a reasonable service loop and does not block airflow. Do not allow wiring from other equipment in the rack to block airflow from the rear panel of R/2.

Rear Panel



A. Power
B. RS232
C. Control Port
D. USB Ports
E. NIC 1
F. NIC 2
G. Audio 5-8
H. Audio 1-4

Power

The R/2 is equipped with dual power supplies (120V – 240V, 50 – 60Hz) to allow for facilities with redundant power feeds to the equipment racks from separate sources. An LED on the supply will indicate the presence of AC power and proper operation. If power is lost on one of the AC inputs or one of the supplies should fail, the R/2 will sound an alarm. The only way to mute this alarm is to connect power to both supplies or (in the case of power supply failure) remove the failed supply.

Note: If only one of the two power supplies is connected, an alarm will sound. Connect both supplies to AC power, even if you don't have redundant AC power feeds to the equipment rack.

RS-232

Not currently implemented. Inserting stream metadata via RS-232 will be supported in a future release. See chapter 10 (Metadata Sources and Filters) for more details about stream metadata.

"Control" Ethernet Port

This port (if populated) is intended for out-of-band system management. Not currently implemented.

USB Ports

Future expansion or factory service. Not currently implemented.

NIC 1 and NIC 2

The R/2 offers dual network adapters. Typically, one adapter (NIC1) would be used for Audio over IP, while the second (NIC2) would be dedicated to streaming audio traffic. Either interface can be used for management via the web GUI. Connect your network(s) to NIC 1 and/or NIC 2. Do not connect both interfaces to the same network. Doing so may cause routing and connectivity issues.

Note: Do not connect both interfaces to the same network simultaneously. Routing and connectivity issues may result. Ensure each interface is connected to a different IP network, (different VLAN, physical switch, IP address block, etc.) as necessary. Consult your IT department or Telos technical support if you are not sure how the interfaces should be configured. Only one interface may be needed in some rare cases.

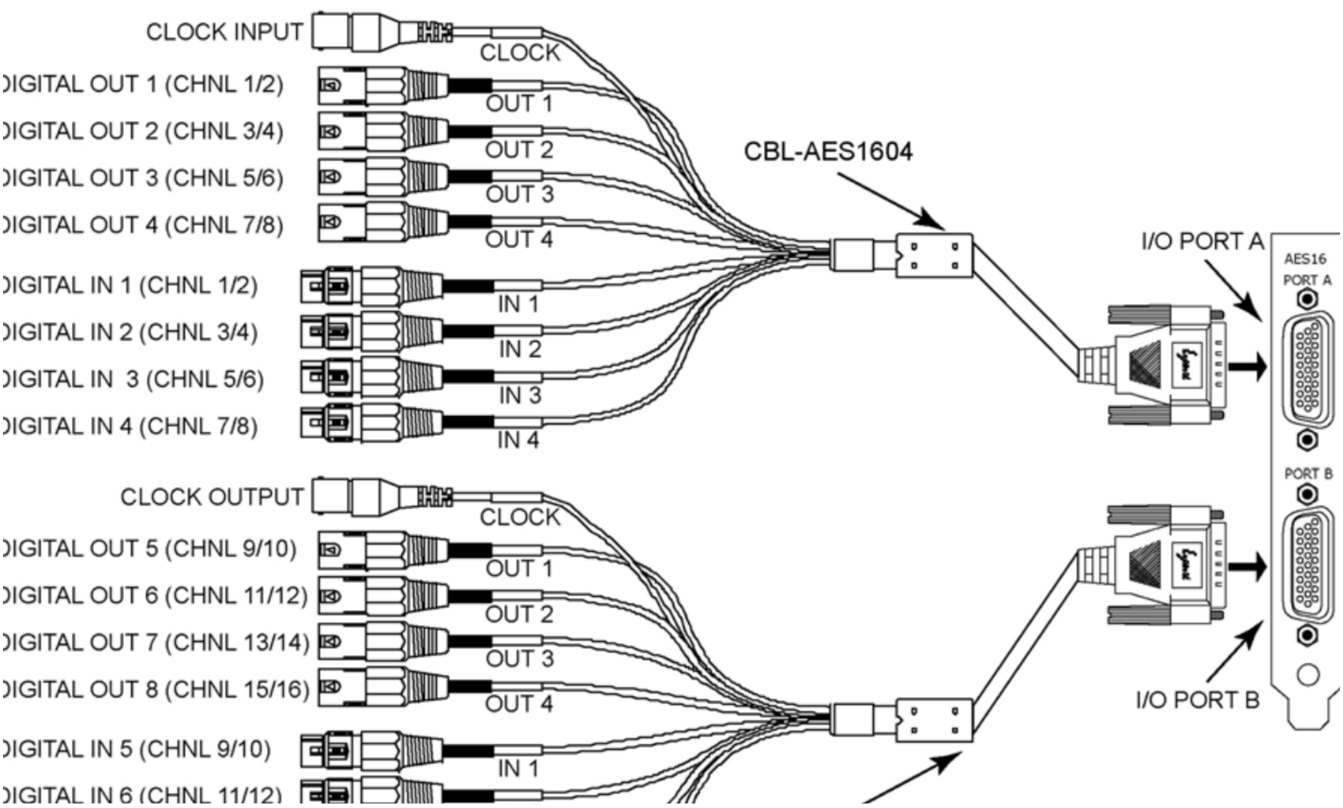
VGA

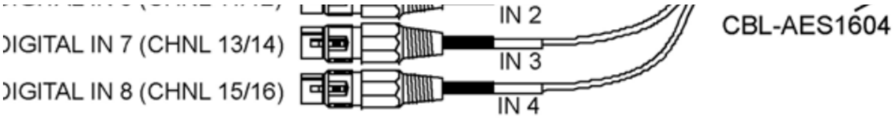
Factory service only. Not currently implemented.

Audio (AES/EBU)

Eight AES/EBU audio connections are available through two high-density 26-pin D-sub connectors. Connect the provided breakout cable(s) to the D-sub connectors. Use the thumbscrews to securely attach the cables in place. Sample rate conversion is provided on all inputs. The BNC Word Clock connectors are not currently implemented. Each audio channel can be configured to generate a stream at 44.1 or 48 kHz (see Chapter 7: Audio Source Configuration).

AES Breakout Cable Layout





Audio Card Pinout

Port A, Jumpers							
	Pin	Signal	Pin	Signal	Pin	Signal	
	1	Clock Gnd	10	Clock In			
	2	OUT4 Cold	11	OUT4 Hot	19	OUT4 Gnd	
	3	OUT3 Gnd	12	OUT3 Cold	20	OUT3 Hot	
	4	OUT2 Cold	13	OUT2 Hot	21	OUT2 Gnd	
	5	OUT1 Gnd	14	OUT1 Cold	22	OUT1 Hot	
	6	IN4 Cold	15	IN4 Hot	23	IN4 Gnd	
	7	IN3 Gnd	16	IN3 Cold	24	IN3 Hot	
	8	IN2 Cold	17	IN2 Hot	25	IN2 Gnd	
	9	IN1 Gnd	18	IN1 Cold	26	IN1 Hot	
	Port B, Jumpers						
	Pin	Signal	Pin	Signal	Pin	Signal	
	1	Clock Gnd	10	Clock Out			
	2	OUT8 Cold	11	OUT8 Hot	19	OUT8 Gnd	

	3	OUT7	12	OUT7	20	OUT7
	4	OUT6 Cold	13	OUT6 Hot	21	OUT6 Gnd

Power Up

Press the “On/Off” button located on the right half of the front panel to power up the unit (See Chapter 4 for a full front panel description). During the power up process, the front panel will light up and display a splash screen. If the display remains dark and fans do not spin up, verify that AC power is present at both power supply inputs.

If the fans spin up but the display remains dark or does not show a splash screen, contact support.

Fans

The fans in R/2 are thermostatically controlled. As the unit powers up, the fans will start at full speed. They should spin down to a much slower speed after a short period of time. If they do not return to low speed, it may indicate a faulty fan or other hardware issue. During normal operation, these fans may increase in speed if the unit’s internal temperature rises. If the fans remain at full speed during operation and the environmental temperature of the unit appears normal, contact support.

Note: The fans will run at full speed for a brief period of time during boot. They should return to a relatively moderate speed within a short period of time. If the fans remain at full speed, it may indicate a hardware fault. Slightly higher fan speed during operation is normal if the internal temperature of the unit increases.

Input Level Meters

Once the unit has fully booted, the front panel LCD will display the input audio level meter screen. As of software version 1.0.34, these meters can display either AES or Livewire input levels by using the up and down navigation arrows on the front panel.

Livewire and other Audio over IP input sources (as well as output level metering and processing metering) can also be metered via the unit’s web GUI or via NfRemote for Omnia.9 processing (See Chapter 16 for Omnia.9 processing information).

Note: Select between Livewire or AES input levels by using the up and down navigation arrows from this screen (see chapter 7 for information about defining audio sources). Metering can also be accessed from within the web GUI or NfRemote software (if using Omnia.9 processing).

Front Panel

4 Front Panel Interface

Overview

This chapter describes the front panel controls and indicators of R/2.

Status LEDs

A column of four LEDs is located directly to the left of the main LCD. The top LED illuminates when the system is operating. The others are not currently implemented.

Main LCD

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The main LCD provides unit status and configuration. If this LCD fails to display the Z/IPStream R/2 splash screen at startup, or remains blank during operation, please contact technical support.

Navigation Cluster

A six button navigation cluster is used for initial IP address configuration and navigating the front panel LCD. The controls are relatively self explanatory. Up and down will increment or decrement values, or scroll between parameter settings. Left and right will move between digits in IP address entry, or move between screens on the front panel. The red “X” button will cancel an entry, the green “” (OK button) will confirm the current setting or select a parameter for modification.

System Status LEDs

A set of five LEDs next to the power and reset buttons indicates system status.

Info

Used for alert indication or system identification.

- Solid red: The system has overheated. This may be caused by reduced airflow or high ambient

temperature in the rack. Ensure there is nothing obstructing the front or back of the unit and that the temperature in the rack is normal.
- Blinking red (1 second intervals): Fan failure. Contact technical support.
- Blinking red (1/4 second intervals): Power supply failure. Check for a failed power supply, contact technical support.
- Blue (solid or blinking): Local or remote Unit ID function—Not currently implemented.

Net 2

Indicates NIC 2 link and activity. Flashes during activity. This LED will remain dark if NIC 2 is not connected.

Net 1

Indicates NIC 1 link and activity. Flashes during activity. This LED will remain dark if NIC 1 is not connected.

SSD

Indicates activity of the internal Solid State Drive. Flashes intermittently during normal operation.

Pwr

This is the main system power LED and will illuminate whenever the system is powered up.

System Controls

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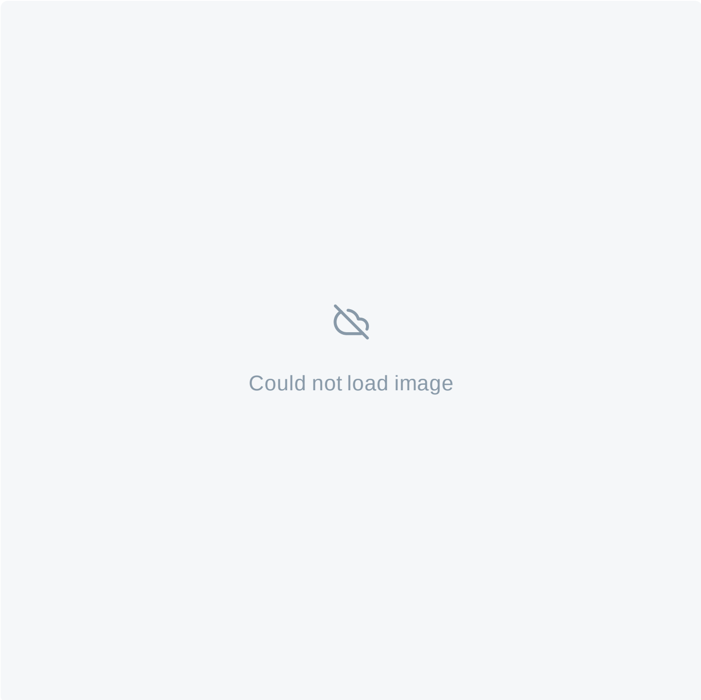
Reset

Performs a “hard” reset on R/2 to reboot. Should not be used during normal operation, but can be used if the R/2 becomes unresponsive for some reason. Should this occur, contact technical support.

On/Off

Performs an orderly power up or shut down. A slight delay before shut down after pressing this button is normal. The SSD indicator will flash rapidly for a short period of time during shut down.

Front Panel USB Port



This port is used for software upgrades, as well as factory configuration and troubleshooting. Contact technical support for additional information.

IP Address Configuration

Overview

After physical installation of R/2 has been completed, at least one of the two network interfaces must be assigned an IP address to allow further configuration and normal operation.

IP Configuration

Each of the two network interfaces on R/2 can be assigned a static IP address, or obtain an IP address automatically via DHCP (Dynamic Host Configuration Protocol). In most applications, R/2 would be assigned static IP addresses, but it may also be helpful to use DHCP with reserved assignments tied to the MAC addresses of R/2. Contact your network administrator if you are unsure of what your network requires.

Metering Screen

After the unit boots, it should be displaying the audio input level metering screen. This is the normal “home” condition for the R/2 display. The up/down arrows will select between AES and Livewire input levels (see Chapter 7). “X” or the left/right arrows will allow you to return to this point from other areas of the front panel screens.

NIC Selection

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From the metering screen, use the right arrow to move to the “Interfaces” screen. Select “NIC 1” or “NIC 2” by using the up/down arrow keys. The display arrow will move to indicate the current selection. Press the “” key to select the desired interface.

Network Mode

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Once you have selected the desired interface (displayed here in the upper right hand corner of the LCD), you can select which mode that interface should operate in. By default, R/2 ships with both interfaces configured for DHCP.

Verify Current Interface Parameters

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To verify the current IP address and mode, from the “Network Mode” screen, press the right arrow. In the example above, NIC 1 is set for DHCP and has an address of 192.168.123.154 with a subnet mask of 255.255.255.0 and a gateway of 192.168.123.253. Press the left arrow or “X” to return to the “Network Mode” screen.

Note: If the network cable is not connected to an interface, one of two conditions may occur…A “self-assigned” IP address of 169.x.x.x or all zeros (even if a static IP has been assigned). This may also occur if the unit is configured for DHCP but not able to reach a DHCP server.

Configuring a Static IP

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From the “Network Mode” screen, press the “” key. You will be asked to switch to static mode. Note the interface number displayed in the upper-right hand corner (NIC 1 in this example).

Use the up arrow key to select “Yes”, then press the “” key. The “Static IP” configuration screen will be displayed.

To modify the static IP configuration:

Use the up/down arrow keys to select the field you want to change (IP address, Netmask, or Gateway).

Press the “” key to begin editing a field.

While editing a field, use the left/right arrow keys to select the digit you wish to change, then the up/down arrow keys to change the value.

When done editing, press the “” key.

When finished entering all values, press the right arrow key. The display will prompt: “Apply Changes?”

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Use the up arrow key to select Yes, then press the “” key to select. After a few moments, you will be returned to the “Network Mode” screen.

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To configure the second NIC, return to the “Interfaces” screen, select “NIC 2”, and repeat the configuration process.

Web GUI Navigation

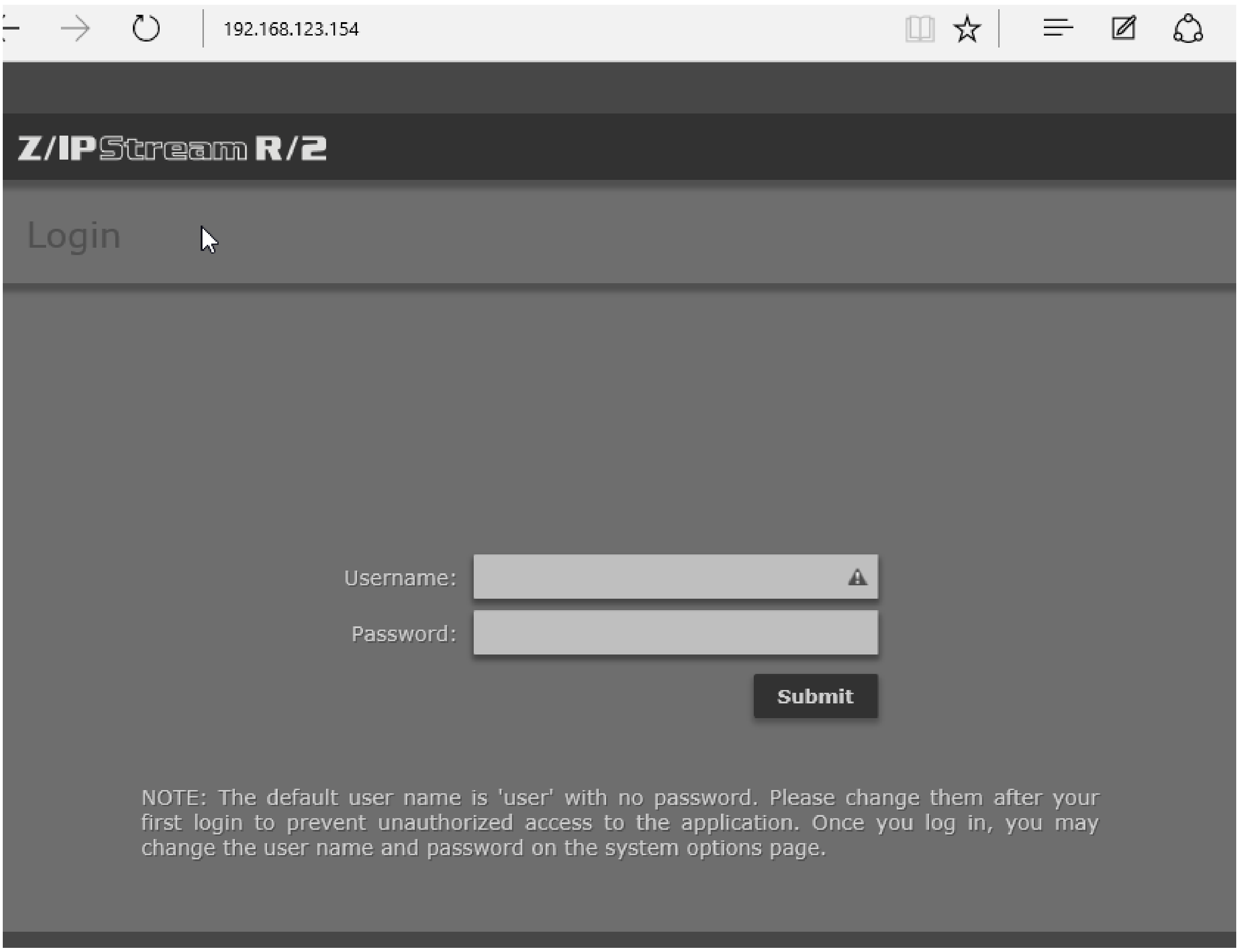
Overview

Beyond initial IP address settings from the front panel, all R/2 configuration is performed through a web browser (or through NfRemote software for Omnia.9 processing—See Chapter 16). This chapter will provide an overview of the R/2 web GUI.

On a computer with access to the same network as R/2, enter the R/2’s IP address (configured from the front panel in the previous chapter) in a web browser. The unit can also be located by the hostname: R2-XXXXXX, replacing XXXXXX with the serial number of the unit. For example, unit 74-1001 will be R2-741001.

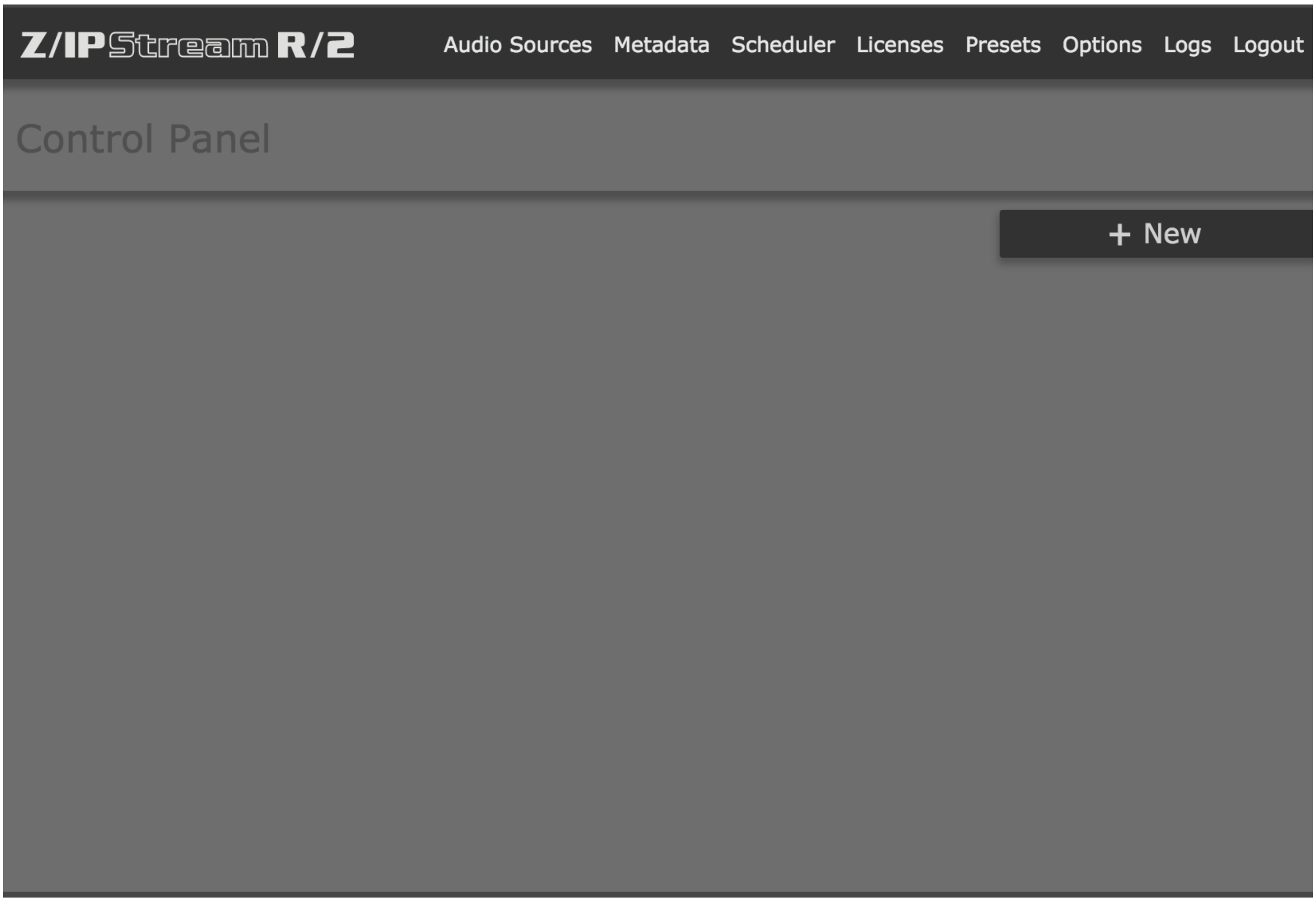
Note: Most modern browsers are supported (including Chrome, Firefox, Safari, Edge, and others that support HTML 5). No Java or Flash is used in the R/2 web GUI.

Login



Enter "**user"** as the user name. There is no default password. Click the Submit button. You should now be logged in to Z/IPStream R/2.

Control Panel

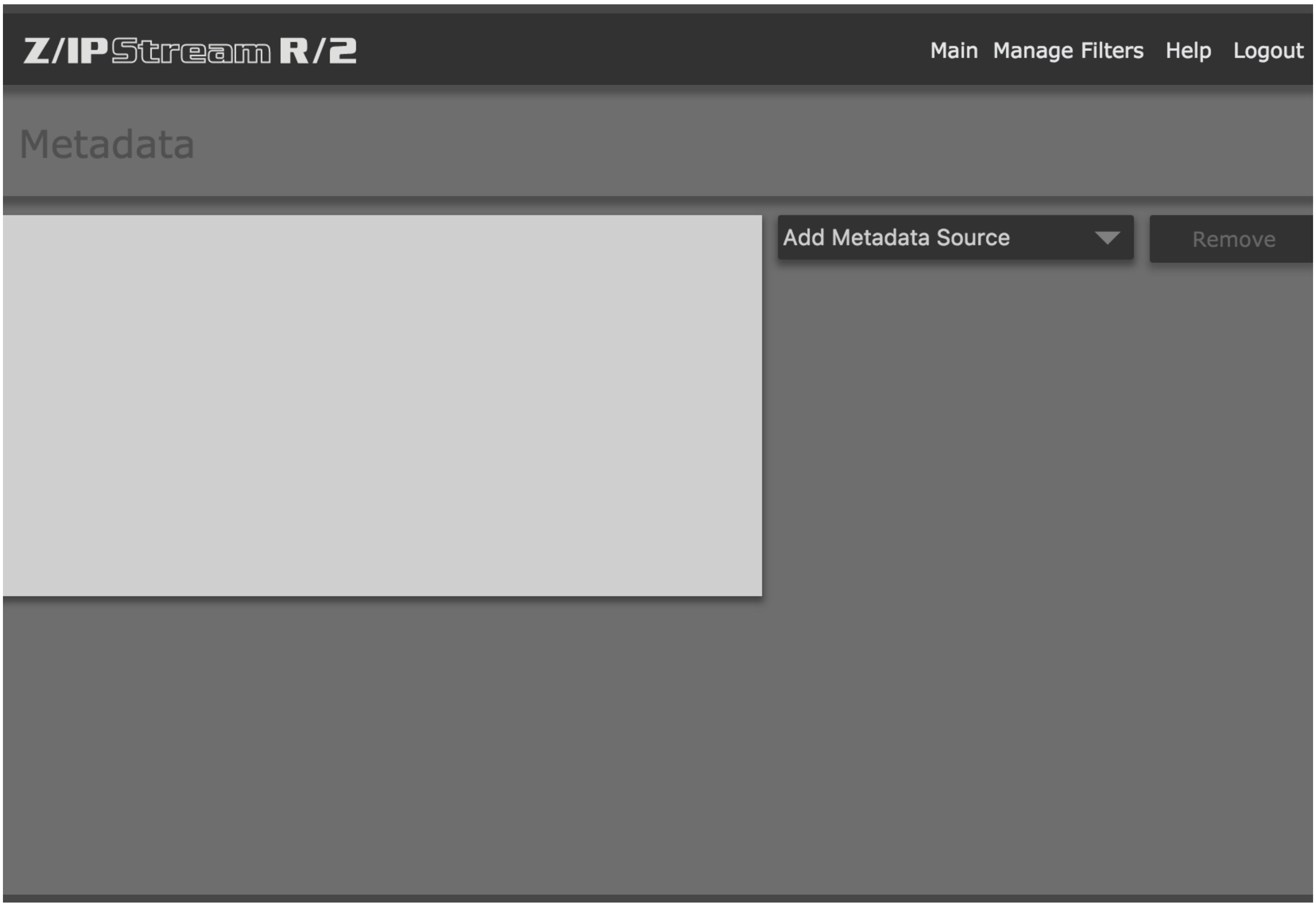


The main “home” display is called the “Control Panel” This is the screen where the status of all processing and encoding instances is displayed, and new processing or encoding instances are defined (Chapter 8). All other configuration functions can be accessed through the sections listed at the top this screen. Click the “Z/IPStream R/2” logo at the upper left corner or “Main” to return here at any time. “Help” in sub-menus will download a copy of this manual, and “Logout” will return to the login screen.

Audio Sources

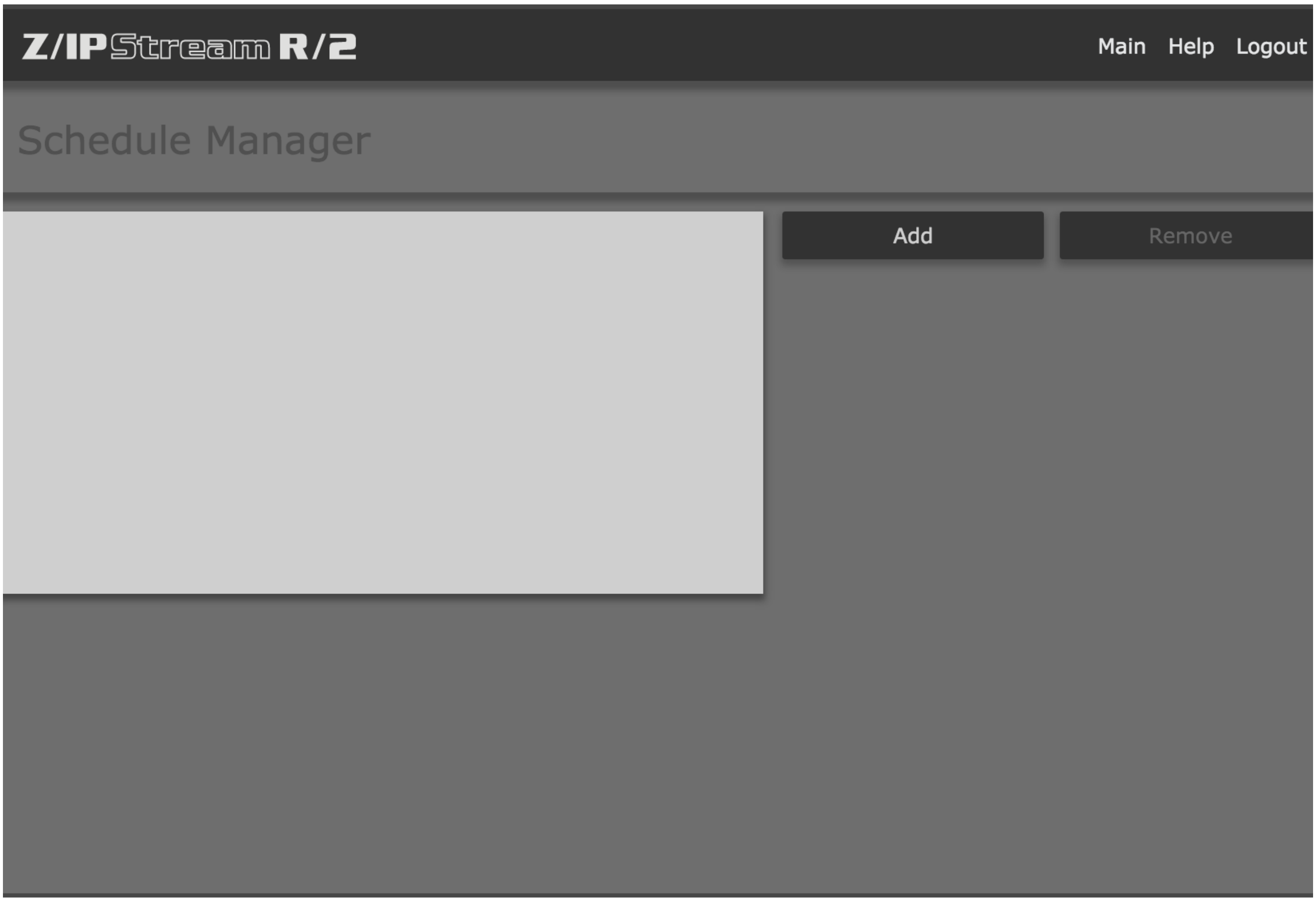
R/2 is licensed by the number of audio sources. These sources must be defined before processing and encoding instances can be configured. The “Audio Sources” screen defines available audio channels for processing and encoding. These sources can be AES inputs from the internal audio card, Livewire channels from an Audio over IP network, or other RTP audio stream (Unicast or Multicast).

Metadata



Web streams frequently have “Now Playing” metadata associated with them for display on websites and other streaming clients. The “Metadata” section of the interface is where these metadata sources are defined. Incoming data via TCP, UDP, is supported. RS-232 will be supported in a future version. Custom XML “Metadata Filters” allow considerable flexibility in data formatting and translation.

Schedule Manager

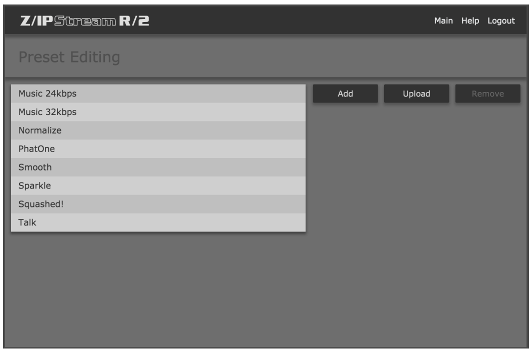


R/2 supports timed scripting of events for various functions. This could include switching to an alternate audio source on a stream, changing a processing preset, or blacking out a sports broadcast or other content for which you do not have streaming rights.

Licenses

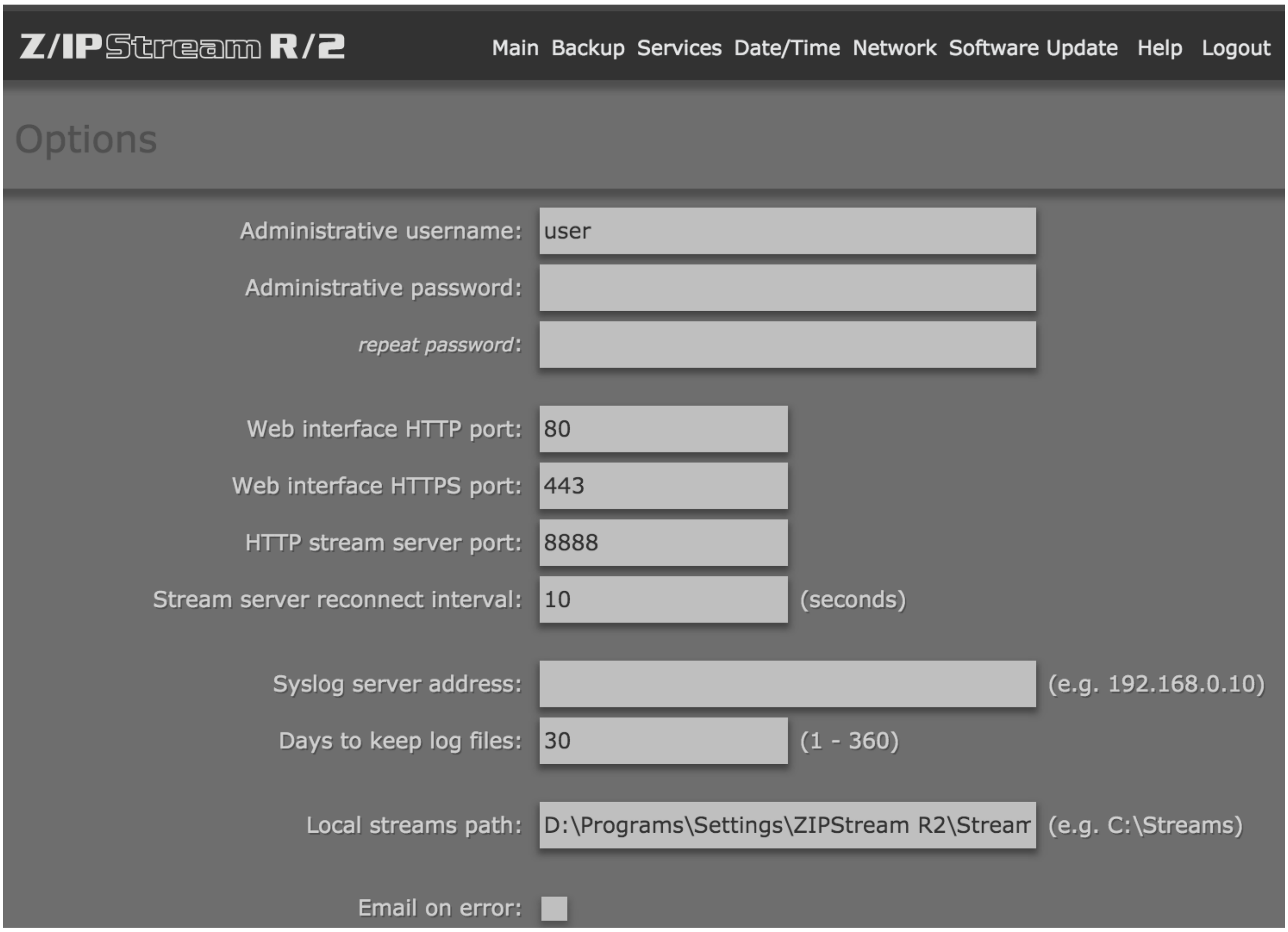
R/2 is a modular platform that can be licensed for various numbers of channels (from 2 up to 8) and types of audio processing (Omnia 3-band or Omnia.9). This screen allows management of the licenses available on the system.

Processing Presets



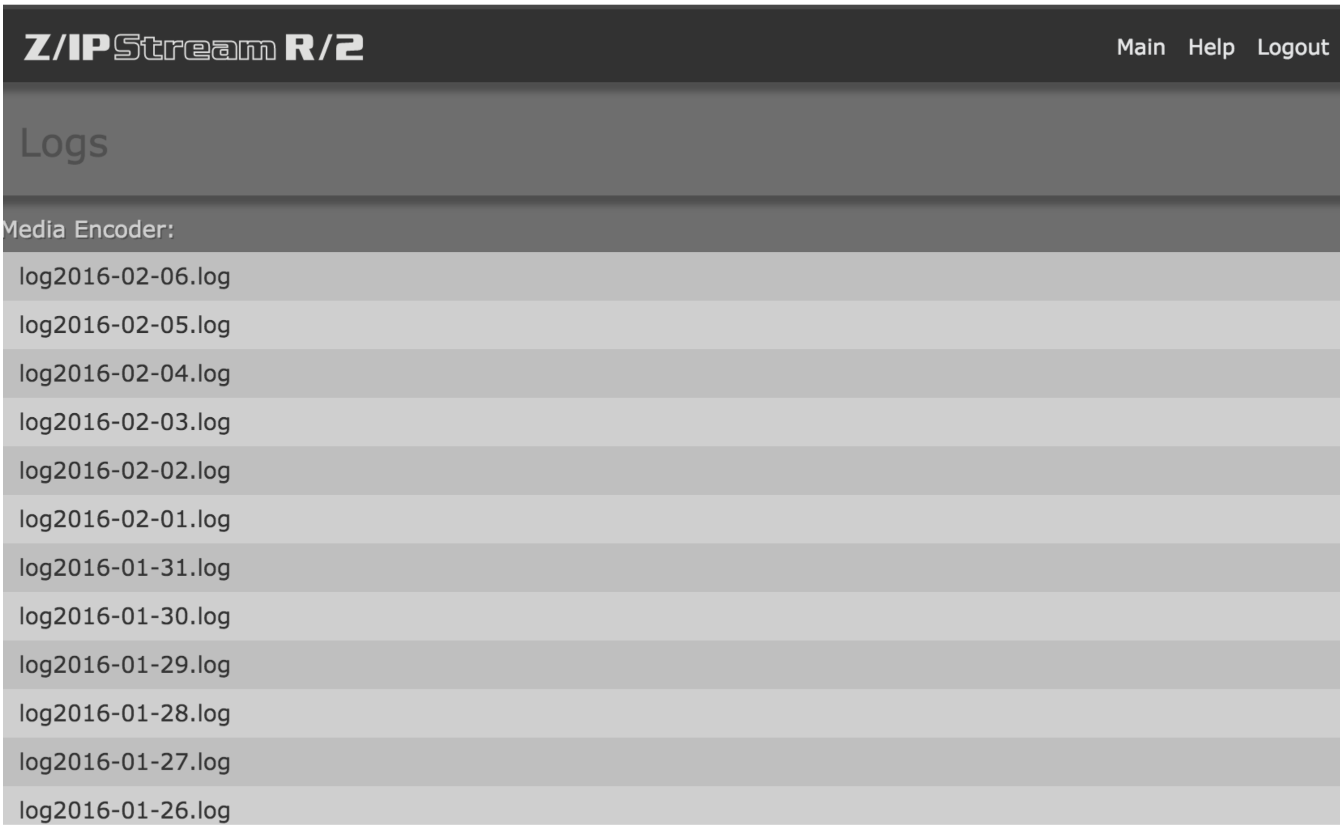
The “Preset Editing” screen allows the management of audio processing presets for the standard Omnia 3 processing. Presets can be added (copied from an existing preset), uploaded from a compatible preset file, or deleted from the unit. Omnia.9 processing presets are managed through NfRemote software.

Options



Includes global configuration options for R/2, including management of background software services, as well as software updates, and backup and restore of system configuration. Network parameters can also be configured from this section.

Logs



Comprehensive logs are kept for diagnostic and other purposes. This screen allows viewing or downloading logs from R/2.

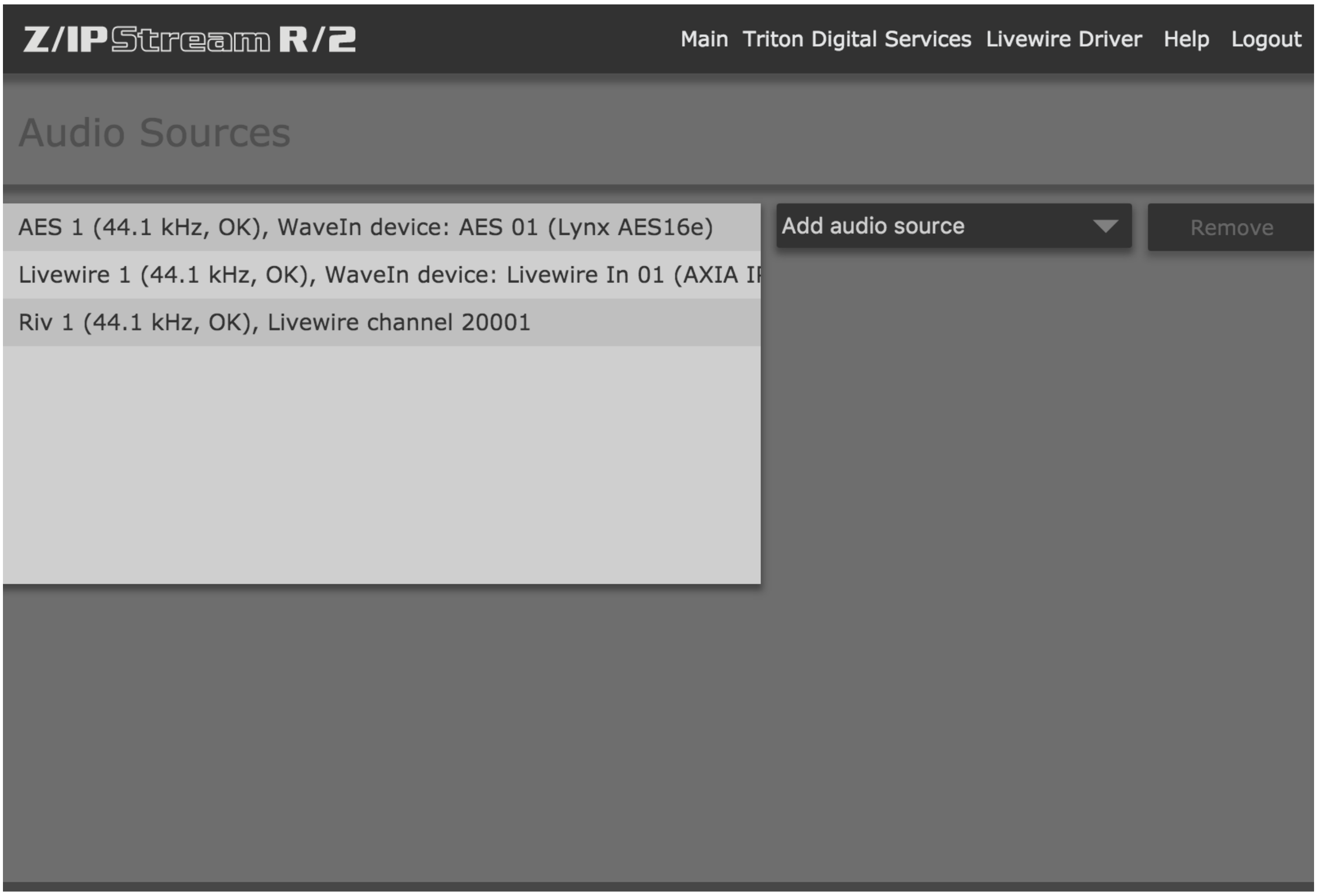
Audio Source Configuration

Overview

This chapter will cover defining the audio sources available for processing and encoding in R/2—one of the most critical steps in configuration after setting the IP addresses and accessing the web GUI.

R/2 is licensed by number of audio sources. Two audio sources are standard with R/2, up to eight can be licensed per unit. Each of these audio sources is available for multiple instances of processing and/or encoding. 3-band Omnia processing is standard, with optional Omnia.9 processing available.

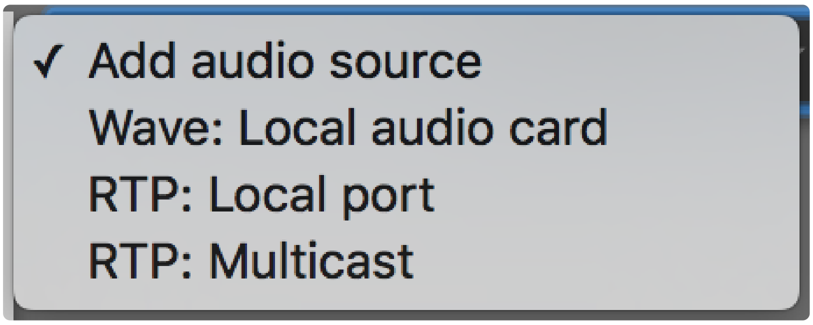
Audio Sources



_Screen%20Shots/06-03-Audio%20Sources.rev2.png

This screen will display the audio sources currently available on the system, and allow you to add new or remove existing sources. To add a new audio source, click on “Add audio source”. To remove an existing source, select it in the list, then click “Remove”. Ensure that the source is no longer in use by any encoding or processing instances before removing it.

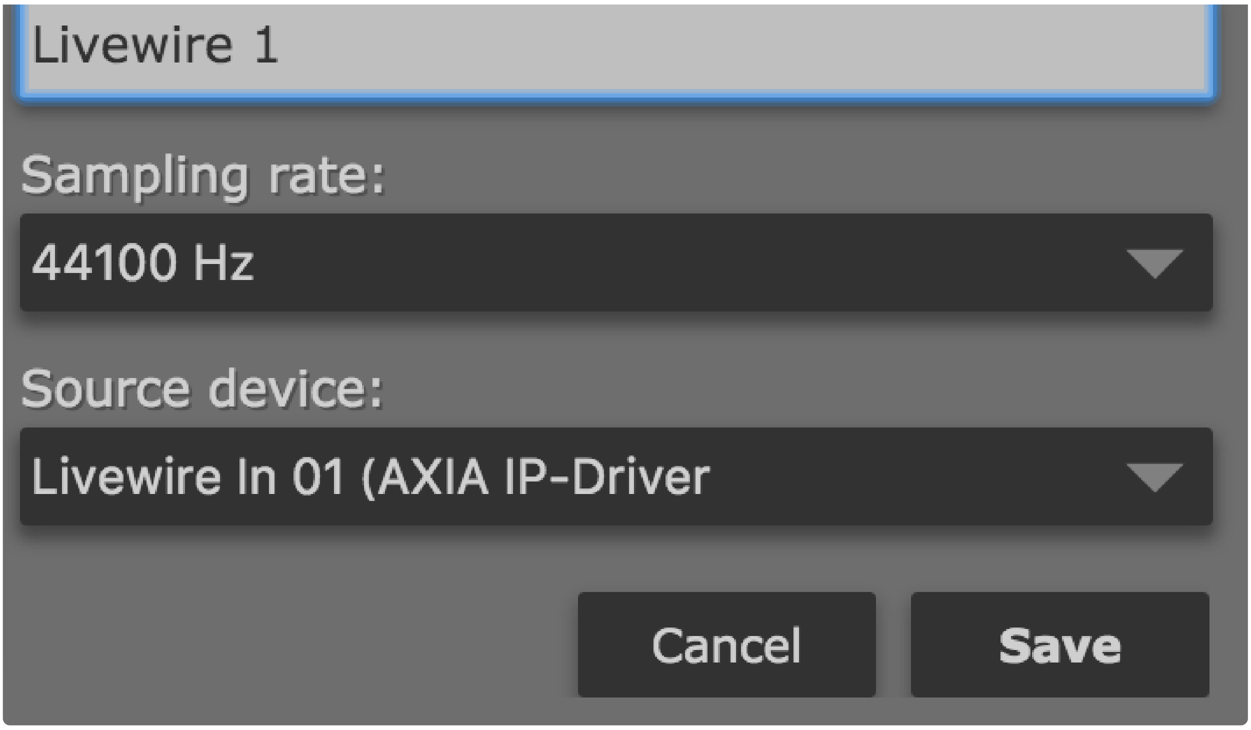
“Add Audio source” menu



_Screen%20Shots/07-02-Available%20Audio%20Sources.png

Select the type of audio source you wish to add.

Wave: Local audio card



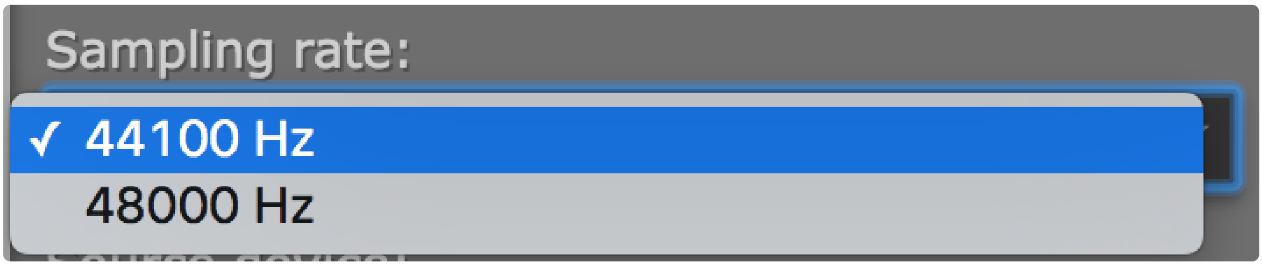
_Screen%20Shots/07-03-Wave%20Audio%20ource.png

Allows input from a local AES/EBU audio source or Livewire audio channel. Selecting this option will open the “Add Wave Audio Source” panel.

Friendly name

Since many audio source names on the system can be very similar, give this audio source a “Friendly” name that will be easier to remember when configuring processing and encoding instances (for example, Livewire 1, AES 1, Classical, Jazz, or whatever else will make it easier to identify in R/2).

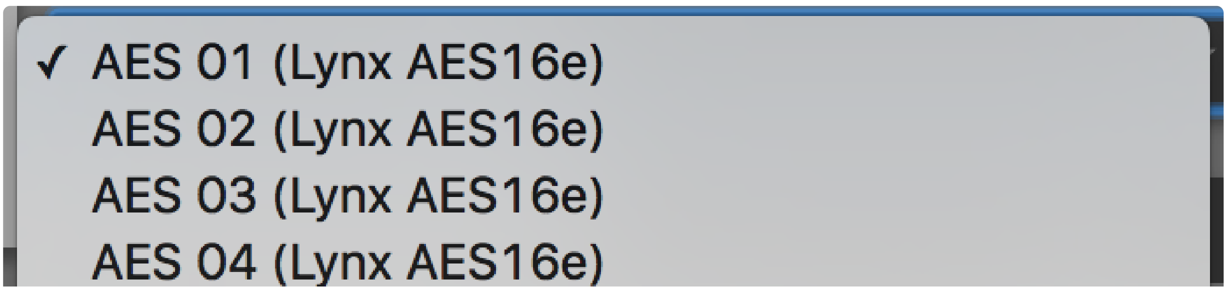
Sampling rate

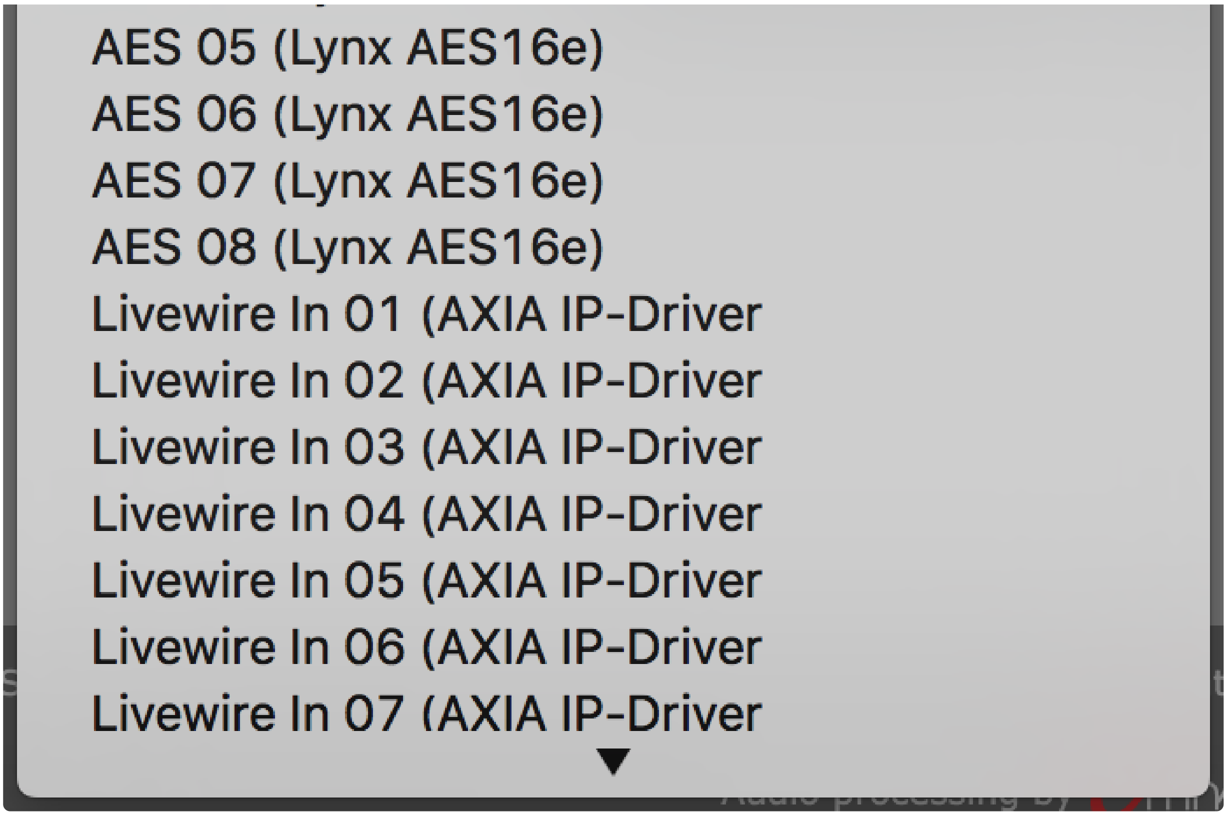


_Screen%20Shots/07-04-Sampling%20Rate.png

This is the sample rate that the streams from this audio channel will be generated at. Available options are 44.1 kHz or 48 kHz. Choose whichever sample rate best suits the streaming application.

Source device





_Screen%20Shots/07-05-Source%20Devices.rev2.png

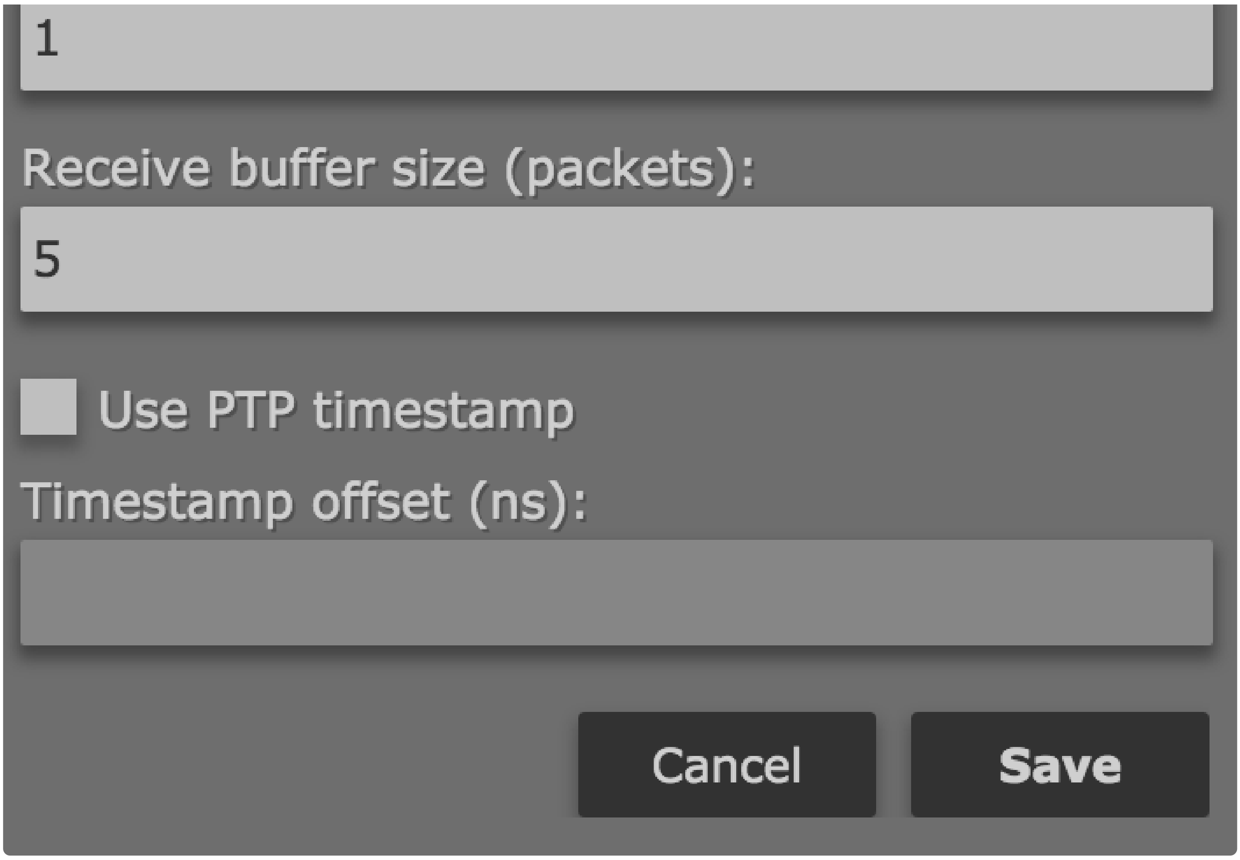
Displays a list of available source devices on the system. There are up to 24 possible Livewire inputs available (defined via the “Livewire Driver” menu) and 8 AES inputs available.

Notes:

- Available Livewire I/O channels are mapped via the “Livewire Driver” menu.
- Audio card port “A” carries channels 01 through 08 (In 1-4) while port “B” carries channels 09 through 16 (In 5-8).

AES Breakout Cable Layout

RTP: Local port



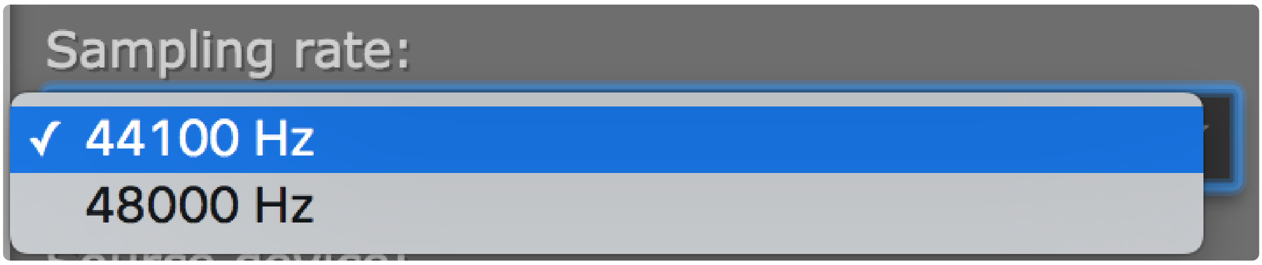
_Screen%20Shots/07-07-RTP%20Audio%20Source.png

Allows input via a unicast RTP stream on a user-defined IP port. Selecting this option will open the “Add RTP Audio Source” panel.

Friendly name

Since many audio source names on the system can be very similar, give this audio source a “Friendly” name that will be easier to remember when configuring processing and encoding instances (for example, Livewire 1, AES 1, Classical, Jazz, or whatever else will make it easier to identify in R/2).

Sampling rate



_Screen%20Shots/07-04-Sampling%20Rate.png

This is the sample rate that the streams from this audio channel will be generated at. Available options are 44.1 kHz or 48 kHz. Choose whichever sample rate best suits the streaming application.

RTP Port

Enter the port that will receive this RTP audio stream.

Receive Buffer Size (Packets)

Allows precise adjustment of the incoming RTP audio stream buffer for time alignment, or other purposes.

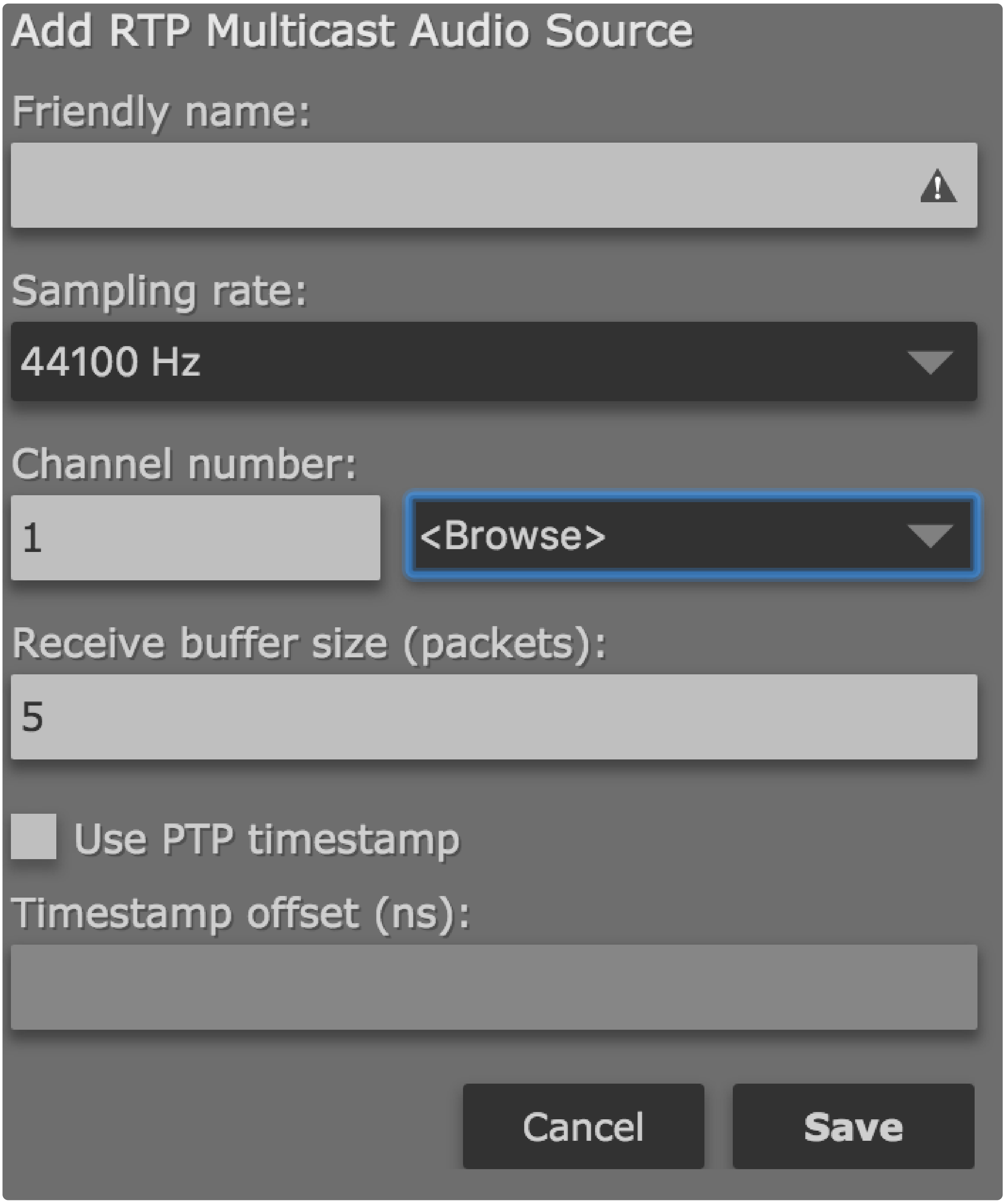
Use PTP timestamp

Specifies use of an IEEE 1588 PTP (Precision Time Protocol) timestamp for stream synchronization purposes.

Timestamp offset (ns)

Allows an offset to the incoming IEEE 1588 PTP timestamp to be specified.

RTP: Multicast



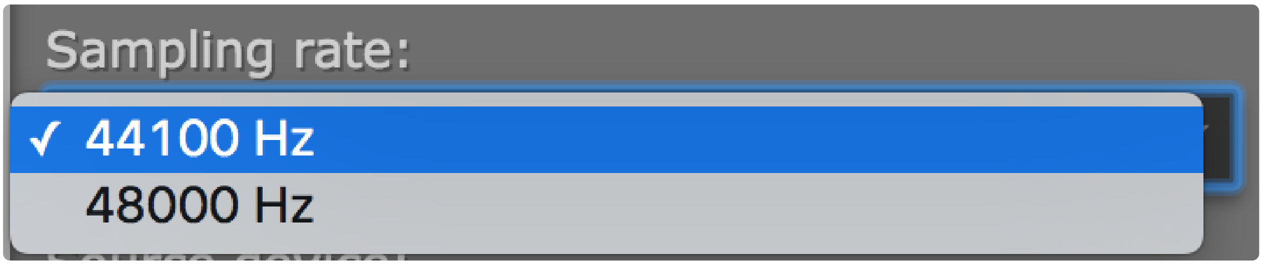
_Screen%20Shots/07-08-RTP%20Multicast%20Audio%20Source.png

Allows input via a multicast RTP stream (including Livewire+/AES67). Selecting this option will open the "Add RTP Multicast Audio Source" panel.

Friendly name

Since many audio source names on the system can be very similar, give this audio source a "Friendly" name that will be easier to remember when configuring processing and encoding instances (for example, Livewire 1, AES 1, Classical, Jazz, or whatever else will make it easier to identify in R/2).

Sampling rate



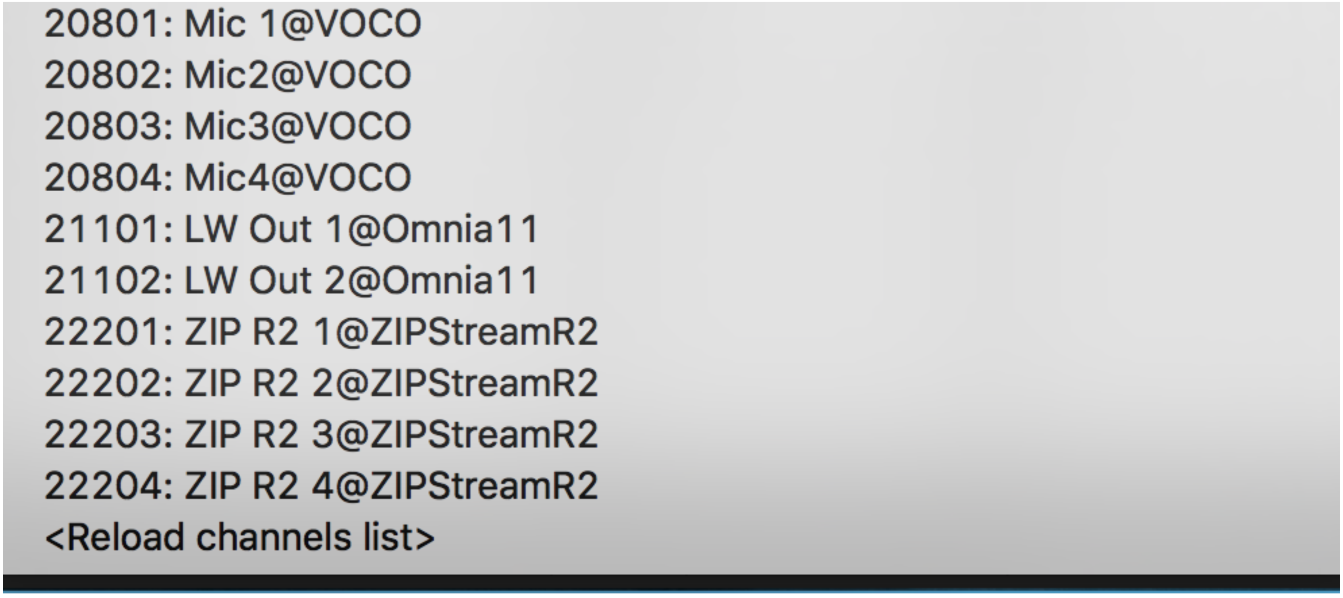
_Screen%20Shots/07-04-Sampling%20Rate.png

This is the sample rate that the streams from this audio channel will be generated at. Available options are 44.1 kHz or 48 kHz. Choose whichever sample rate best suits the streaming application.

Channel number

Specifies the Livewire or other Audio over IP network channel number associated with the desired RTP multicast stream.

Browse



_Screen%20Shots/Screen%20Shot%202016-02-06%20at%201.28.04%20PM.png

Displays a list of multicast channels and streams available on the network. Select a channel from this list to populate the "Channel number" field. Select "Reload channels list" to refresh this list of available channels.

Note: Both this method and the "Livewire Driver" method described later in this chapter can be used to define Livewire channels as available audio sources.

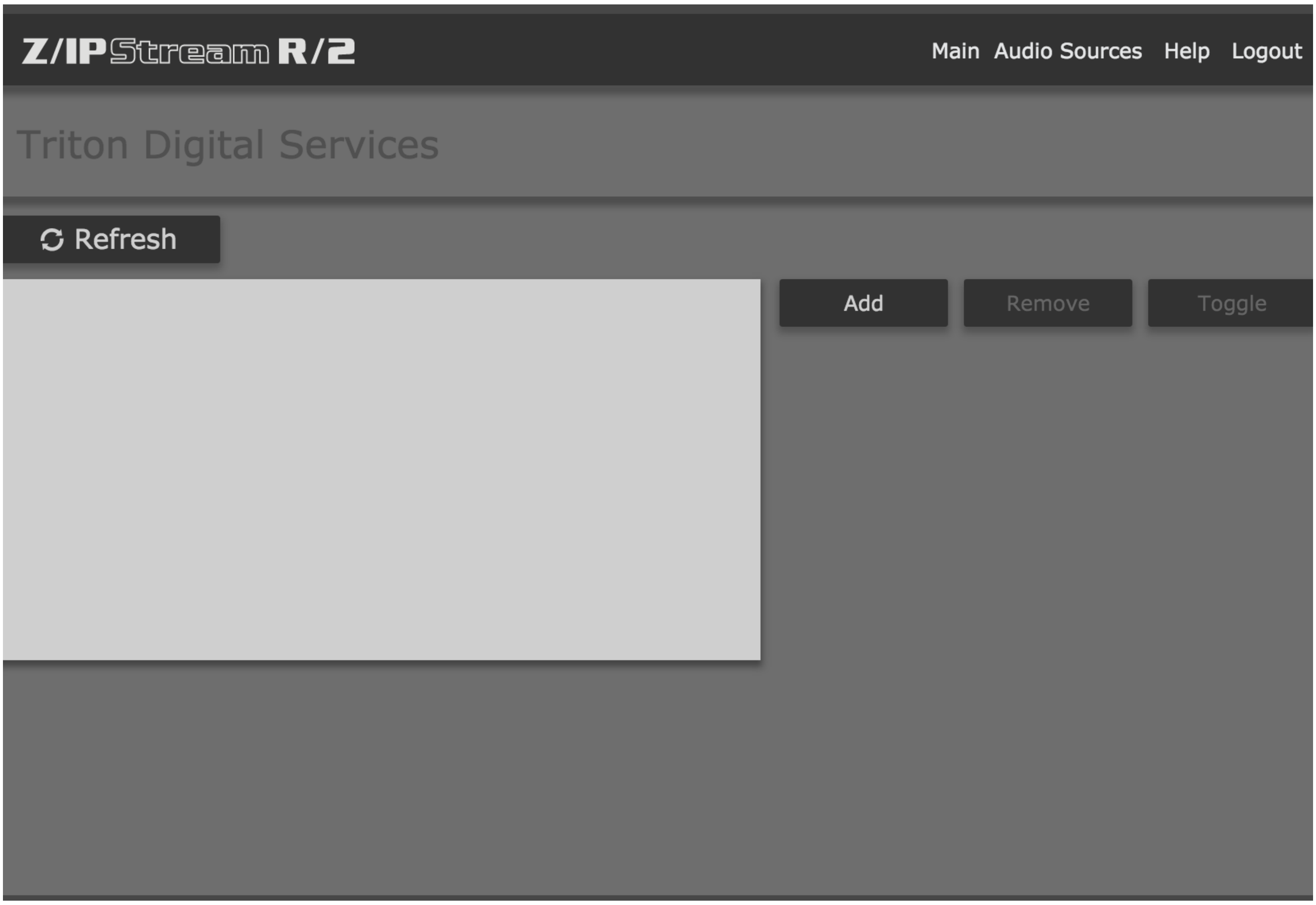
Use PTP timestamp

Specifies use of an IEEE 1588 PTP (Precision Time Protocol) timestamp for stream synchronization purposes.

Timestamp offset (ns)

Allows an offset to the incoming IEEE 1588 PTP timestamp to be specified.

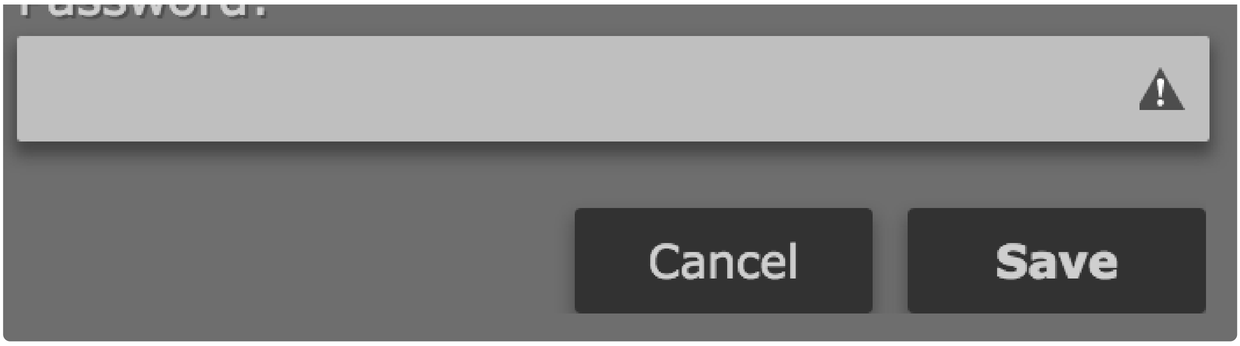
Triton Digital Services



_Screen%20Shots/07-09-Triton%20Digital.rev2.png

R/2 offers a direct streamlined interface to Triton Digital Services for streaming and ad replacement. Stations can be added to R/2 via the “Triton Digital Services” screen. This screen also allows stations to be removed from R/2, or toggled on and off.

Adding a Triton Digital Services Station

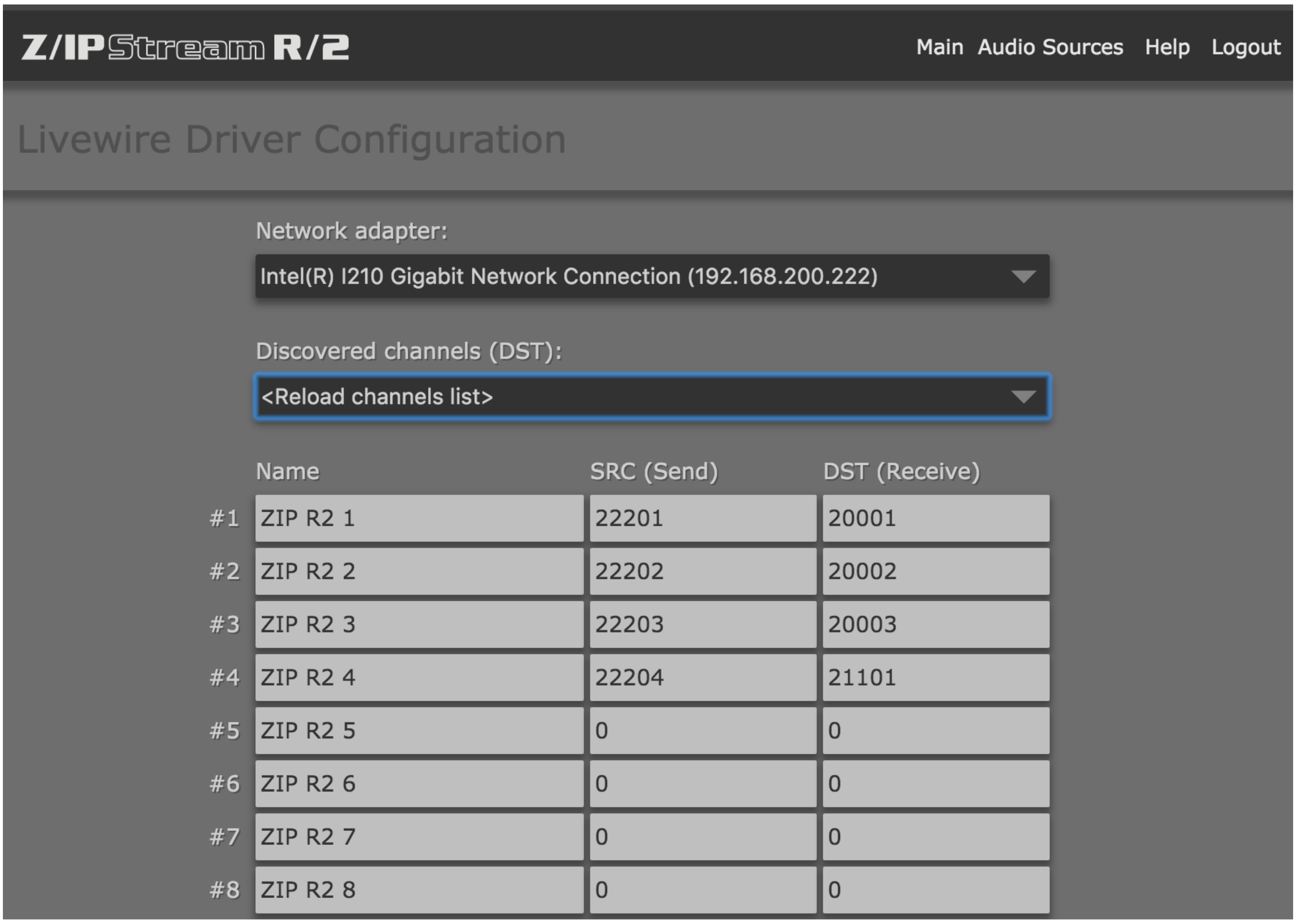


Enter the Triton Digital Services login and password for the station you wish to add and click “Save”.

Click “Remove” to remove a station from the list, or “Toggle” to enable and disable a station.

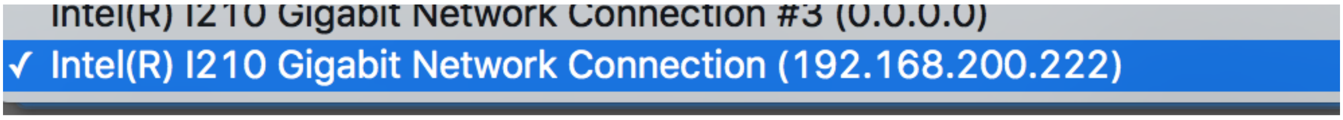
Click “Main” to return to the main R/2 “Control Panel” screen, or “Audio Sources” to return to audio source configuration.

Livewire Driver



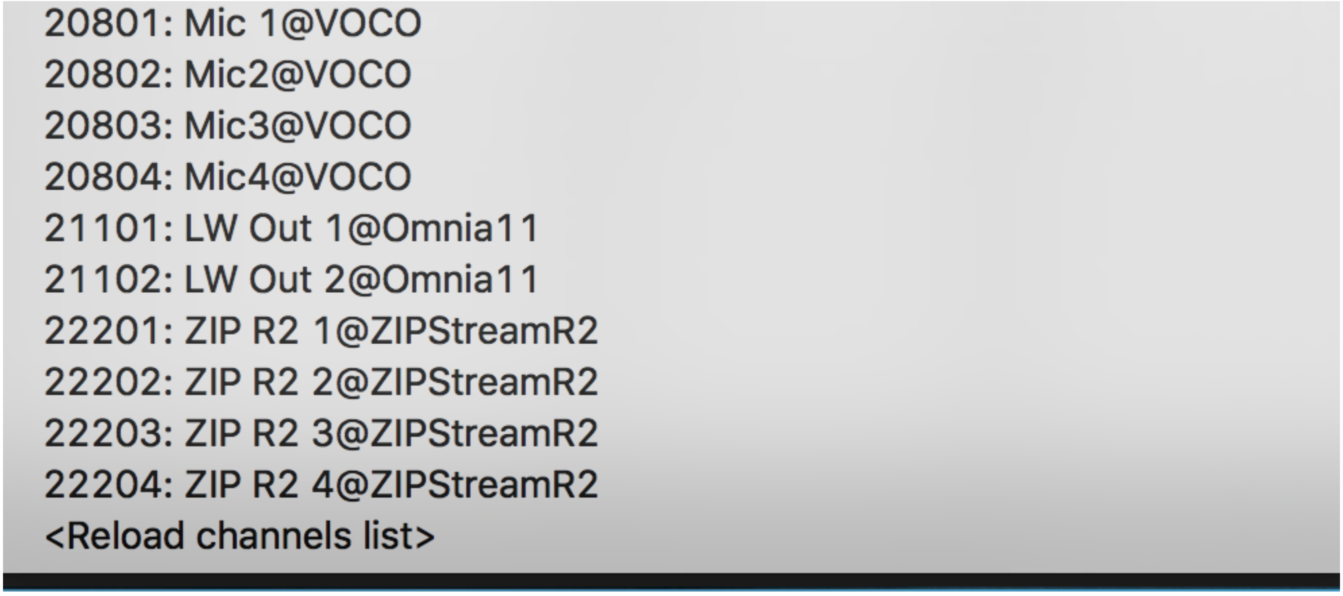
The “Livewire Driver Configuration” screen allows configuration of the integrated Livewire IP Audio Driver and provides a direct mechanism for defining the Livewire channel names and numbers used by R/2. Up to 24 Livewire channels are available and correspond to the 24 Livewire channels available via the “Source Device” menu under “Add Wave Audio Source”.

Network Adapter



Select the desired network interface for to be used for Livewire traffic. The IP addresses listed for these two interfaces should correspond to the ones assigned for NIC 1 and NIC 2.

Discovered channels (DST)



Displays a list of Livewire channels available on the network. This list is for reference only. Selecting channels in this list will not directly populate the channel number fields. Select “Reload channels list” to refresh this list of available channels.

Note: If no channels are displayed in this list, first verify that one of the NIC interfaces is connected to the Livewire network and properly configured. Second, attempt to restart the R/2 service (not the entire R/2 unit, just the R/2 “service” via the “Services” menu under “Options”—Refer to Chapter 14 for details)

Name

Enter a name to uniquely identify this channel on the Livewire network. It will appear on other Livewire devices as name@ZIPStreamR2 (see the above channel list for an example).

SRC (Send)

Enter a unique Livewire network channel number from 1 to 32767 that each IP audio driver channel will SEND to as a SOURCE. Enter “0” to disable sending to the Livewire network from a channel.

Note: Each device MUST use unique channel numbers when sending audio to the Livewire network. It is recommended that you develop a channel numbering scheme to ensure that every device on the network is always assigned a unique block of channel numbers. For example, the last octet of the IP address with the channel number appended is one such example. In this case, the channel numbers assigned to this particular R/2 would be 22201-22224.

DST (Receive)

This section allows R/2 to RECEIVE audio as a DESTINATION FROM the Livewire network. Using the

“Discovered Channels” list, enter a valid channel number from the Livewire network to assign it to an audio driver channel. Audio from this network channel will appear on the corresponding Livewire driver channel in the “Source Device” menu under “Add Wave Audio Source”.

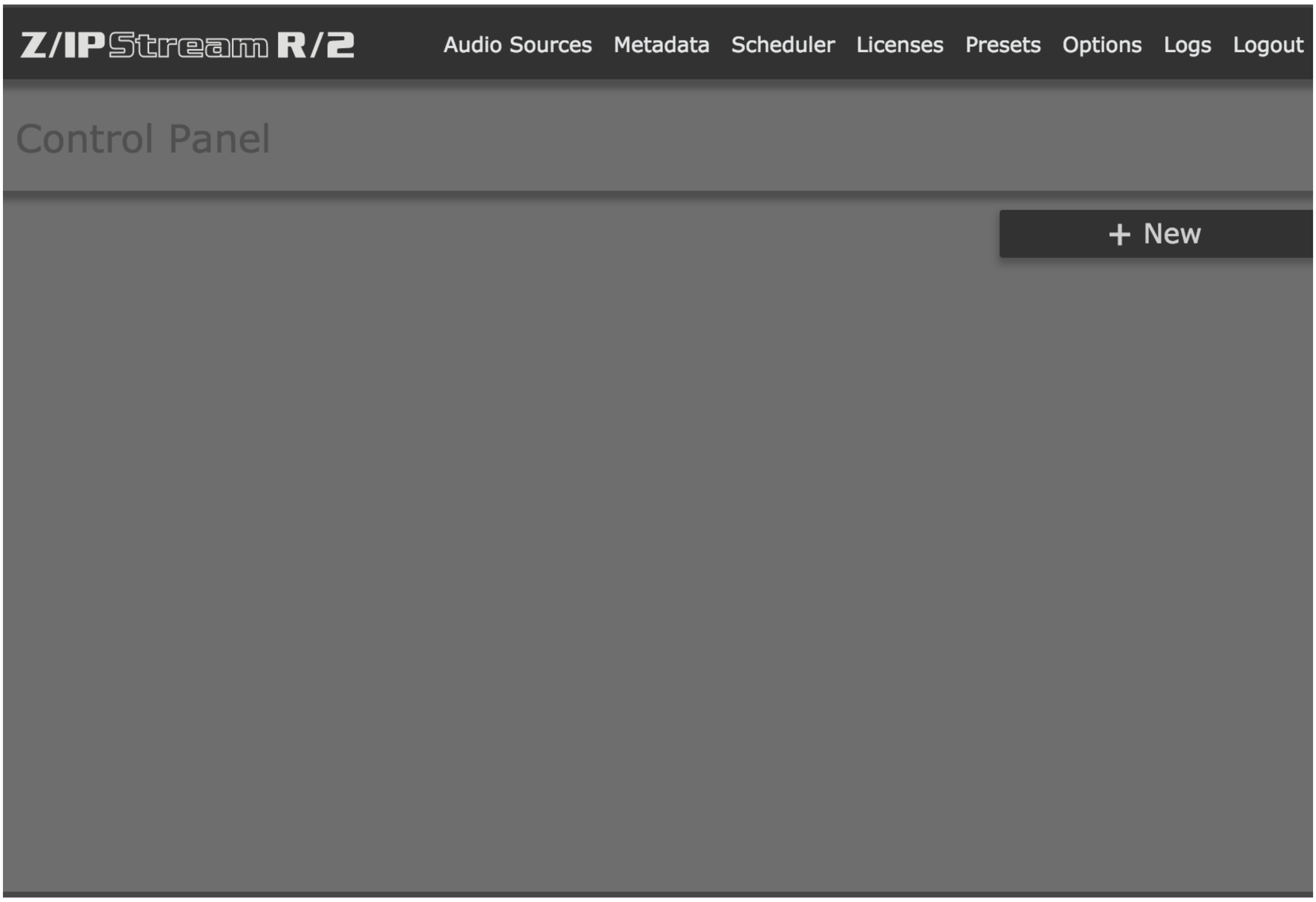
Click “Main” to return to the main R/2 “Control Panel” screen, or “Audio Sources” to return to audio source configuration.

Processing and Encoding

Overview

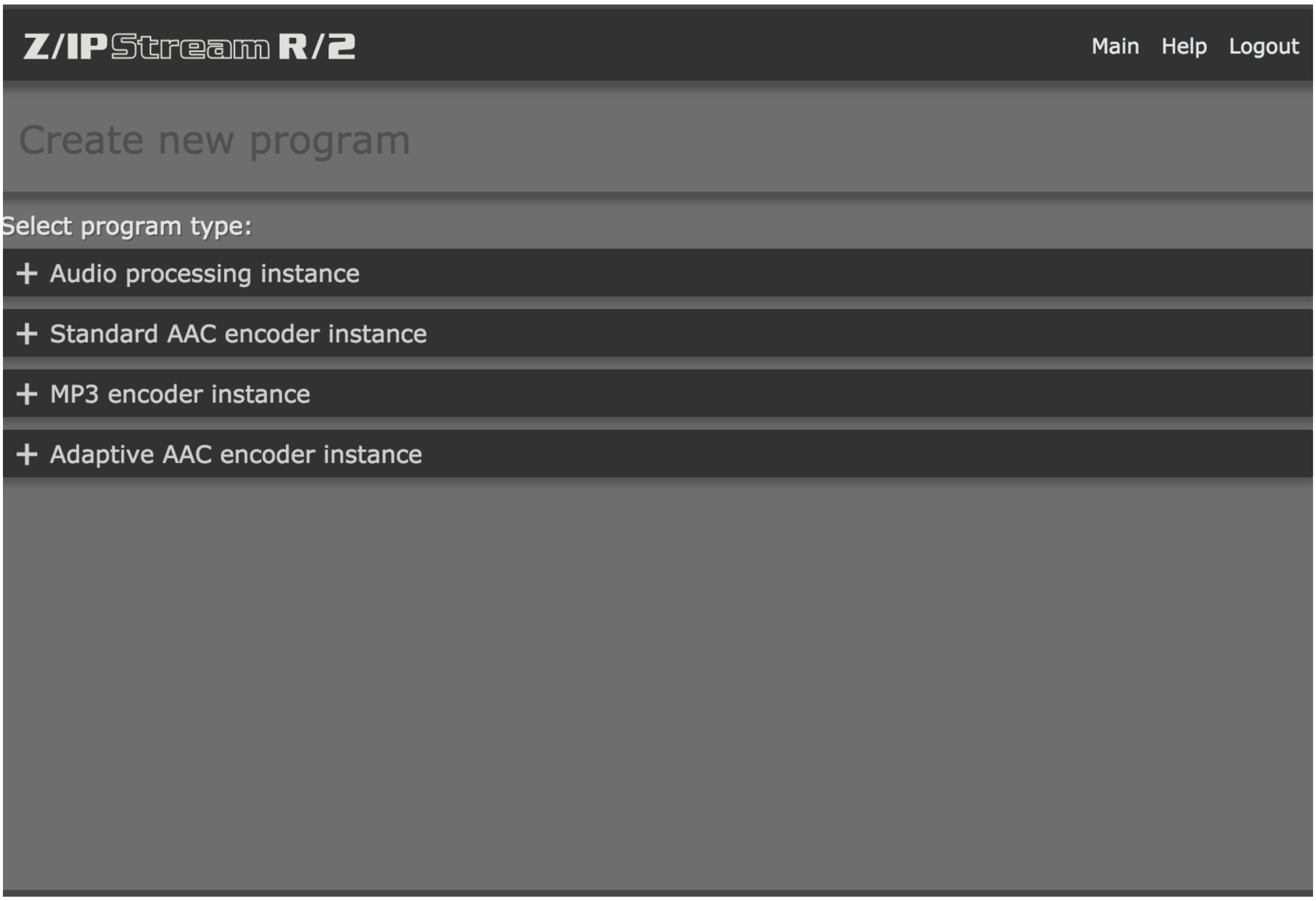
Once the audio sources have been defined, the next step is to configure the processing and encoding instances that will use those sources. Each audio source can be used for audio processing only, encoding only, or any combination of the two. Available encoding modes include AAC, MP3, and adaptive AAC. Multiple codecs, bitrates, and encoding destinations are supported for each audio source. This is potentially one of the most complex yet extremely powerful configuration sections within R/2.

Defining Processing and Encoding Instances



From the Main “Control Panel” window, click “New”.

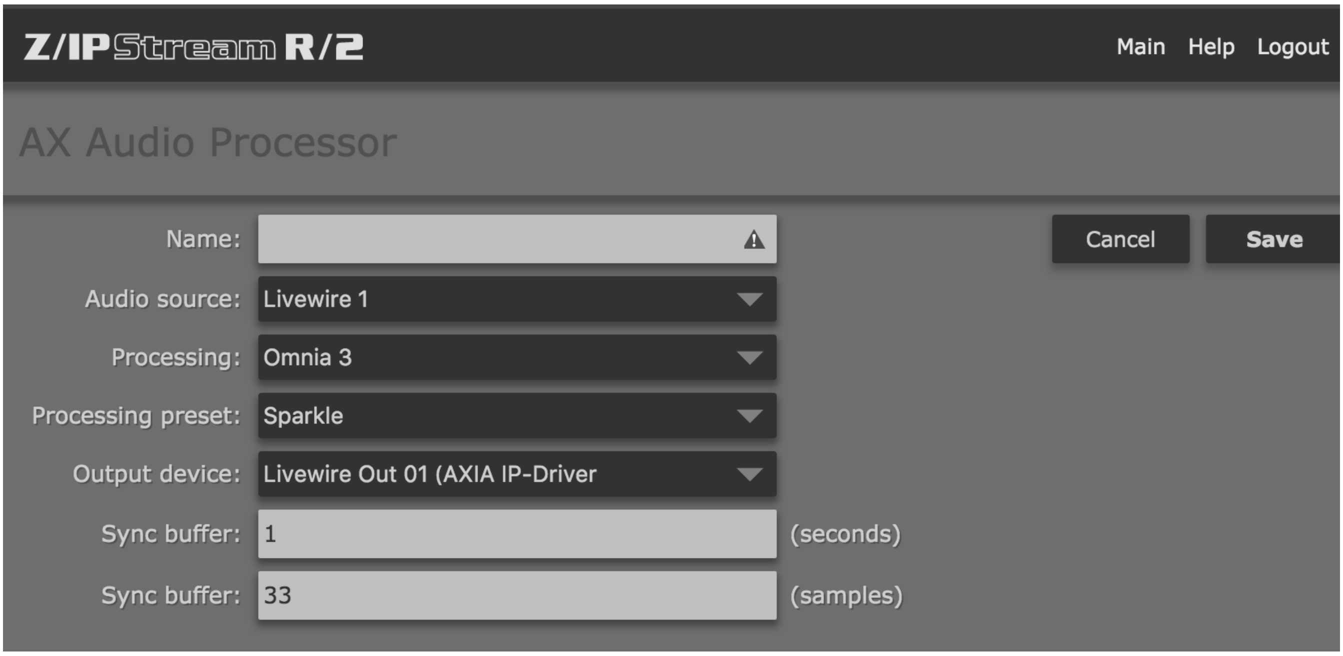
Create New Program



The “Create New Program” window allows selection of the “program type” to be defined. Click on the program type that you wish to create.

Note: Multiple processing and/or encoding instances (program types) are supported for each of the audio sources defined previously (see Chapter 7). Encoder instances can also contain multiple destinations and bitrates for encoding.

Audio processing instance

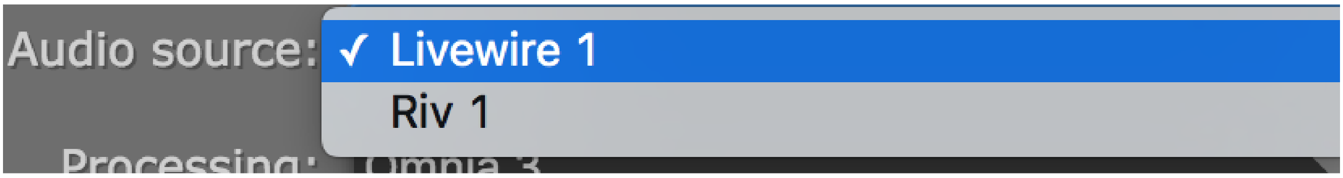


While R/2 is designed primarily for streaming applications, there may be cases where you wish to process an audio signal without encoding a stream for a server. The “Audio Processing Instance” program allows processing an audio I/O without encoding. This can be useful for “utility” processing of audio around a facility, or as “backup” to a main audio processor, particularly when using the optional Omnia.9 processing.

Name

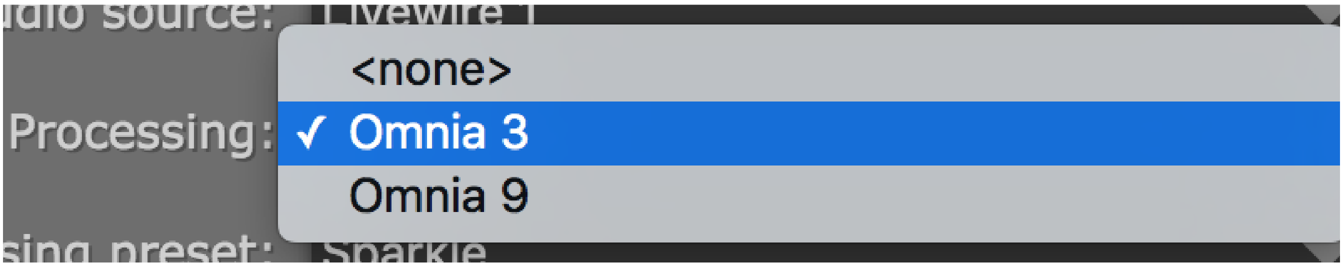
Enter a “friendly name” for this processing instance to easily identify it for status and control from the main “control panel” screen (i.e. Kxxx-Backup Processing).

Audio Source



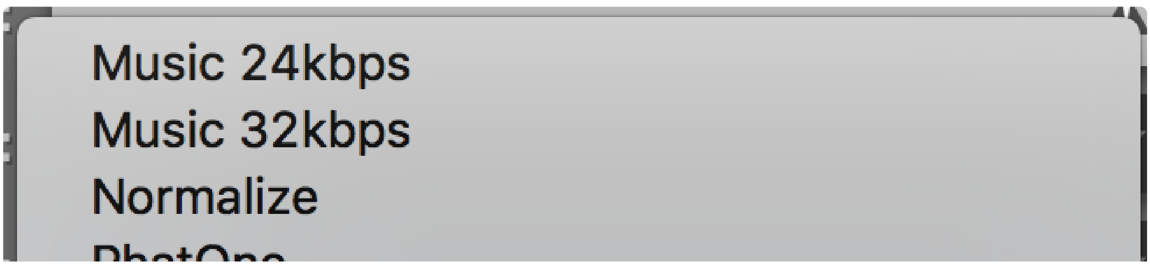
Select one of the previously defined audio sources as the input to this processing instance (see Chapter 7). Note that these sources can be used in multiple processing or encoding instances simultaneously.

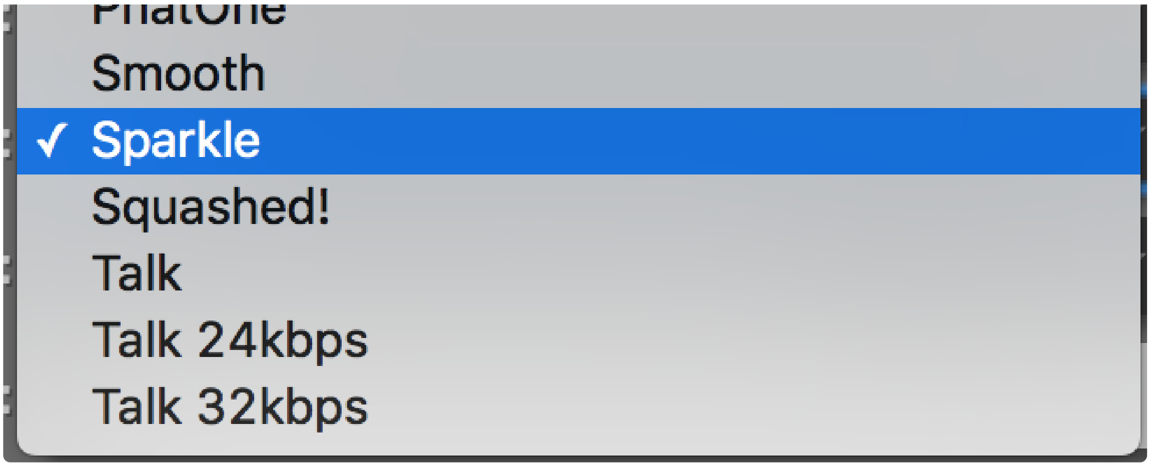
Processing



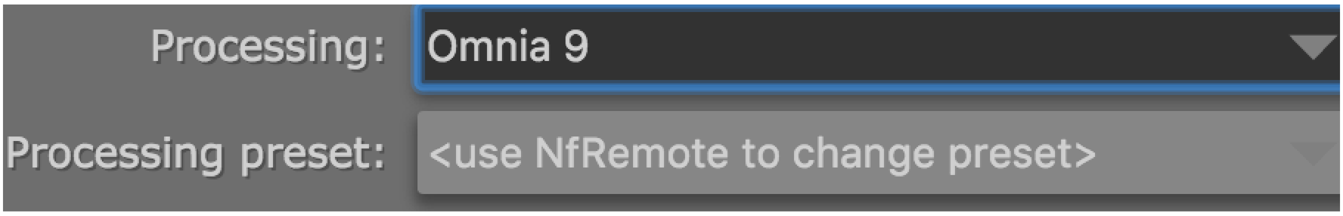
Selects between Omnia 3 and optional Omnia.9 processing (or no processing at all, should you simply want to route audio through this instance without processing). Up to 8 channels of Omnia.9 processing can be licensed per R/2. When you have reached the maximum number of Omnia.9 licenses available (or if Omnia.9 processing is not licensed) this option will not be displayed. See Chapter 16 for more information on Omnia.9 processing.

Processing preset



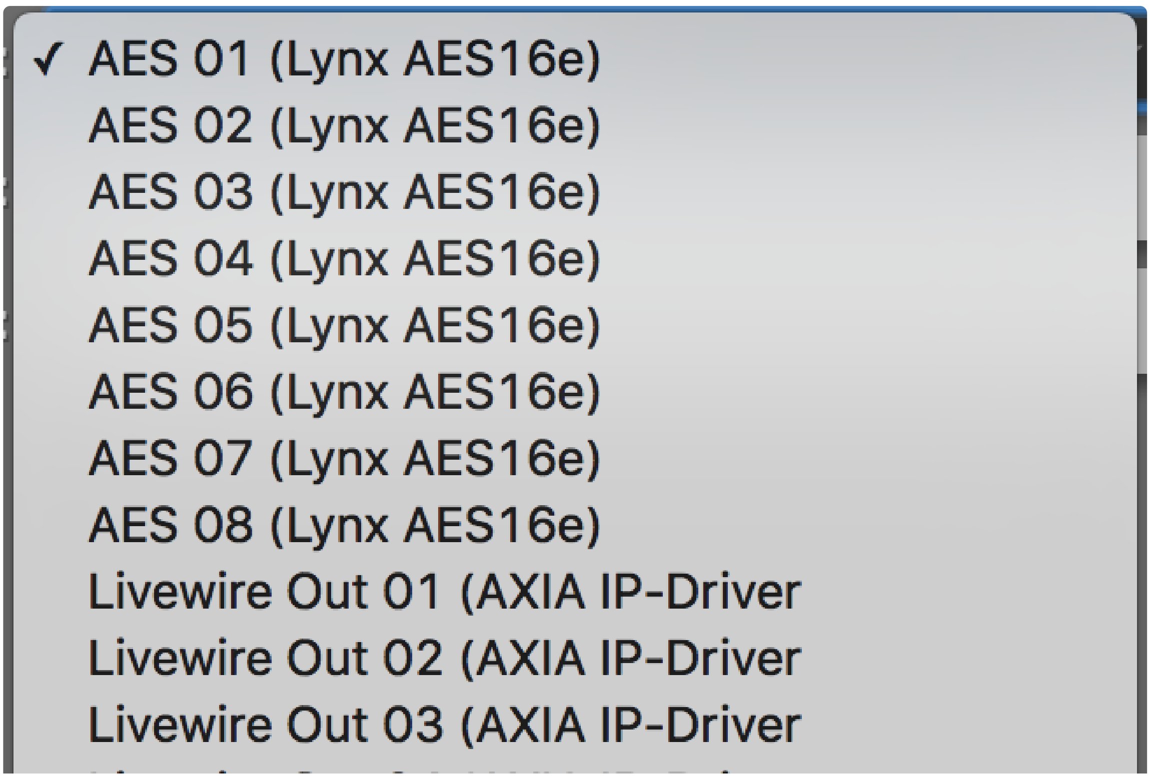


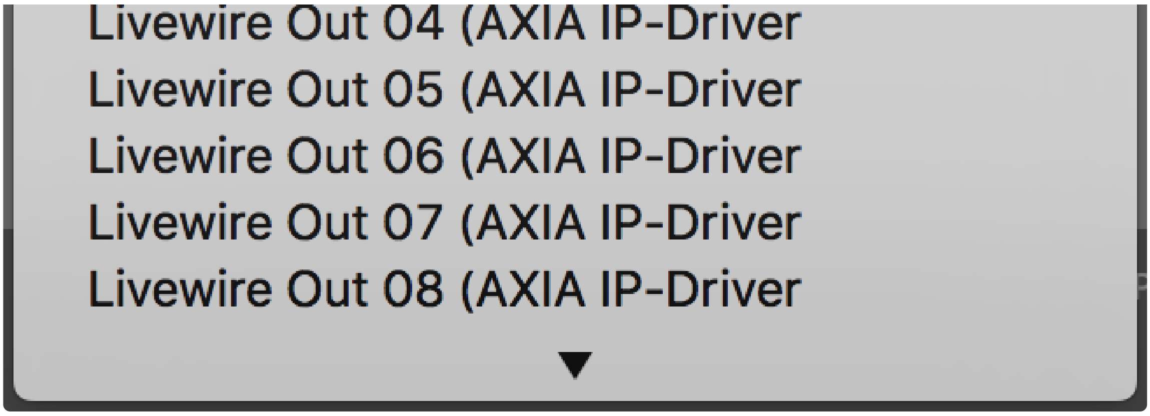
Selects the processing preset to be used for this instance. When Omnia 3 processing is selected, the list of available presets will appear. These presets are not format specific, though there are bitrate specific presets for “Music” and “Talk”. Experiment with various presets to find one that works best for your programming format and application. Presets can be copied and modified to create a sound customized exactly to your needs. See Chapter 9 to learn how to “fine tune” the Omnia 3 processing, and Chapter 13 for more information on Omnia 3 preset management.



If optional Omnia.9 processing is enabled, the preset must be selected via NfRemote software due to the powerful processing tools and test instrumentation available through Omnia.9. See Chapter 16 for more details.

Output device





The available output destinations for audio from this instance are listed here (up to 24 Livewire outputs, and 8 local AES outputs). Select the output destination for this processing instance. Livewire channels are configured through the "Livewire Driver Configuration" interface. See Chapter 7 for more details.

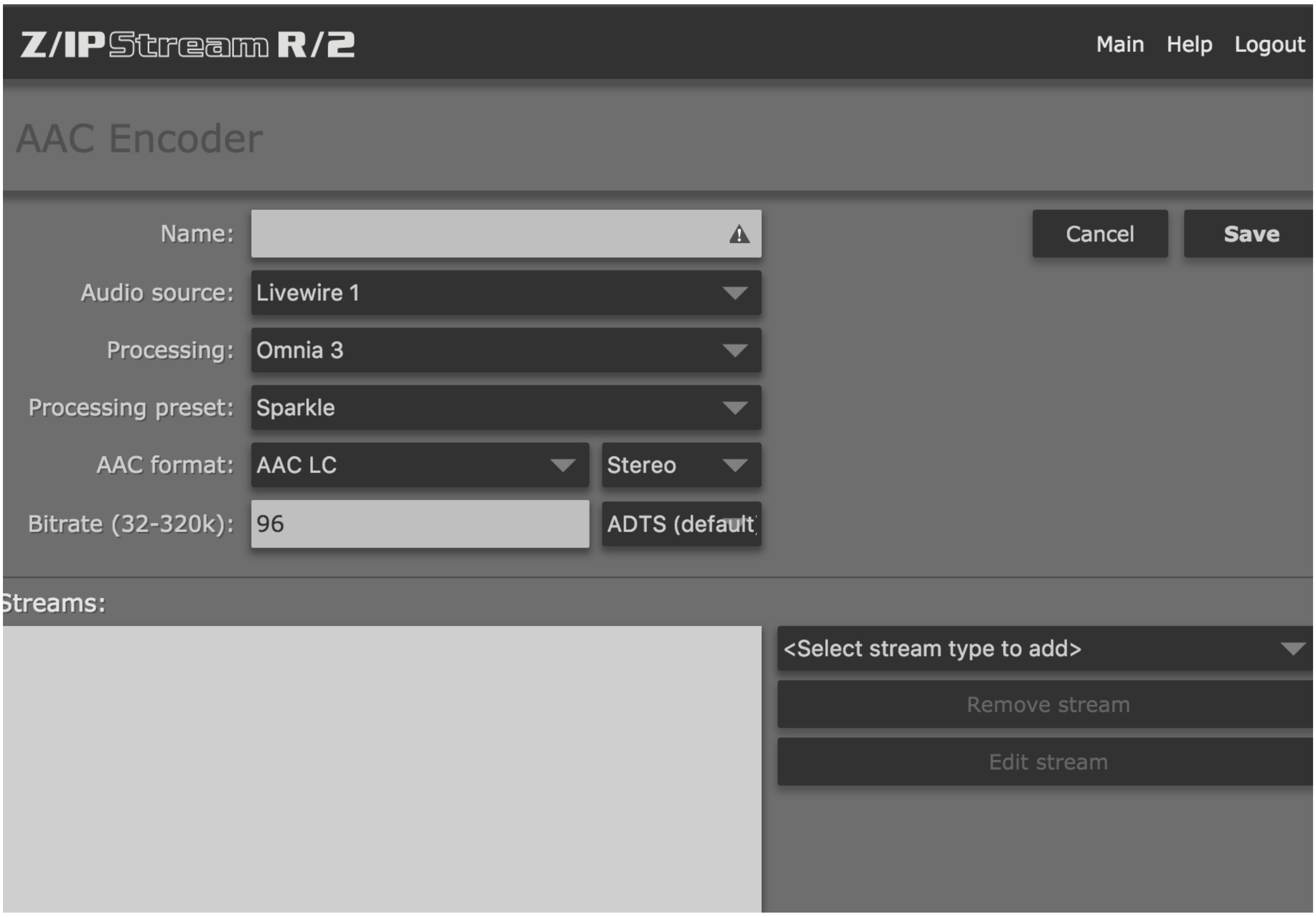
Sync buffer (Samples, Seconds)

These values will allow adjustment of the synchronization buffer between input and output. It is recommended that you leave these at the defaults unless advised by technical support. Improper adjustment can cause buffering issues.

Cancel/Save

Once you have finished configuring the processing instance, click "Save" to save it, or "Cancel" to return to the Control Panel screen without saving changes.

Standard AAC Encoder Instance

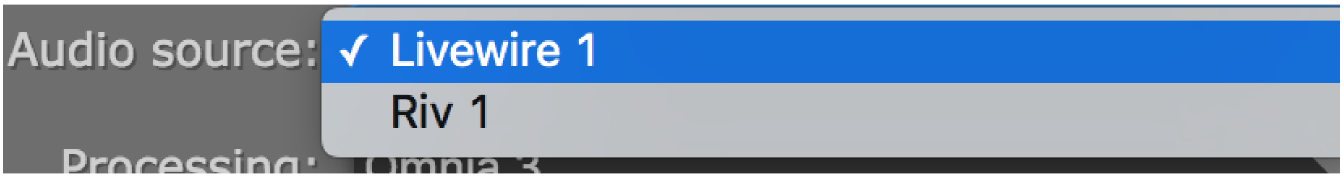


The Standard AAC Encoder instance accepts audio from a source, encodes it to AAC, and then passes the encoded stream along to one or more servers that you specify. Remember that that the encoded stream can be sent simultaneously to multiple servers, which will then replicate the stream to the client. Multiple AAC bitrates, formats, or processing parameters are supported from the same audio source by creating multiple AAC encoder instances.

Name

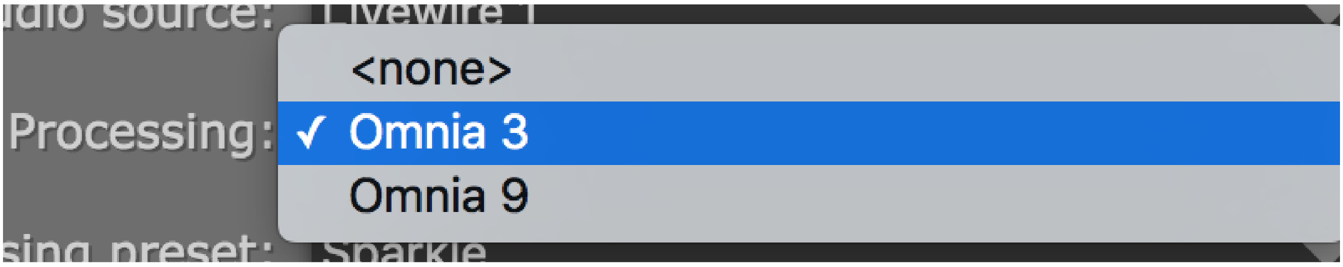
Enter a “friendly name” for this encoder instance to easily identify it for status and control from the main “control panel” screen (i.e. Kxxx-AAC Stream 1).

Audio Source



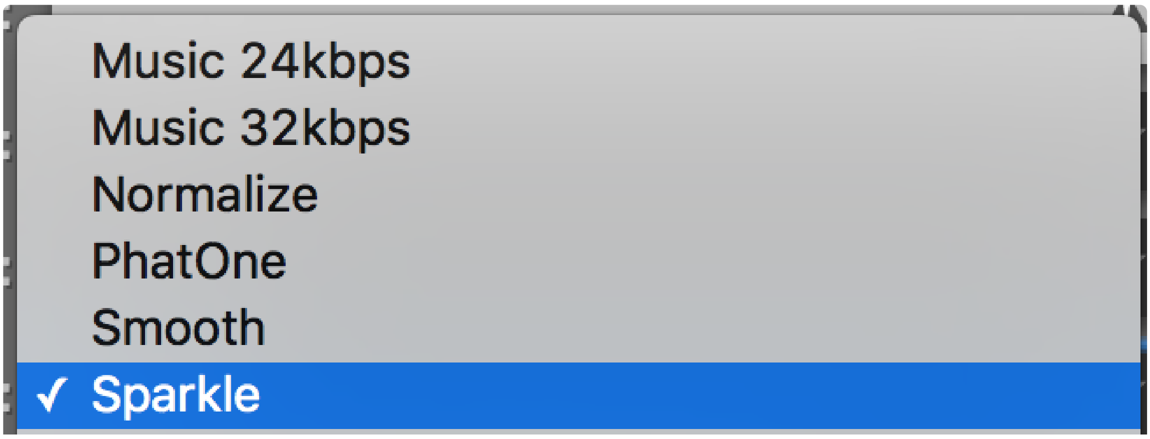
Select one of the previously defined audio sources as the input to this processing instance (see Chapter 7). Note that these sources can be used in multiple processing or encoding instances simultaneously.

Processing

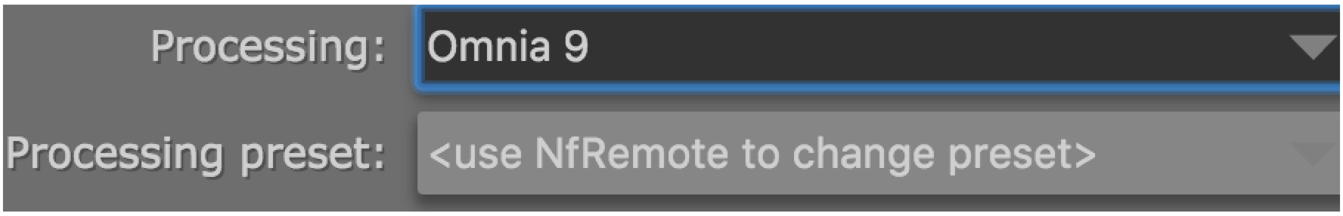


Selects between Omnia 3 and optional Omnia.9 processing (or no processing at all, should you simply want to route audio through this instance without processing). Up to 8 channels of Omnia.9 processing can be licensed per R/2. When you have reached the maximum number of Omnia.9 licenses available (or if Omnia.9 processing is not licensed) this option will not be displayed. See Chapter 16 for more information on Omnia.9 processing.

Processing preset

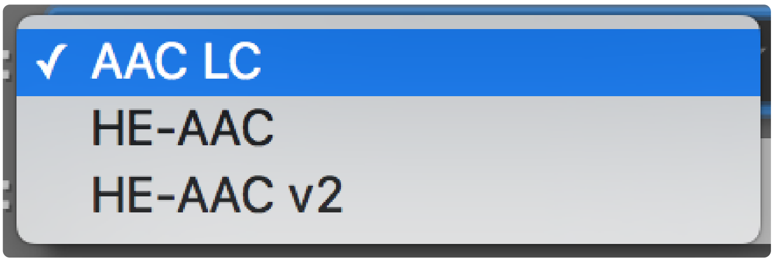


Selects the processing preset to be used for this instance. When Omnia 3 processing is selected, the list of available presets will appear. These presets are not format specific, though there are bitrate specific presets for “Music” and “Talk”. Experiment with various presets to find one that works best for your programming format and application. Presets can be copied and modified to create a sound customized exactly to your needs. See Chapter 9 to learn how to “fine tune” the Omnia 3 processing, and Chapter 13 for more information on Omnia 3 preset management.



If optional Omnia.9 processing is enabled, the preset must be selected via NfRemote software due to the powerful processing tools and test instrumentation available through Omnia.9. See Chapter 16 for more details.

AAC Format



There are several different “flavors” of AAC. The most common form (and the default for this instance) is AAC-LC (Advanced Audio Coding-Low Complexity). HE-AAC is “High Efficiency” AAC. It is designed for bitrates between 24 and 96 kbps. An “improved” version of HE-AAC (HE-AAC v2) was developed for lower bitrates between 14 and 56 kbps. HE-AAC v2 also offers improved stereo performance at lower bitrates. It is worth noting while HE-AAC v2 is widely supported, it may not be compatible with some players.

Note: If in doubt, choose AAC-LC. This is the most common form of AAC encoding, though the HE (High Efficiency) variants are becoming more common for lower bitrate applications.

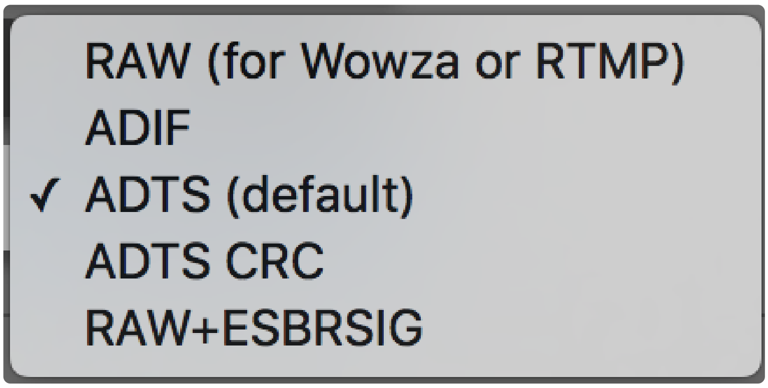
Mono/Stereo

Select whether this stream will be encoded in Mono or Stereo. Mono should provide a mono sum of the incoming audio. This setting will be locked to stereo for HE-AAC v2.

Bitrate

Enter the bitrate for this stream. The valid bitrate range will change depending on which AAC format is selected.

Stream Output Format



Selects the output file format for the stream. Even though the streams are not a “file” per-se, this setting controls how the AAC encoded audio is encapsulated. The default is generally sufficient for most applications, but other settings may be required for different server platforms (such as Wowza or other RTMP server). Although multiple server destinations are supported per instance, only one output format can be selected. Use another instance if you need to send a different format to another server (i.e. HTTP server and Wowza/RTMP simultaneously).

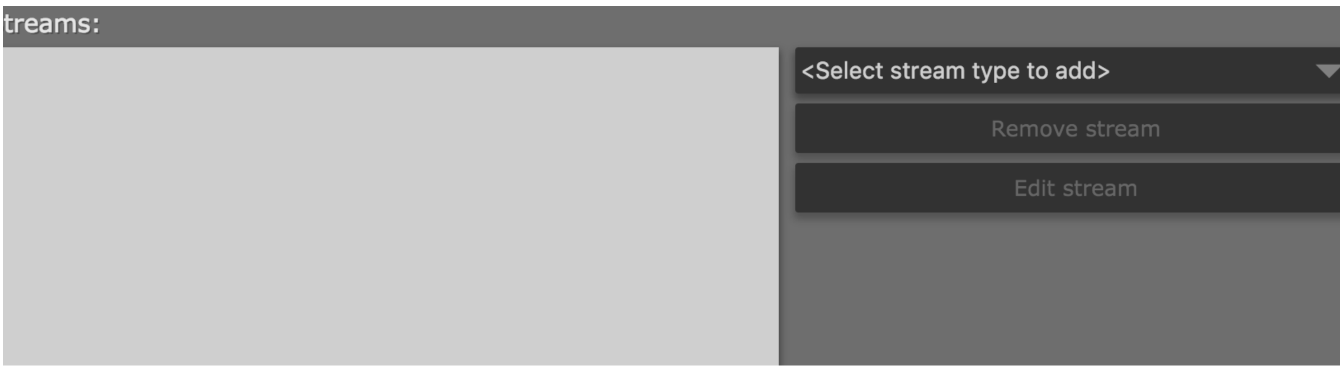
RAW – Send unformatted (raw encoded) audio to your stream. Use this option with RTMP streams.

ADIF - Audio Data Interchange Format, a file format to exchange AAC data.

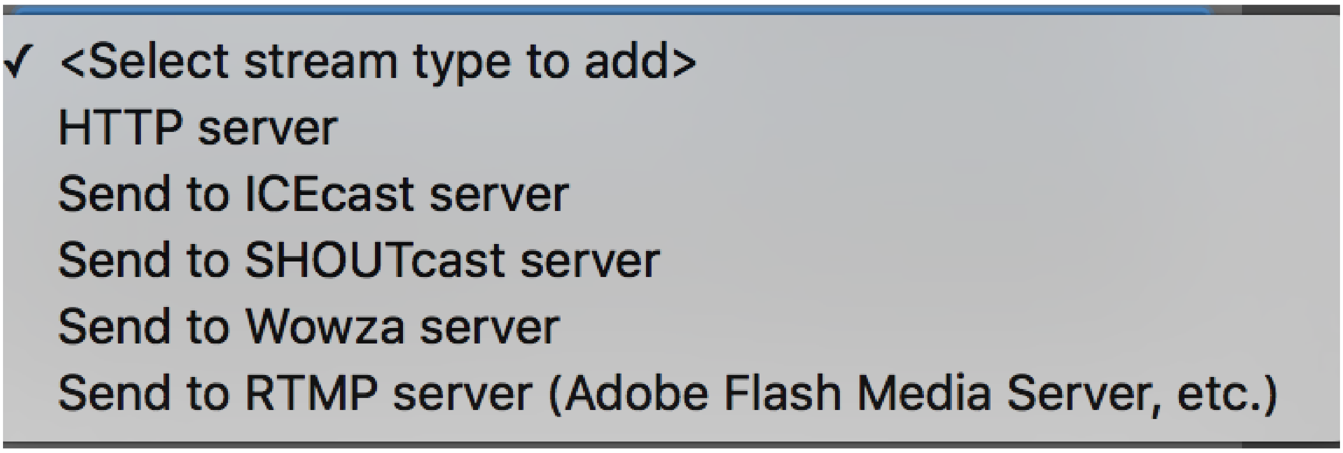
ADTS - Audio Data Transport Stream, a format, used by all other stream types.

RAW+ESBRSIG – Send unformatted audio along to your stream that includes explicit backwards compatible signaling method. This option can be used with RTMP streams.

Adding Streams

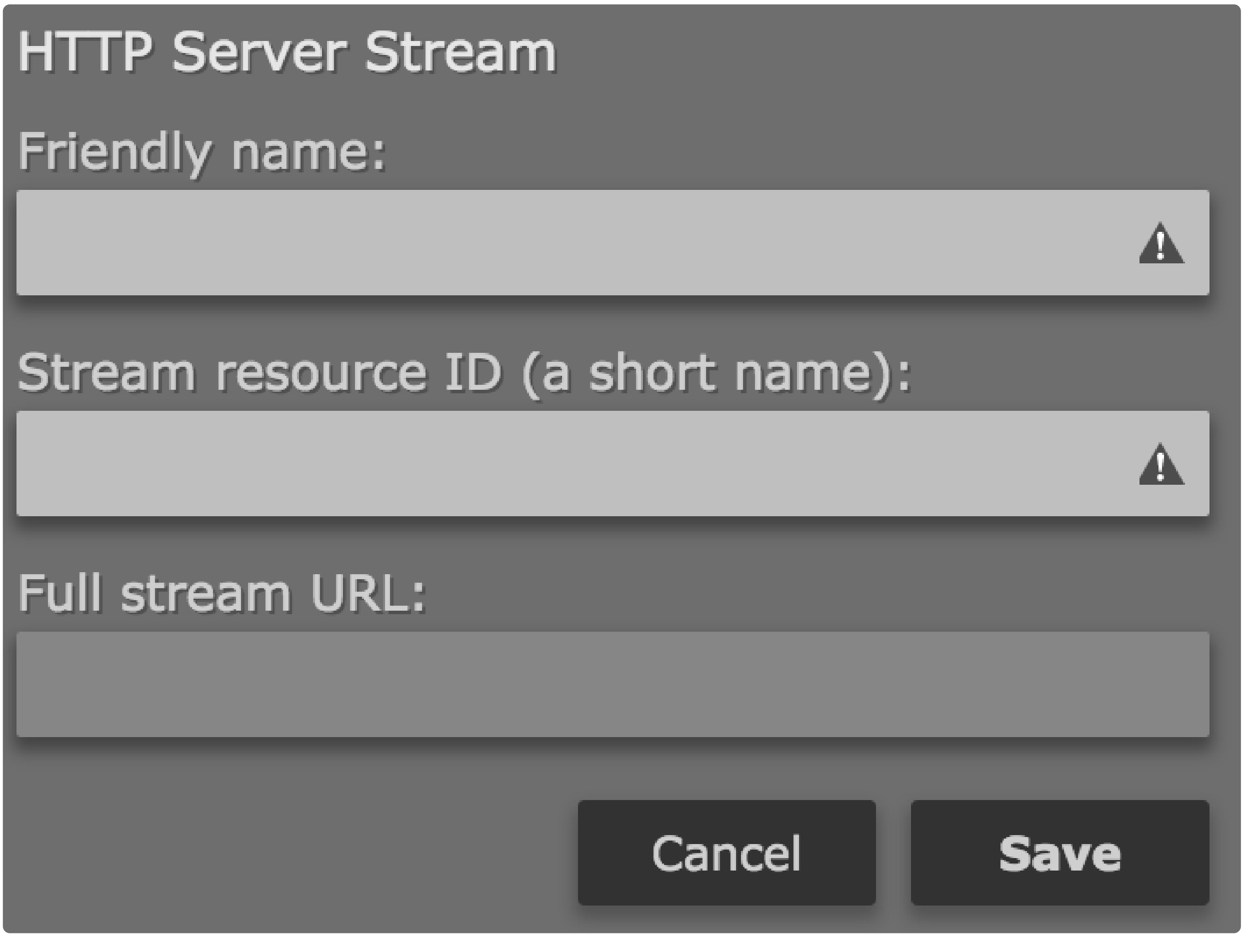


Multiple server destinations can be specified for each encoder instance. To add a destination, click on “Select stream type to add”. Supported server destinations include HTTP, Icecast, SHOUTcast, Wowza, or other RTMP (Adobe Flash Media Server, etc.)



From the dropdown menu, select the server type that you wish to send an output from this encoder instance to.

HTTP Server



The HTTP option will send the output to the HTTP stream server built into R/2. Using this server, you can monitor the encoded stream directly from R/2. This is often helpful in troubleshooting server connections, or just as a quick way to test that everything is working as expected. The HTTP server is also useful when setting up audio processing, as the delay is much shorter than that experienced with an external server.

Friendly Name

Specify an easy to remember “Friendly” name to identify this stream output in the list of streams that have been defined (i.e. Kxxx – Internal Test Stream).

Stream resource ID

This resource ID will become part of the URL that you use to reach this stream. For example, if the stream server is active on port 8888, and you use a resource ID of “TestStream” then you would listen to this stream by entering http://ip_address_of_r2:8888/TestStream in your browser or media player.

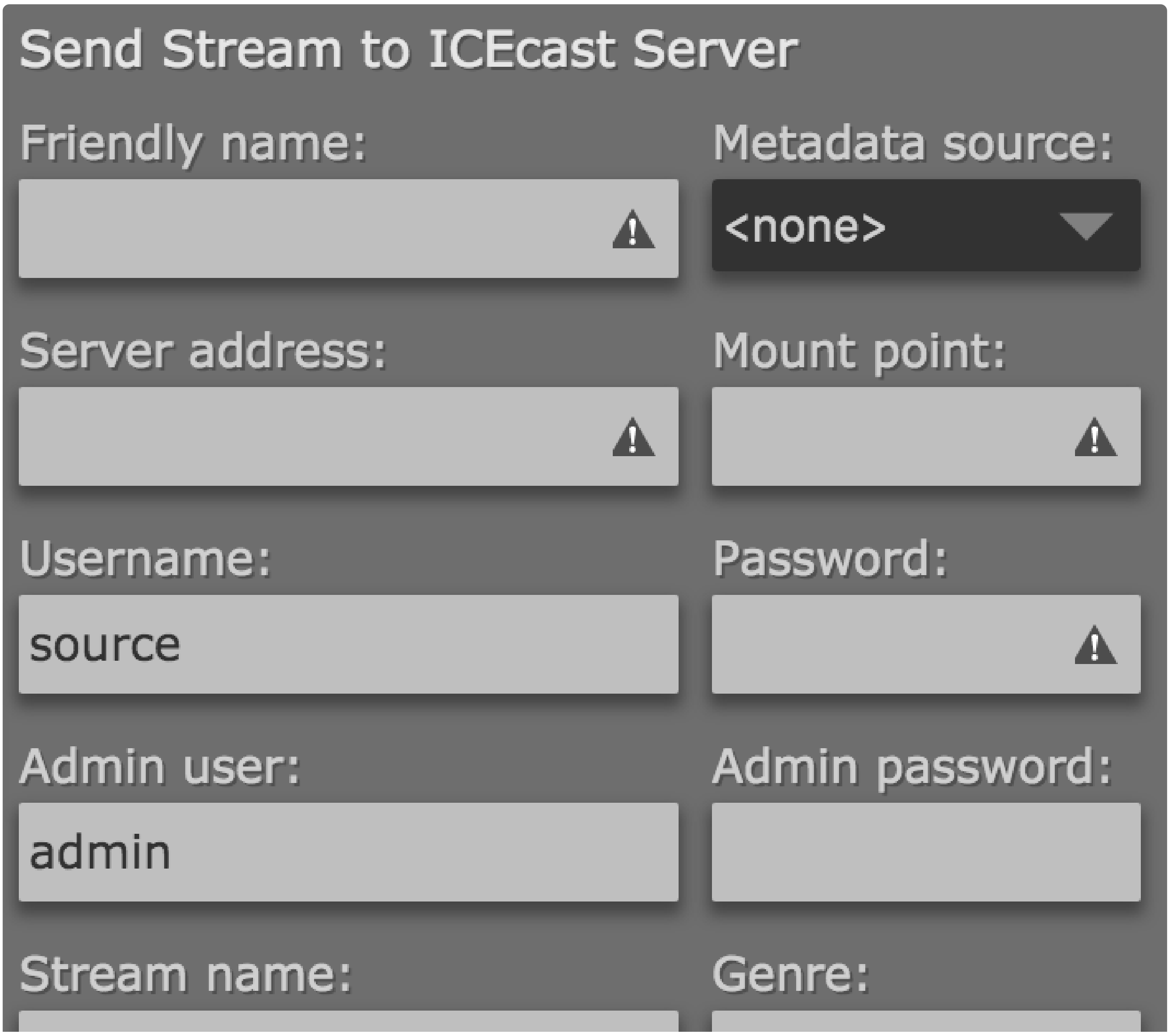
Full stream URL

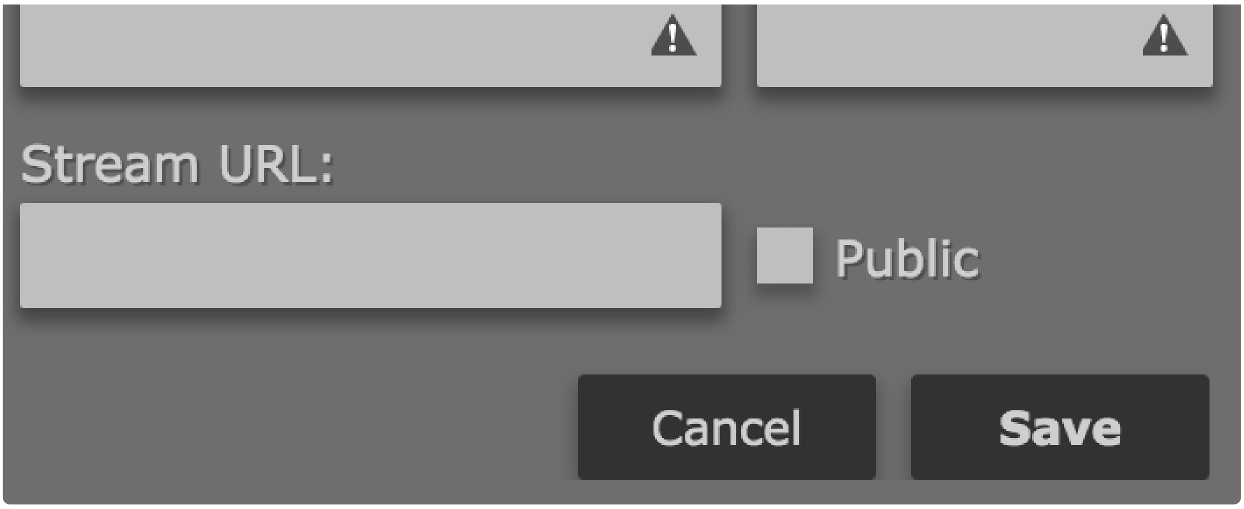
Displays the full URL of this HTTP stream (including the port and Stream resource ID).

Note: As of this writing, R/2 currently displays “localhost” rather than the actual IP address of R/2. Replace “localhost” in the URL with the IP address or hostname of R/2.

When finished, click “Save” to save and return to the “AAC Encoder” screen or click “Cancel” to cancel any changes.

Send to ICEcast Server





The ICEcast Server option allows sending a stream to an ICEcast (or compatible) server. ICEcast is an open source streaming server available as a free download for various platforms. A separate option is provided for SHOUTcast compatible servers.

Friendly name

Specify an easy to remember “Friendly” name to identify this stream output in the list of streams that have been defined (i.e. Kxxx – Icecast Stream).

Metadata source

Select a metadata input source to be used with this encoding instance (if desired). See Chapter 10 for more details.

Server address

Enter the IP address and port of the ICEcast server.

Mount point

Each ICEcast server may contain multiple broadcasts (or mount points) each containing a separate stream of content. Thus, 'mount point' is just a unique name that identifies a particular stream. The mount point needs to be configured properly in order for the stream to work. Mount points should not contain any spaces or odd characters.

Username

Enter a user name here for access to the audio stream. This should be the same username configured on your ICEcast server for sources. The default username is “source”.

Password

Enter the password configured on your ICEcast server for encoding sources. (You DID change it, didn’t you? Please tell me it isn’t still set to “hackme”)

Admin user

Enter the “admin” username for your ICEcast server. The default is “admin”.

This user will be used for the metadata functions of ICEcast.

Admin password

Enter the administrator password configured on your ICEcast server here. (Again, I REALLY hope it isn't still set to "hackme")

Stream name

Enter a name for your stream. This will be used for stream directory listings (if enabled) and displayed in the stream metadata.

Genre

Enter the format of your stream or station here, for example rock, country or urban. ICEcast will display this information in the metadata field.

Stream URL

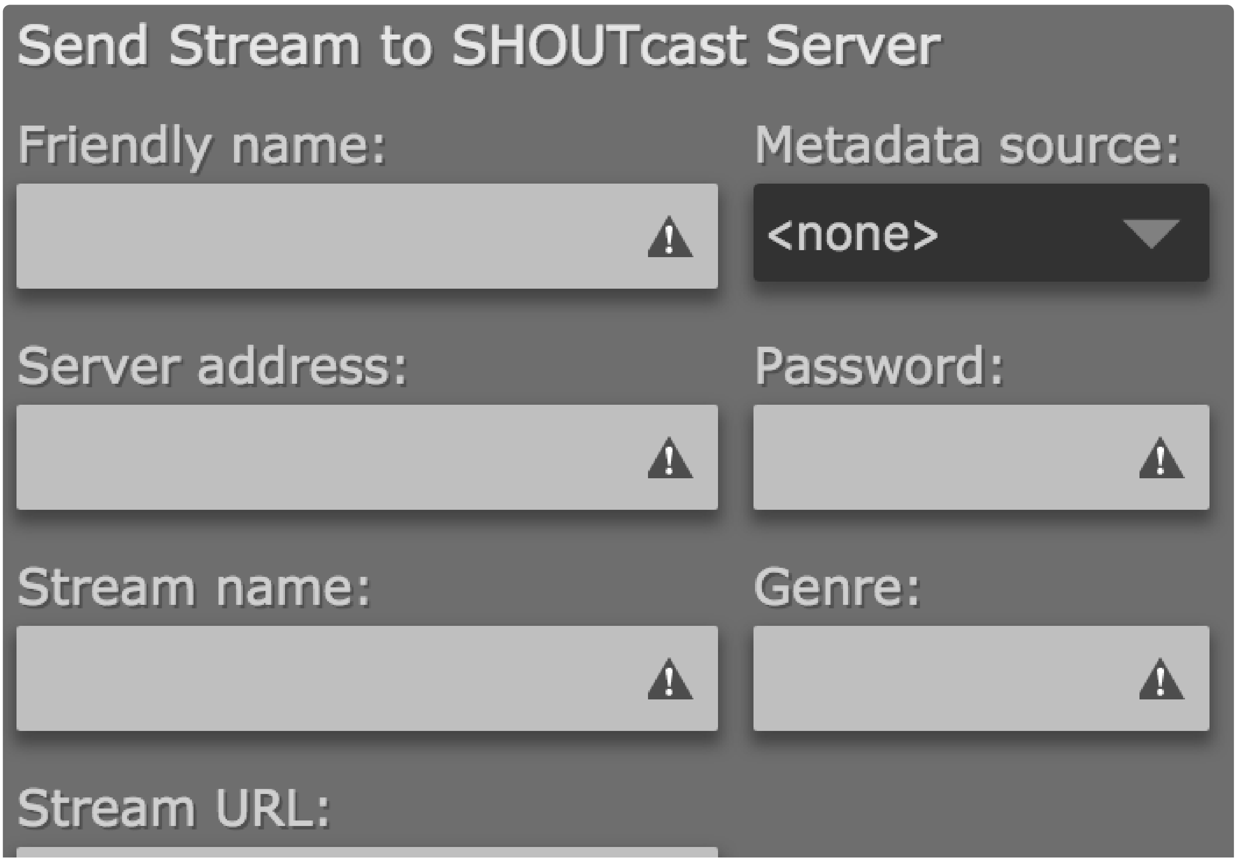
Enter a URL to be displayed with the stream metadata. This can be a station's website, or other URL associated with the stream.

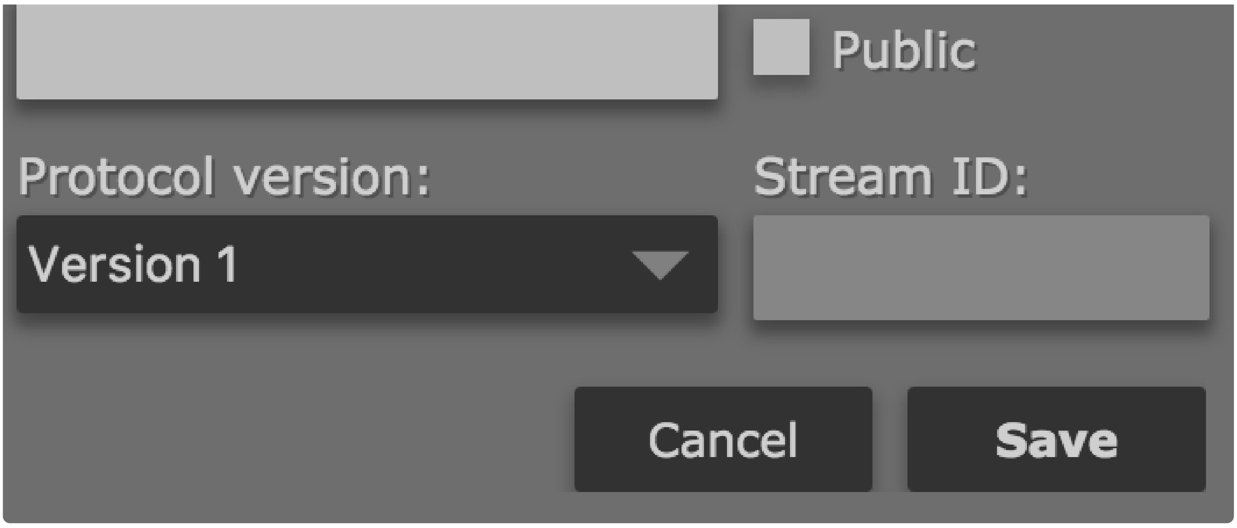
Check the Public checkbox to list this stream in public directories, or leave it un-checked if you do not want the stream listed.

Note: Even with the "Public" checkbox un-checked, it may still be possible for stream "aggregators" to find and list your stream as they crawl the Internet searching for publicly available audio streams. There are very few practical ways to avoid this, unfortunately.

When finished, click "Save" to save and return to the "AAC Encoder" screen or click "Cancel" to cancel any changes.

Send to SHOUTcast Server





SHOUTcast DNAS (Distributed Network Audio Software) is a media server developed by Nullsoft, now distributed by Radionomy. SHOUTcast is available in multiple levels from from a basic (free) version, through paid options that allow monetization. Currently it is available for Windows, Linux and Mac OSX. Cloud-hosted options are also available.

R/2 replaces the DSP encoding plugin for Winamp normally used for broadcasting to a SHOUTcast server.

Friendly name

Specify an easy to remember "Friendly" name to identify this stream output in the list of streams that have been defined (i.e. Kxxx – SHOUTcast Stream).

Metadata source

Select a metadata input source to be used with this encoding instance (if desired). See Chapter 10 for more details.

Server address

Enter the IP address and port of the SHOUTcast server.

Password

Enter the password configured on your SHOUTcast server for encoding sources.

Stream name

Enter a name for your stream. This will be used for stream directory listings (if enabled) and displayed in the stream metadata.

Genre

Enter the format of your stream or station here, for example rock, country or urban. SHOUTcast will display this information in the metadata field.

Stream URL

Enter a URL to be displayed with the stream metadata. This can be a station's website, or other URL associated with the stream.

Check the Public checkbox to list this stream in public directories, or leave it un-checked if you do not want the stream listed.

Note: Even with the “Public” checkbox un-checked, it may still be possible for stream “aggregators” to find and list your stream as they crawl the Internet searching for publicly available audio streams. There are very few practical ways to avoid this, unfortunately.

Protocol Version

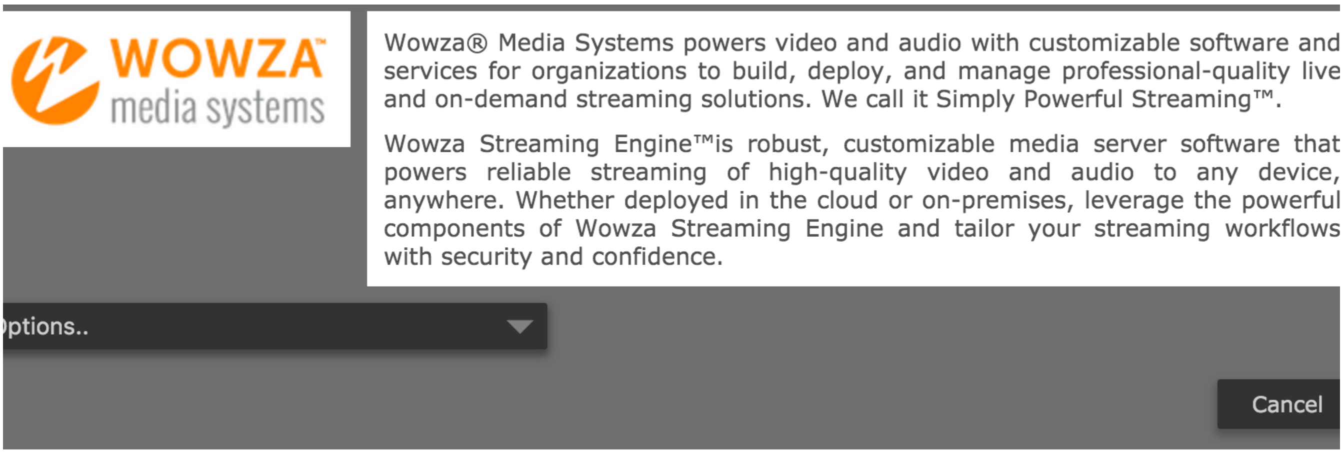
Choose between SHOUTcast Version 1 or Version 2, depending on the requirements of your SHOUTcast server platform.

Stream ID

Specify the numerical identifier of the stream for control or referencing the stream configuration. This can only be a numeric value from 1 to 2147483647.

When finished, click “Save” to save and return to the “AAC Encoder” screen or click “Cancel” to cancel any changes.

Send to Wowza server



R/2 offers direct integration with Wowza servers for easy configuration of encoding parameters. Configuration files can be loaded into R/2 as an XML file downloaded from the Wowza server, XML configuration copy/paste, or manually configured.

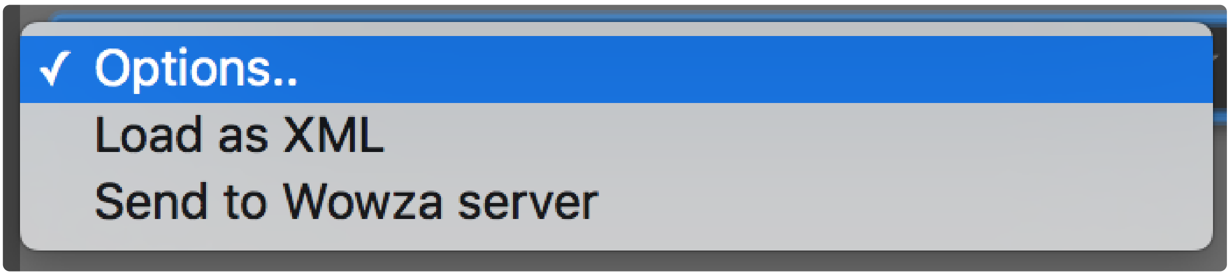
Note: In order to send AAC encoded audio to a Wowza server, you must use “Raw” for the stream file output format, as described earlier in this chapter. Other stream file formats cannot be used simultaneously in the same instance (i.e. ADTS for the internal HTTP server or other servers). Create a new instance if you wish to use another stream file format simultaneously.

Although it can be disabled, Wowza is set by default to use authentication for RTMP encoding sources such as R/2, and it is highly recommended. You will need to define at least one valid “Publisher” account in the “Server” setup for Wowza.

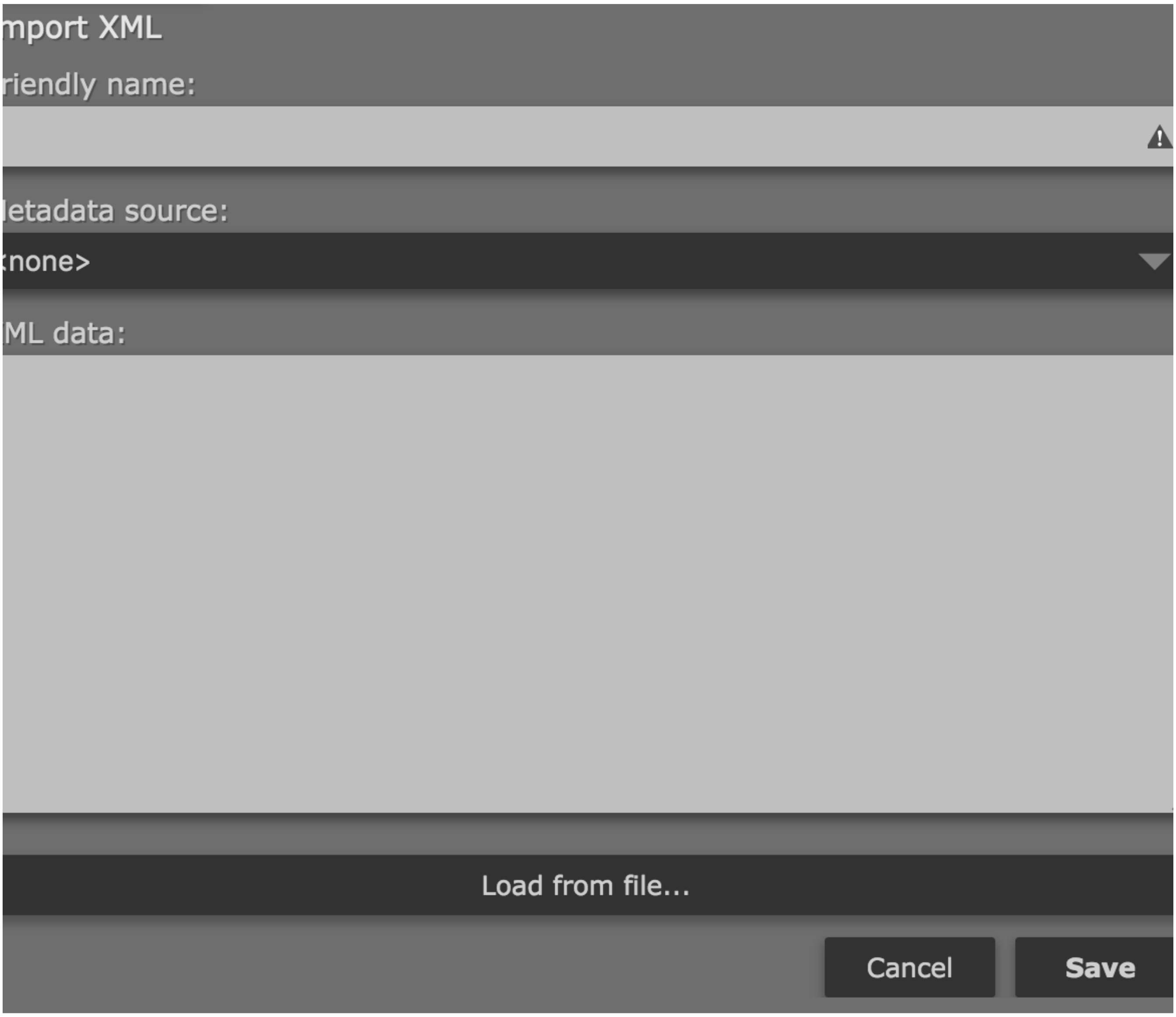
In the Wowza Streaming Engine Manager console, Navigate to “Server”, then to “Source Authentication”. Click on “Add Source” and specify a username and password. This is necessary for both XML-based and manual configuration of a Wowza stream in R/2.

Configuring R/2 for a Wowza stream

Click on the “Options” dropdown to choose a method for configuring a Wowza connection.



Load as XML



Allows loading an XML configuration for a Wowza connection either from a file, or by copy/paste.

This file can be downloaded from the Wowza server by clicking on the desired streaming application in Wowza (live, for example) then clicking on “Incoming Publishers”. Select “The Telos Alliance” from the list of available encoder manufacturers. The appropriate configuration parameters will appear, along with a button to download an XML file containing the configuration for R/2. You will also need to define an account for the incoming source.

Friendly name

Specify an easy to remember “Friendly” name to identify this stream output in the list of streams that have been defined (i.e. Kxxx – Wowza Stream).

Metadata source

Select a metadata input source to be used with this encoding instance (if desired). See Chapter 10 for more details.

XML data

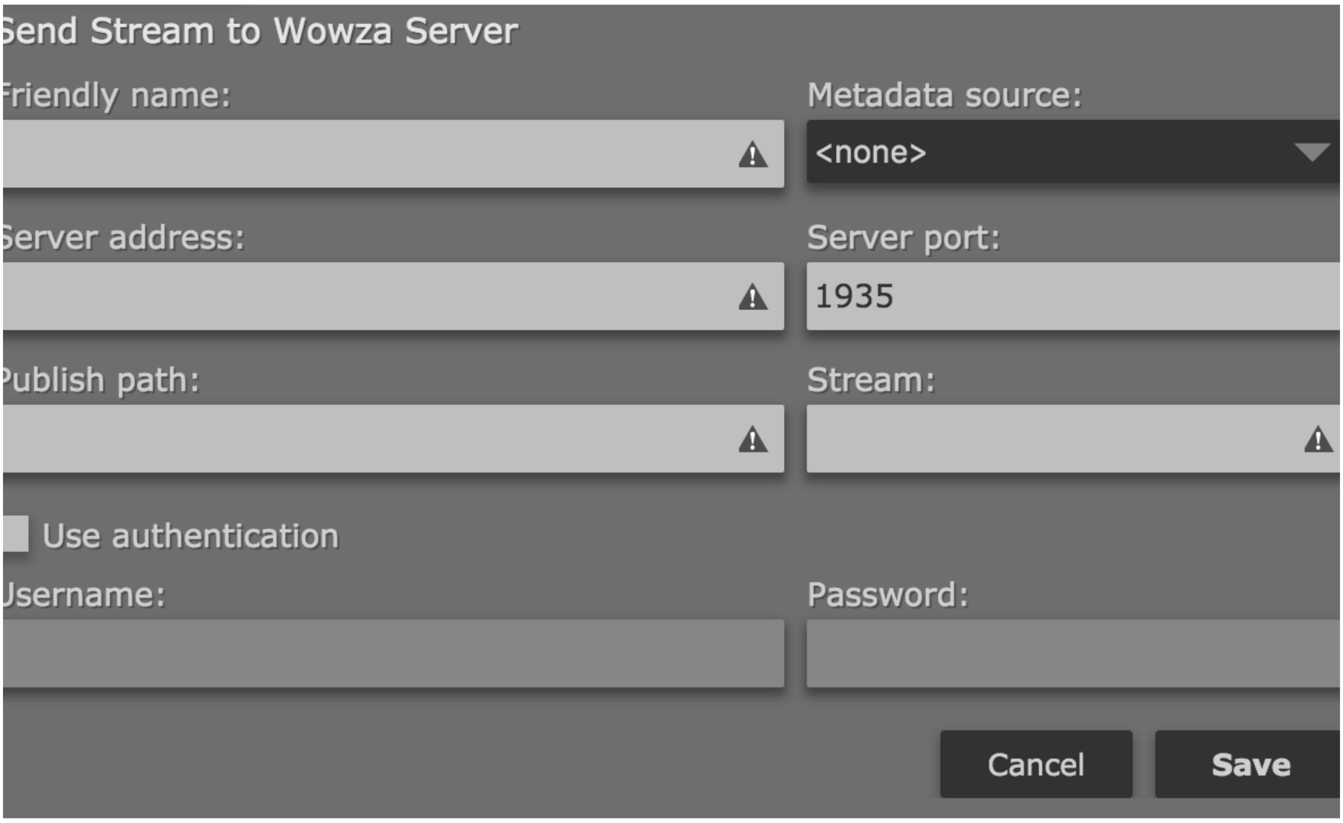
Copy and paste the XML configuration data for this connection from the Wowza server here.

Load from file

Click here to load Wowza XML configuration data from a file.

When finished, click “Save” to save and return to the “AAC Encoder” screen or click “Cancel” to cancel any changes.

Send to Wowza Server



Allows manual configuration of a stream pushed to a Wowza server.

Friendly name

Specify an easy to remember “Friendly” name to identify this stream output in the list of streams that have been defined (i.e. Kxxx – Wowza Stream).

Metadata source

Select a metadata input source to be used with this encoding instance (if desired). See Chapter 10 for more details.

Server address

Enter the IP address of the Wowza server.

Server port

Enter the port of the Wowza server. The default is 1935.

Publish path

Enter the path to the desired streaming application on your Wowza server (live, for example).

Stream

Enter the name of the stream you wish to create on your Wowza server. This stream will be created under the specified path (application).

Use Authentication

Check this box to use authentication when sending a stream to the Wowza server. Authentication for RTMP publisher sources such as R/2 is enabled by default in Wowza and will likely need to be enabled here as well.

Username

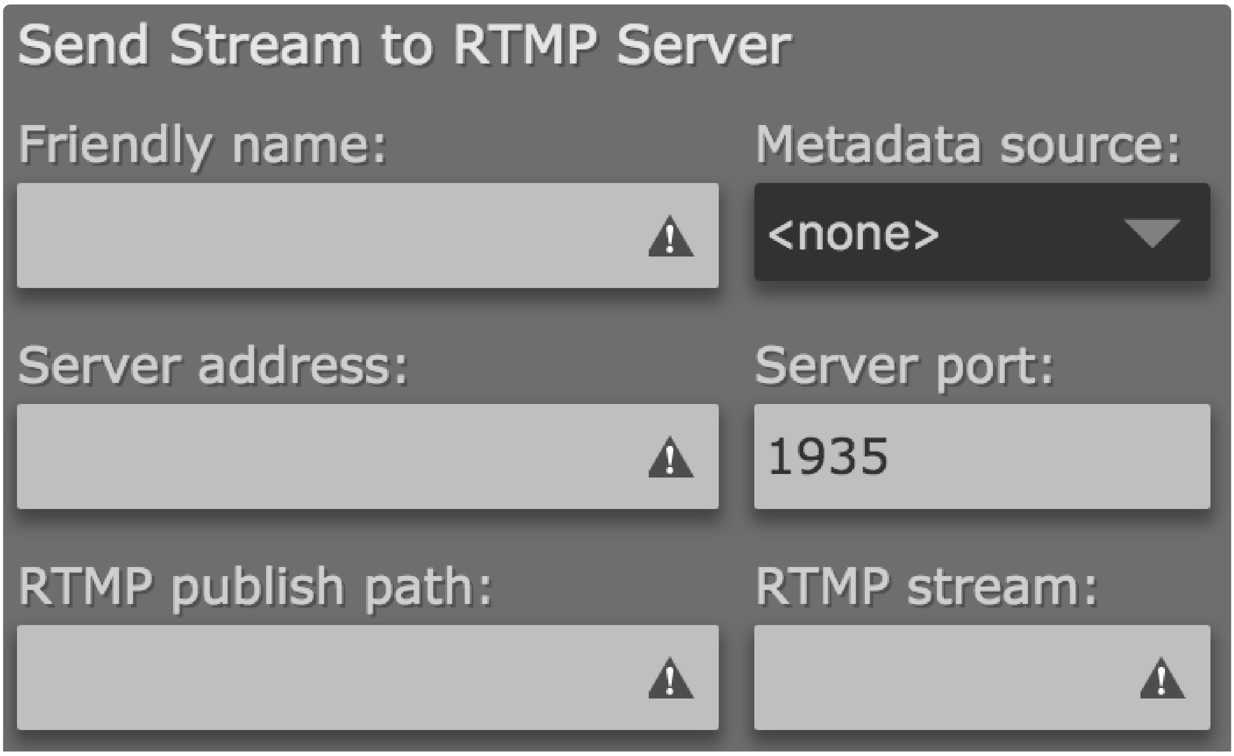
Enter the username specified as a publisher in the Wowza server.

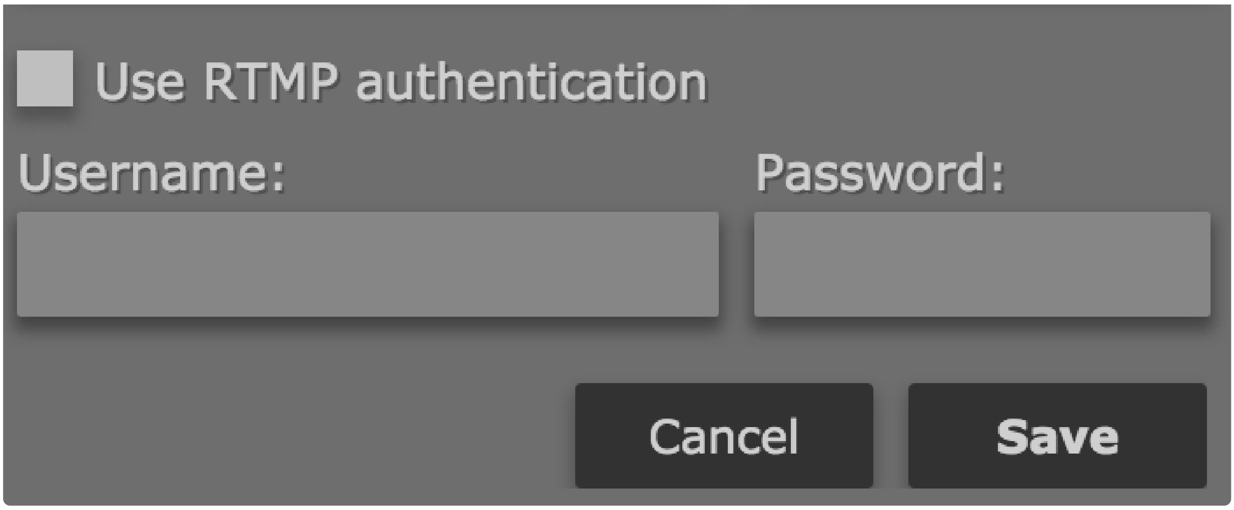
Password

Enter the password associated with the publisher username in the Wowza server.

When finished, click “Save” to save and return to the “AAC Encoder” screen or click “Cancel” to cancel any changes.

Send to RTMP Server





RTMP (Real Time Messaging Protocol) is a TCP-based protocol initially developed by Macromedia to deliver streams to the Flash player. The specification is now available (albeit in incomplete form) for public use. A number of servers support the RTMP protocol, including Adobe Flash Media Server, Wowza and others.

Note: In order to send AAC encoded audio to an RTMP server, you must use “Raw” for the stream file output format, as described earlier in this chapter. Other stream file formats cannot be used simultaneously in the same instance (i.e. ADTS for the internal HTTP server or other servers). Create a new instance if you wish to use another stream file format simultaneously.

Friendly name

Specify an easy to remember “Friendly” name to identify this stream output in the list of streams that have been defined (i.e. Kxxx – RTMP Stream).

Metadata source

Select a metadata input source to be used with this encoding instance (if desired). See Chapter 10 for more details.

Server address

Enter the IP address of the RTMP server.

Server port

Enter the port of the RTMP server. The default is 1935.

RTMP Publish path

Enter the publishing path for the RTMP server—generally provided by your CDN, and may also be referred to as an “entry point”.

RTMP Stream

Enter the name of the RTMP stream name. The stream name may be fixed or chosen, it depends on your CDN. Some providers, such as Akamai, require the stream name to be very specific, and they provide the format that must be used.

Use RTMP Authentication

Check this box to use authentication when sending a stream to the RTMP server.

Username

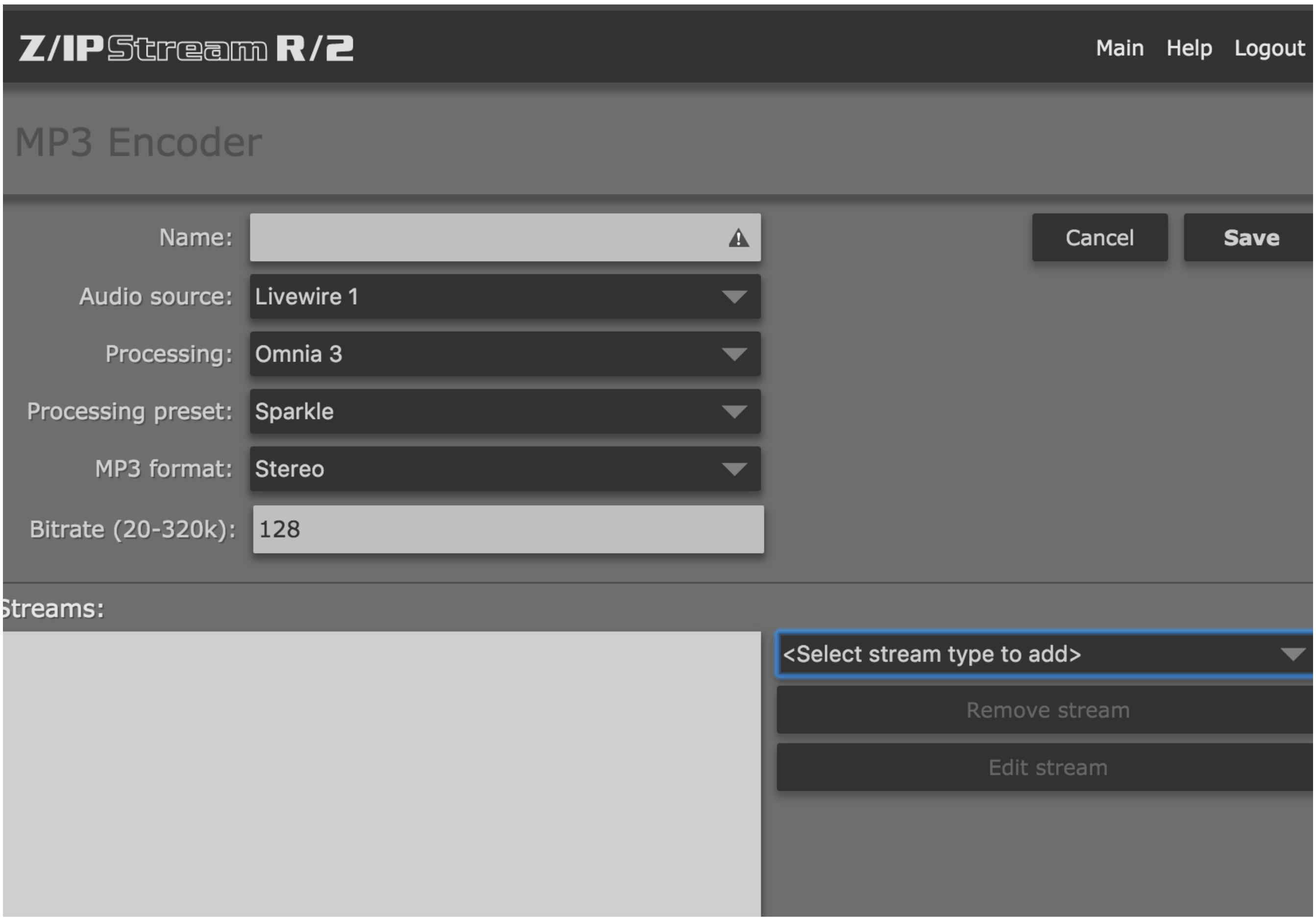
Enter the username for encoding to your RTMP server.

Password

Enter the password for encoding to your RTMP server.

When finished, click “Save” to save and return to the “AAC Encoder” screen or click “Cancel” to cancel any changes.

MP3 Encoder Instance



An MP3 Encoder instance accepts data from an audio source, processes it, encodes it to MPEG Layer-3, and makes it available to media servers. The encoded stream can also be made available through the HTTP stream server built into the application. The internal HTTP server is intended for monitoring the stream by a very small number of listeners. It is not designed to feed a large number of clients.

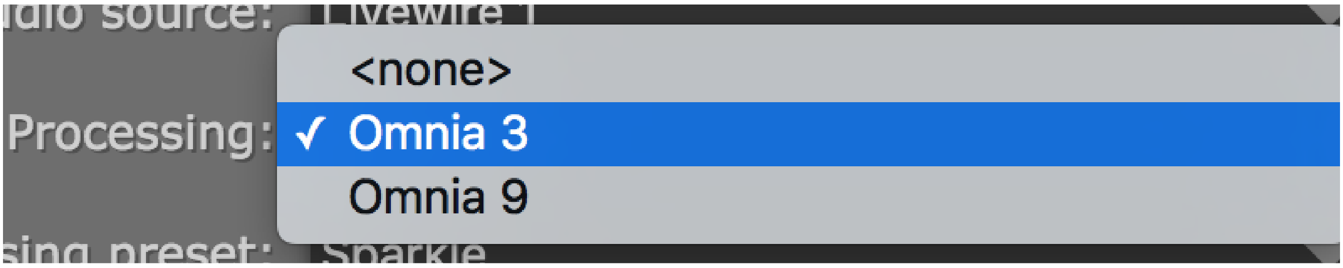
Name

Enter a “friendly name” for this encoder instance to easily identify it for status and control from the main “control panel” screen (i.e. Kxxx-MP3 Stream 1).

Audio Source

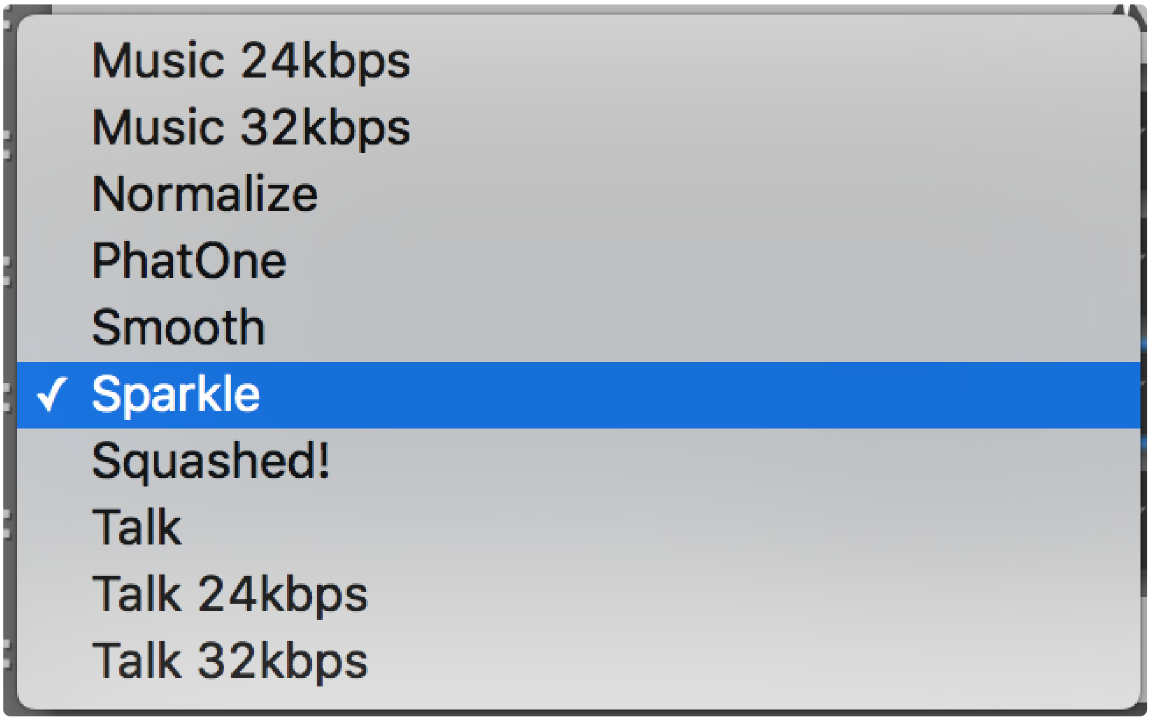
Select one of the previously defined audio sources as the input to this processing instance (see Chapter 7). Note that these sources can be used in multiple processing or encoding instances simultaneously.

Processing

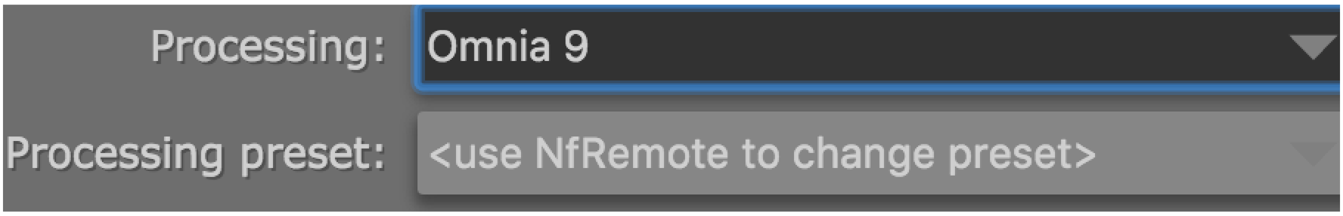


Selects between Omnia 3 and optional Omnia.9 processing (or no processing at all, should you simply want to route audio through this instance without processing). Up to 8 channels of Omnia.9 processing can be licensed per R/2. When you have reached the maximum number of Omnia.9 licenses available (or if Omnia.9 processing is not licensed) this option will not be displayed. See Chapter 16 for more information on Omnia.9 processing.

Processing preset

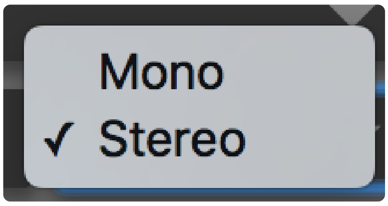


Selects the processing preset to be used for this instance. When Omnia 3 processing is selected, the list of available presets will appear. These presets are not format specific, though there are bitrate specific presets for “Music” and “Talk”. Experiment with various presets to find one that works best for your programming format and application. Presets can be copied and modified to create a sound customized exactly to your needs. See Chapter 9 to learn how to “fine tune” the Omnia 3 processing, and Chapter 13 for more information on Omnia 3 preset management.



If optional Omnia.9 processing is enabled, the preset must be selected via NfRemote software due to the powerful processing tools and test instrumentation available through Omnia.9. See Chapter 16 for more details.

MP3 Format

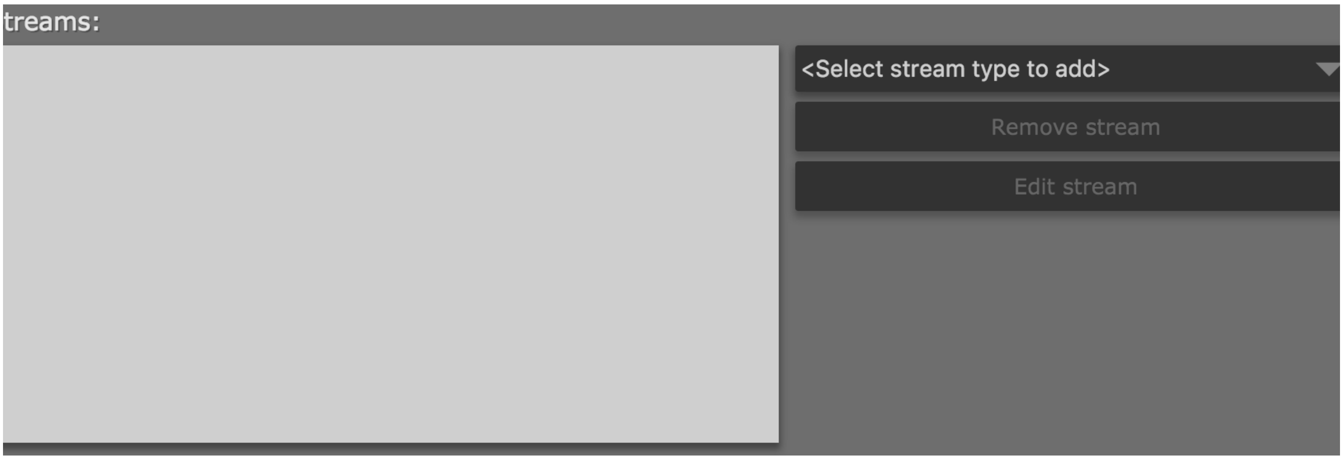


Select whether this stream will be encoded in Mono or Stereo. Mono should provide a mono sum of the incoming audio.

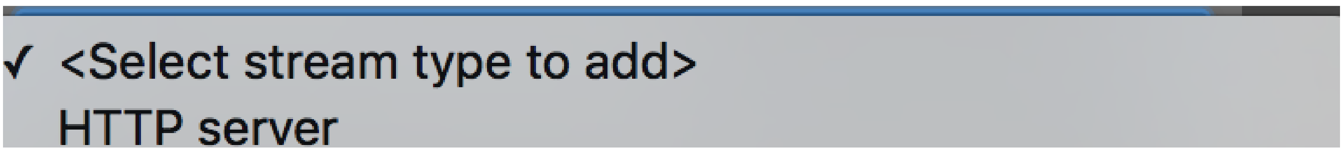
Bitrate

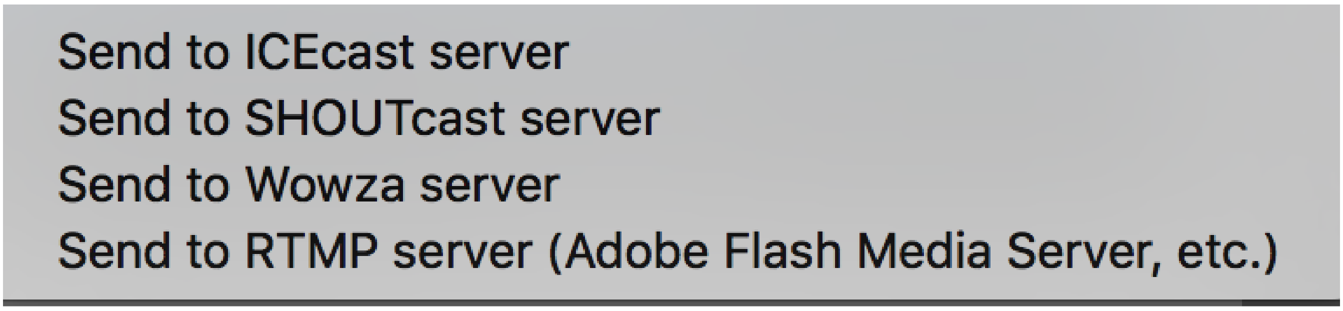
Enter the bitrate for this stream. Valid MP3 bitrates are 20 kbps to 320 kbps.

Adding Streams



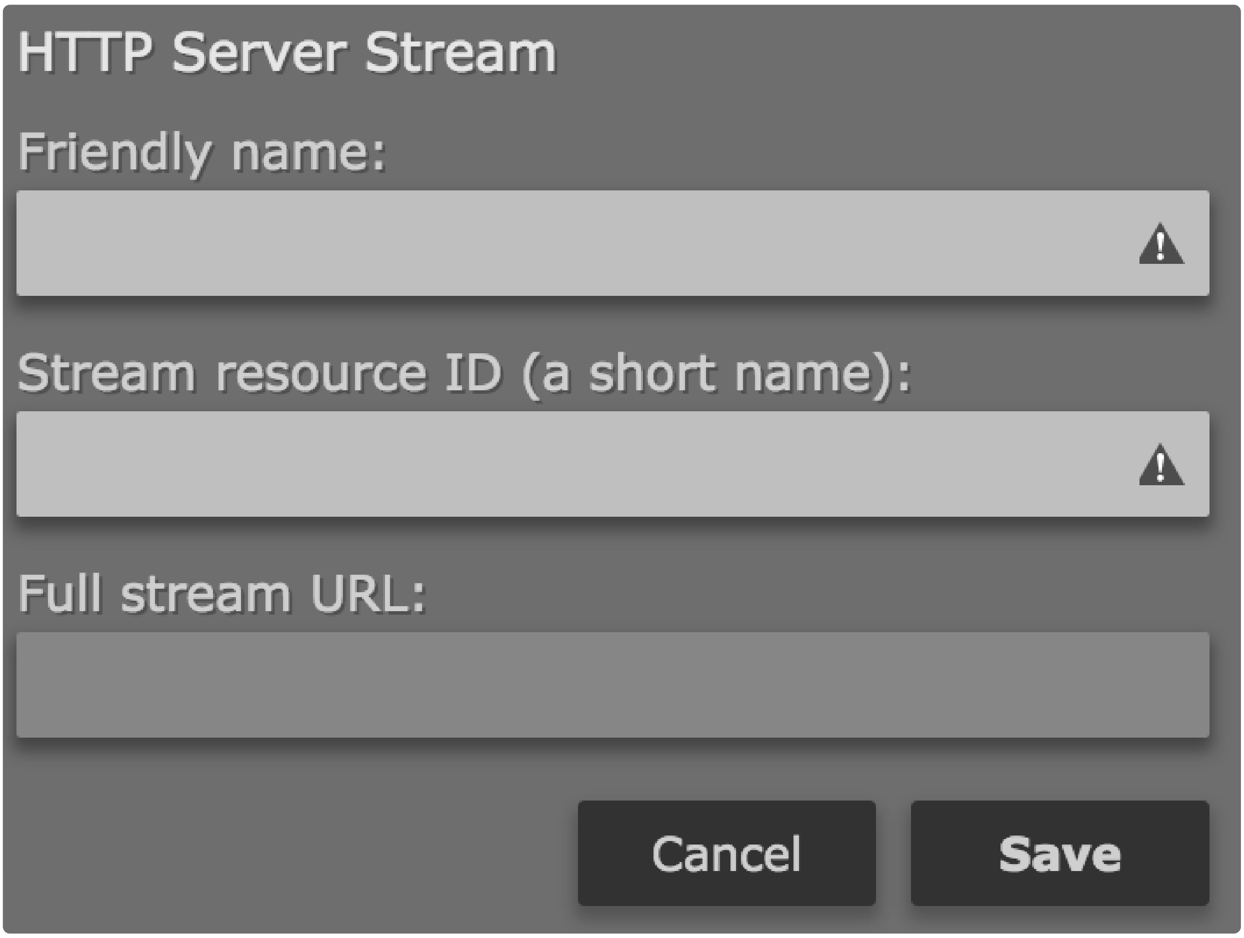
Multiple server destinations can be specified for each encoder instance. To add a destination, click on “Select stream type to add”. Supported server destinations include HTTP, Icecast, SHOUTcast, Wowza, or other RTMP (Adobe Flash Media Server, etc.)





From the dropdown menu, select the server type that you wish to send an output from this encoder instance to.

HTTP Server



The HTTP option will send the output to the HTTP stream server built into R/2. Using this server, you can monitor the encoded stream directly from R/2. This is often helpful in troubleshooting server connections, or just as a quick way to test that everything is working as expected. The HTTP server is also useful when setting up audio processing, as the delay is much shorter than that experienced with an external server.

Friendly Name

Specify an easy to remember "Friendly" name to identify this stream output in the list of streams that have been defined (i.e. Kxxx – Internal Test Stream).

Stream resource ID

This resource ID will become part of the URL that you use to reach this stream. For example, if the stream server is active on port 8888, and you use a resource ID of "TestStream" then you would listen to this stream by entering http://ip_address_of_r2:8888/TestStream in your browser or media player.

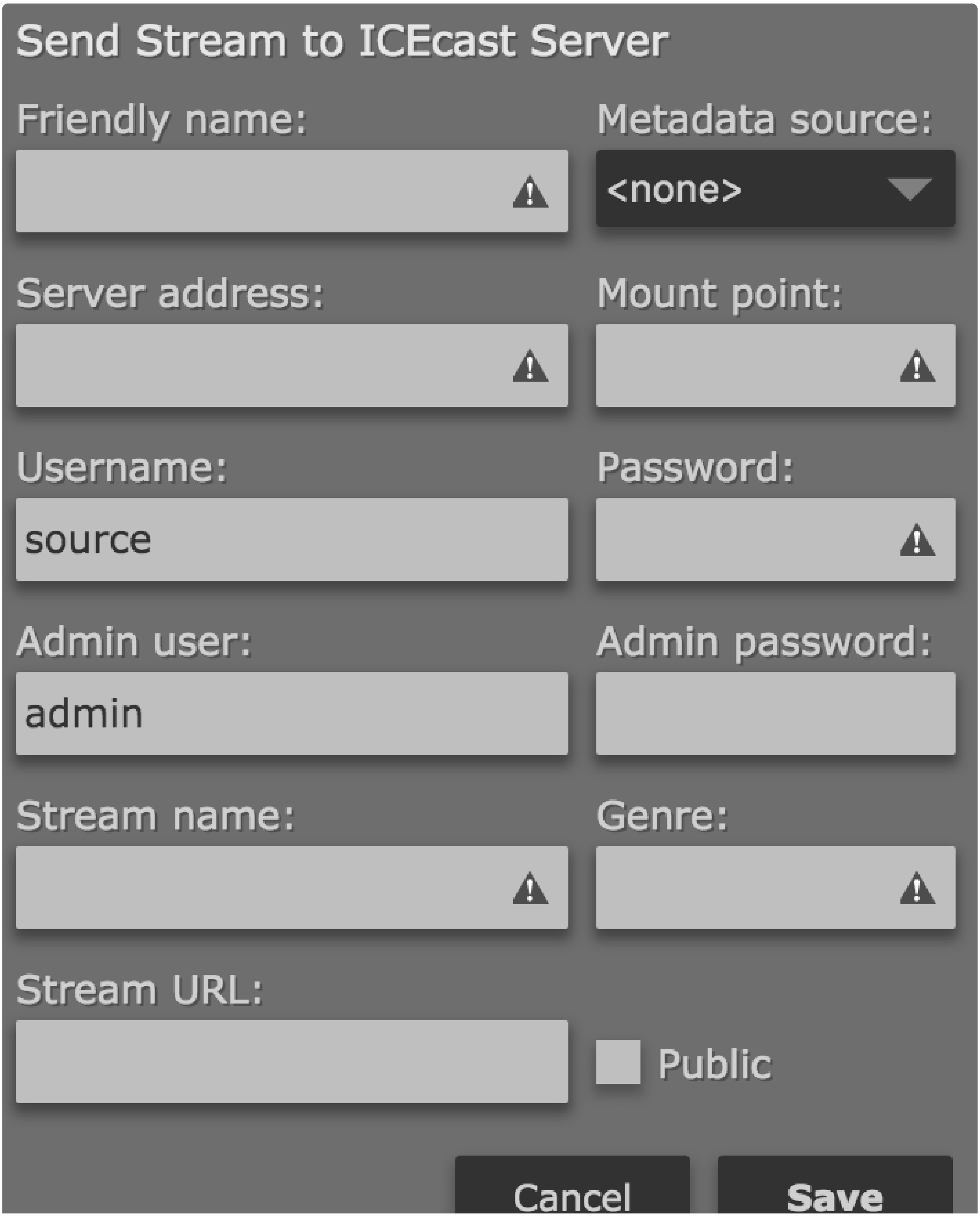
Full stream URL

Displays the full URL of this HTTP stream (including the port and Stream resource ID).

Note: As of this writing, R/2 currently displays “localhost” rather than the actual IP address of R/2. Replace “localhost” in the URL with the IP address or hostname of R/2.

When finished, click “Save” to save and return to the “MP3 Encoder” screen or click “Cancel” to cancel any changes.

Send to ICEcast Server



The ICECast Server option allows sending a stream to an ICEcast (or compatible) server. ICECast is an open source streaming server available as a free download for various platforms. A separate option is provided for SHOUTcast compatible servers.

Friendly name

Specify an easy to remember “Friendly” name to identify this stream output in the list of streams that have been defined (i.e. Kxxx – Icecast Stream).

Metadata source

Select a metadata input source to be used with this encoding instance (if desired). See Chapter 10 for more details.

Server address

Enter the IP address and port of the ICEcast server.

Mount point

Each ICEcast server may contain multiple broadcasts (or mount points) each containing a separate stream of content. Thus, 'mount point' is just a unique name that identifies a particular stream. The mount point needs to be configured properly in order for the stream to work. Mount points should not contain any spaces or odd characters.

Username

Enter a user name here for access to the audio stream. This should be the same username configured on your ICECast server for sources. The default username is “source”.

Password

Enter the password configured on your ICECast server for encoding sources. (You DID change it, didn’t you? Please tell me it isn’t still set to “hackme”)

Admin user

Enter the “admin” username for your ICECast server. The default is “admin”.

This user will be used for the metadata functions of ICEcast.

Admin password

Enter the administrator password configured on your ICECast server here. (Again, I REALLY hope it isn’t still set to “hackme”)

Stream name

Enter a name for your stream. This will be used for stream directory listings (if enabled) and displayed in the stream metadata.

Genre

Enter the format of your stream or station here, for example rock, country or urban. ICECast will display this information in the metadata field.

Stream URL

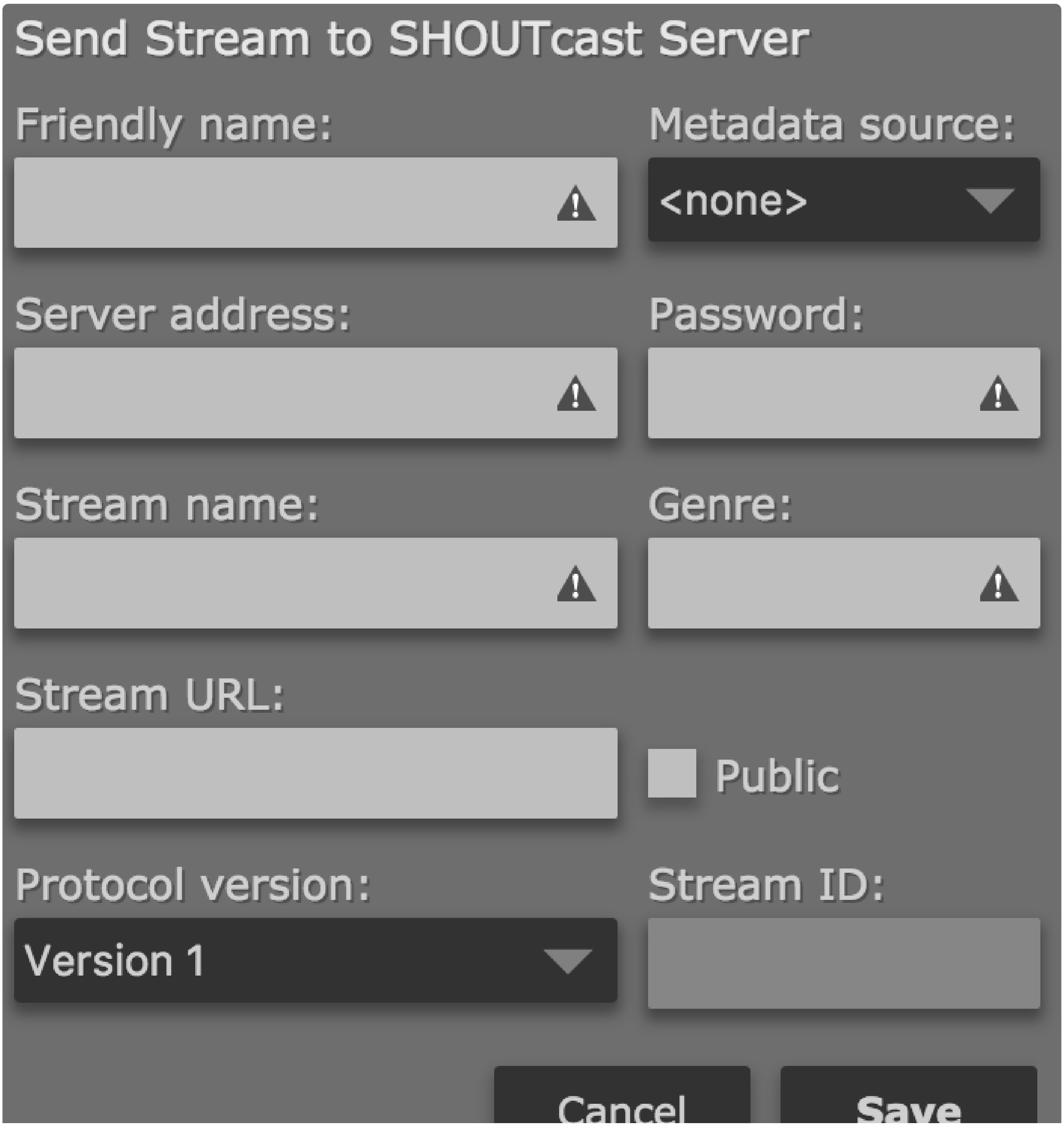
Enter a URL to be displayed with the stream metadata. This can be a station's website, or other URL associated with the stream.

Check the Public checkbox to list this stream in public directories, or leave it un-checked if you do not want the stream listed.

Note: Even with the "Public" checkbox un-checked, it may still be possible for stream "aggregators" to find and list your stream as they crawl the Internet searching for publicly available audio streams. There are very few practical ways to avoid this, unfortunately.

When finished, click "Save" to save and return to the "MP3 Encoder" screen or click "Cancel" to cancel any changes.

Send to SHOUTcast Server



SHOUTcast DNAS (Distributed Network Audio Software) is a media server developed by Nullsoft, now distributed by Radionomy. SHOUTcast is available in multiple levels from from a basic (free) version through paid options that allow monetization. Currently it is available for Windows, Linux and Mac OSX. Cloud-hosted options are also available.

R/2 replaces the DSP encoding plugin for Winamp normally used for broadcasting to a SHOUTcast server.

Friendly name

Specify an easy to remember “Friendly” name to identify this stream output in the list of streams that have been defined (i.e. Kxxx – SHOUTcast Stream).

Metadata source

Select a metadata input source to be used with this encoding instance (if desired). See Chapter 10 for more details.

Server address

Enter the IP address and port of the SHOUTcast server.

Password

Enter the password configured on your SHOUTcast server for encoding sources.

Stream name

Enter a name for your stream. This will be used for stream directory listings (if enabled) and displayed in the stream metadata.

Genre

Enter the format of your stream or station here, for example rock, country or urban. SHOUTcast will display this information in the metadata field.

Stream URL

Enter a URL to be displayed with the stream metadata. This can be a station’s website, or other URL associated with the stream.

Check the Public checkbox to list this stream in public directories, or leave it un-checked if you do not want the stream listed.

Note: Even with the “Public” checkbox un-checked, it may still be possible for stream “aggregators” to find and list your stream as they crawl the Internet searching for publicly available audio streams. There are very few practical ways to avoid this, unfortunately.

Protocol Version

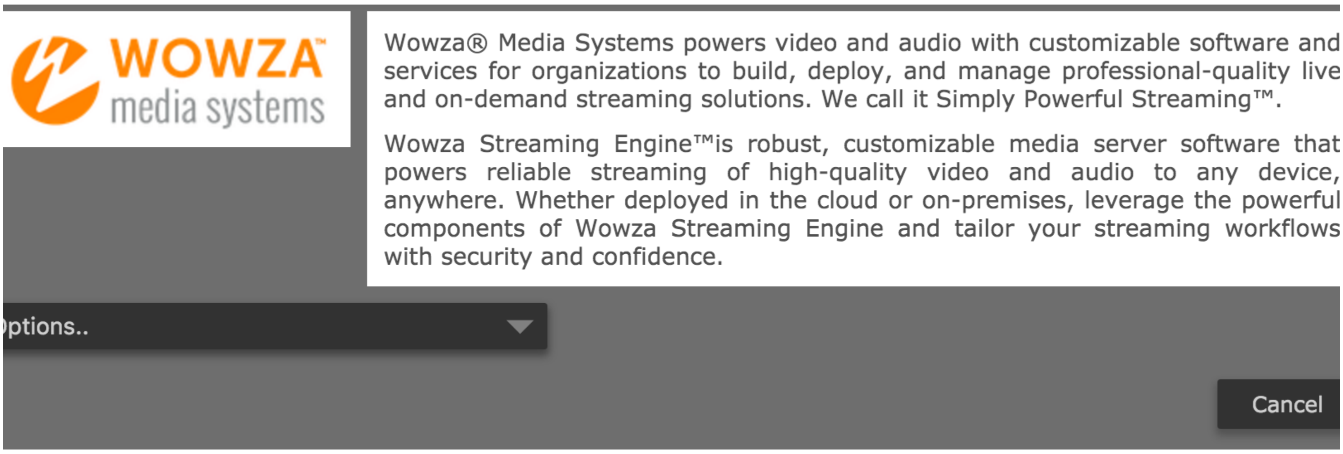
Choose between SHOUTcast Version 1 or Version 2, depending on the requirements of your SHOUTcast server platform.

Stream ID

Specify the numerical identifier of the stream for control or referencing the stream configuration. This can only be a numeric value from 1 to 2147483647.

When finished, click “Save” to save and return to the “MP3 Encoder” screen or click “Cancel” to cancel any changes.

Send to Wowza server



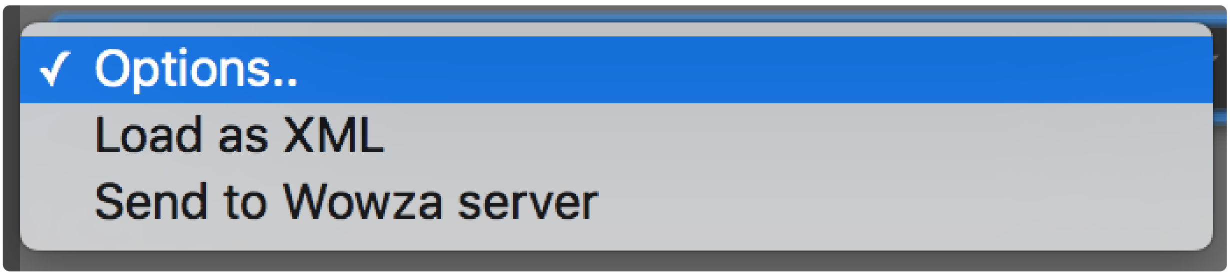
R/2 offers direct integration with Wowza servers for easy configuration of encoding parameters. Configuration files can be loaded into R/2 as an XML file downloaded from the Wowza server, XML configuration copy/paste, or manually configuration.

Although it can be disabled, Wowza is set by default to use authentication for RTMP encoding sources such as R/2, and it is highly recommended. You will need to define at least one valid “Publisher” account in the “Server” setup for Wowza.

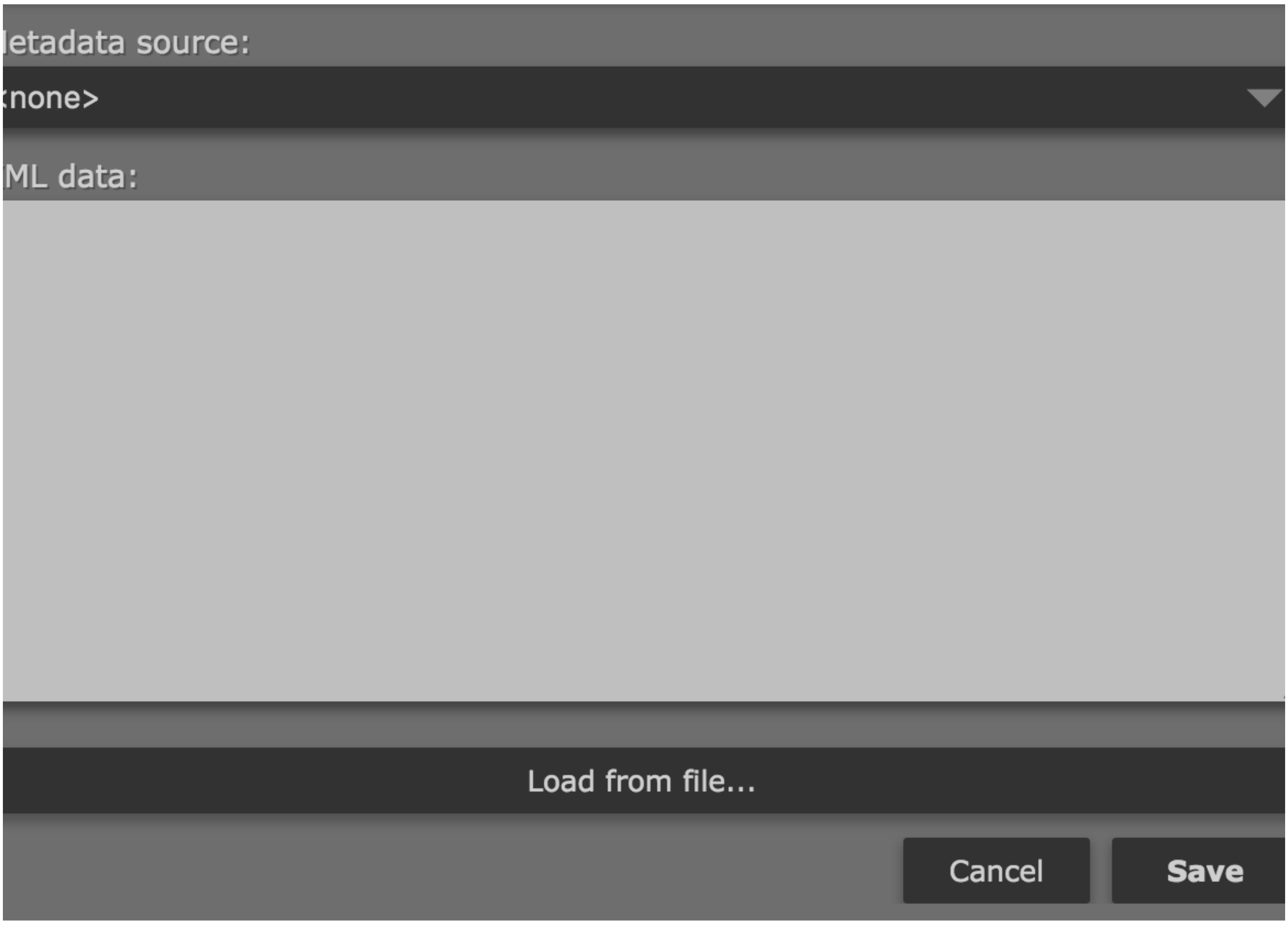
In the Wowza Streaming Engine Manager console, Navigate to “Server”, then to “Source Authentication”. Click on “Add Source” and specify a username and password. This is necessary for both XML-based and manual configuration of a Wowza stream in R/2.

Configuring R/2 for a Wowza stream

Click on the “Options” dropdown to choose a method for configuring a Wowza connection.



Load as XML



Allows loading an XML configuration for a Wowza connection either from a file, or by copy/paste.

This file can be downloaded from the Wowza server by clicking on the desired streaming application in Wowza (live, for example) then clicking on “Incoming Publishers”. Select “The Telos Alliance” from the list of available encoder manufacturers. The appropriate configuration parameters will appear, along with a button to download an XML file containing the configuration for R/2. You will also need to define an account for the incoming source.

Friendly name

Specify an easy to remember “Friendly” name to identify this stream output in the list of streams that have been defined (i.e. Kxxx – Wowza Stream).

Metadata source

Select a metadata input source to be used with this encoding instance (if desired). See Chapter 10 for more details.

XML data

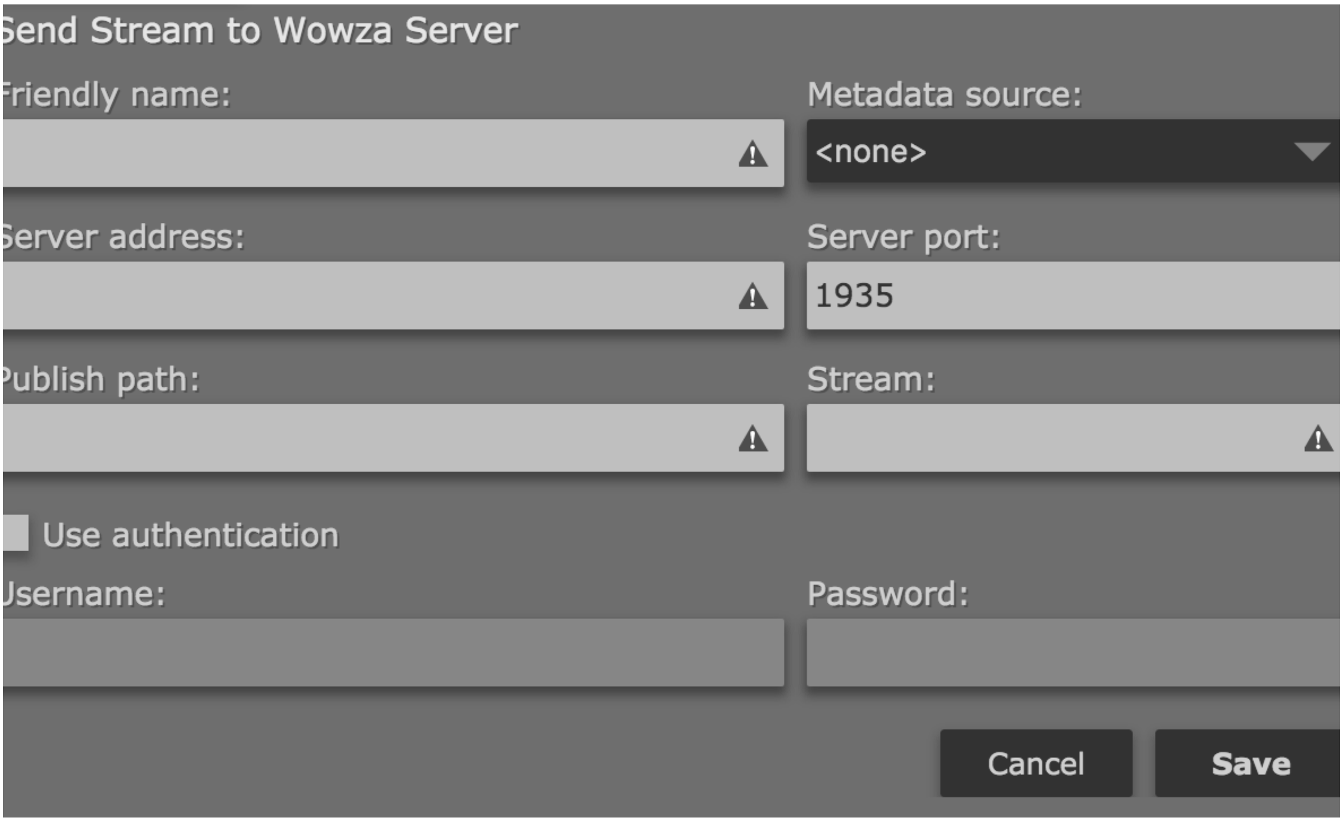
Copy and paste the XML configuration data for this connection from the Wowza server here.

Load from file

Click here to load Wowza XML configuration data from a file.

When finished, click “Save” to save and return to the “MP3 Encoder” screen or click “Cancel” to cancel any changes.

Send to Wowza Server



Allows manual configuration of a stream pushed to a Wowza server.

Friendly name

Specify an easy to remember “Friendly” name to identify this stream output in the list of streams that have been defined (i.e. Kxxx – Wowza Stream).

Metadata source

Select a metadata input source to be used with this encoding instance (if desired). See Chapter 10 for more details.

Server address

Enter the IP address of the Wowza server.

Server port

Enter the port of the Wowza server. The default is 1935.

Publish path

Enter the path to the desired streaming application on your Wowza server (live, for example).

Stream

Enter the name of the stream you wish to create on your Wowza server. This stream will be created under the specified path (application).

Use Authentication

Check this box to use authentication when sending a stream to the Wowza server. Authentication for RTMP publisher sources such as R/2 is enabled by default in Wowza and will likely need to be enabled here as well.

Username

Enter the username specified as a publisher in the Wowza server.

Password

Enter the password associated with the publisher username in the Wowza server.

When finished, click “Save” to save and return to the “MP3 Encoder” screen or click “Cancel” to cancel any changes.

Send to RTMP Server

RTMP (Real Time Messaging Protocol) is a TCP-based protocol initially developed by Macromedia to deliver streams to the Flash player. The specification is now available (albeit in incomplete form) for public use. A number of servers support the RTMP protocol, including Adobe Flash Media Server, Wowza and others.

Friendly name

Specify an easy to remember “Friendly” name to identify this stream output in the list of streams that have been defined (i.e. Kxxx – RTMP Stream).

Metadata source

Select a metadata input source to be used with this encoding instance (if desired). See Chapter 10 for more details.

Server address

Enter the IP address of the RTMP server.

Server port

Enter the port of the RTMP server. The default is 1935.

RTMP Publish path

Enter the publishing path for the RTMP server—generally provided by your CDN, and may also be referred to as an “entry point”.

RTMP Stream

Enter the name of the RTMP stream name. The stream name may be fixed or chosen, it depends on your CDN. Some providers, such as Akamai, require the stream name to be very specific, and they provide the format that must be used.

Use RTMP Authentication

Check this box to use authentication when sending a stream to the RTMP server.

Username

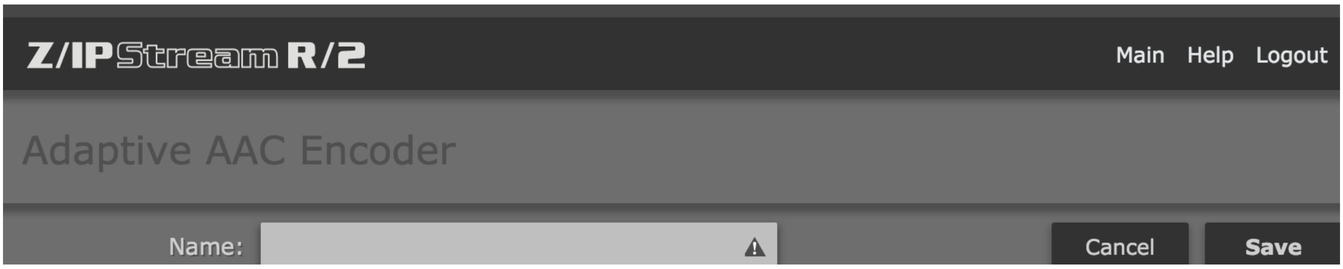
Enter the username for encoding to your RTMP server.

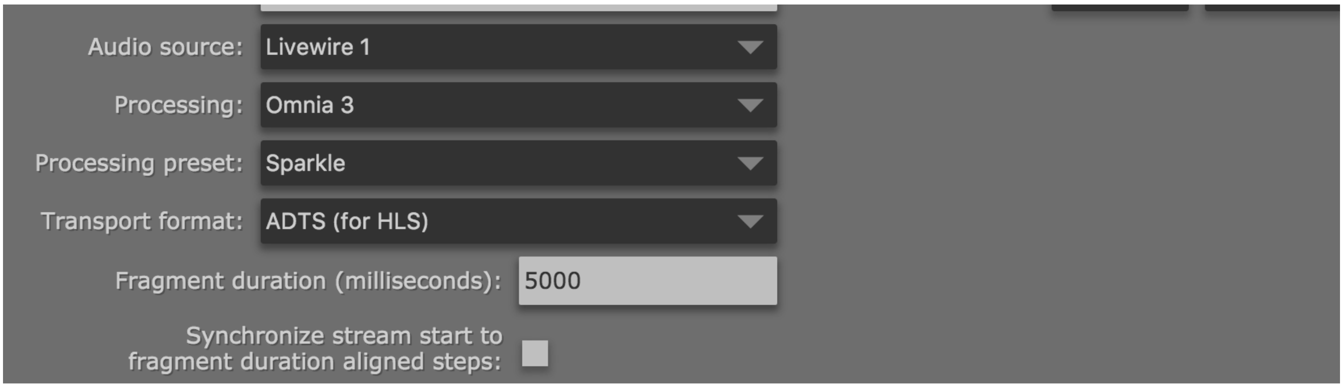
Password

Enter the password for encoding to your RTMP server.

When finished, click “Save” to save and return to the “MP3 Encoder” screen or click “Cancel” to cancel any changes.

Adaptive AAC Encoder



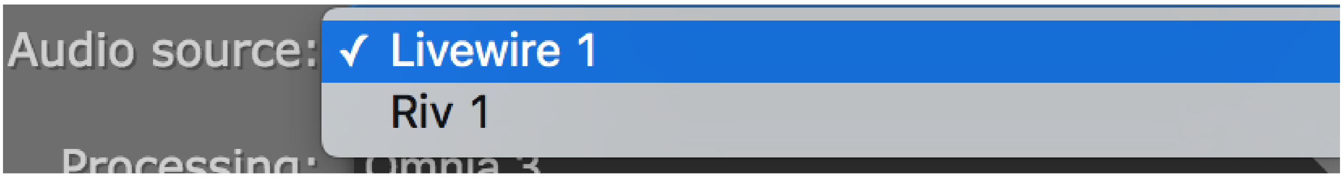


As more audio content is being streamed to mobile devices, adaptive encoders take on increased importance. An adaptive encoder sends an audio stream in multiple bitrates to the server. The client device then automatically chooses the best one based on available bandwidth. As more or less bandwidth becomes available, the mobile device may request higher or lower bitrate audio, so it is able to adapt to network conditions dynamically.

Name

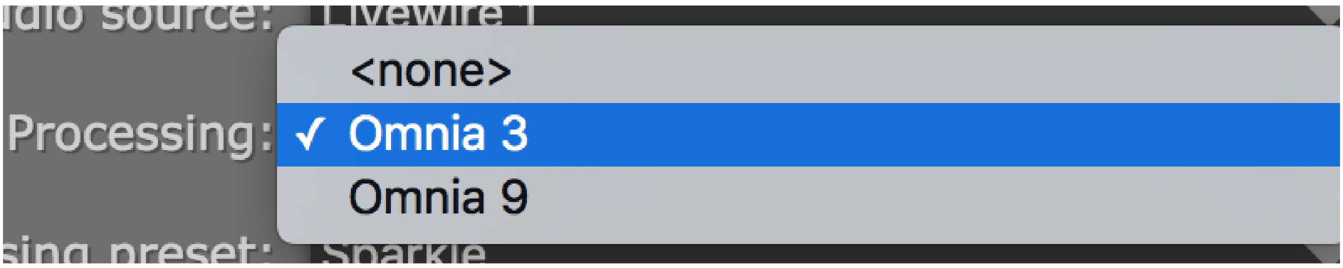
Enter a “friendly name” for this encoder instance to easily identify it for status and control from the main “control panel” screen (i.e. Kxxx-Adaptive Stream 1).

Audio Source



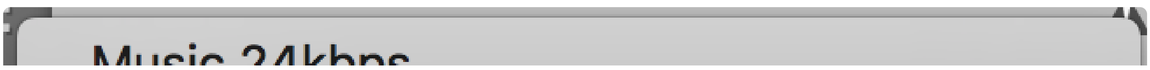
Select one of the previously defined audio sources as the input to this processing instance (see Chapter 7). Note that these sources can be used in multiple processing or encoding instances simultaneously.

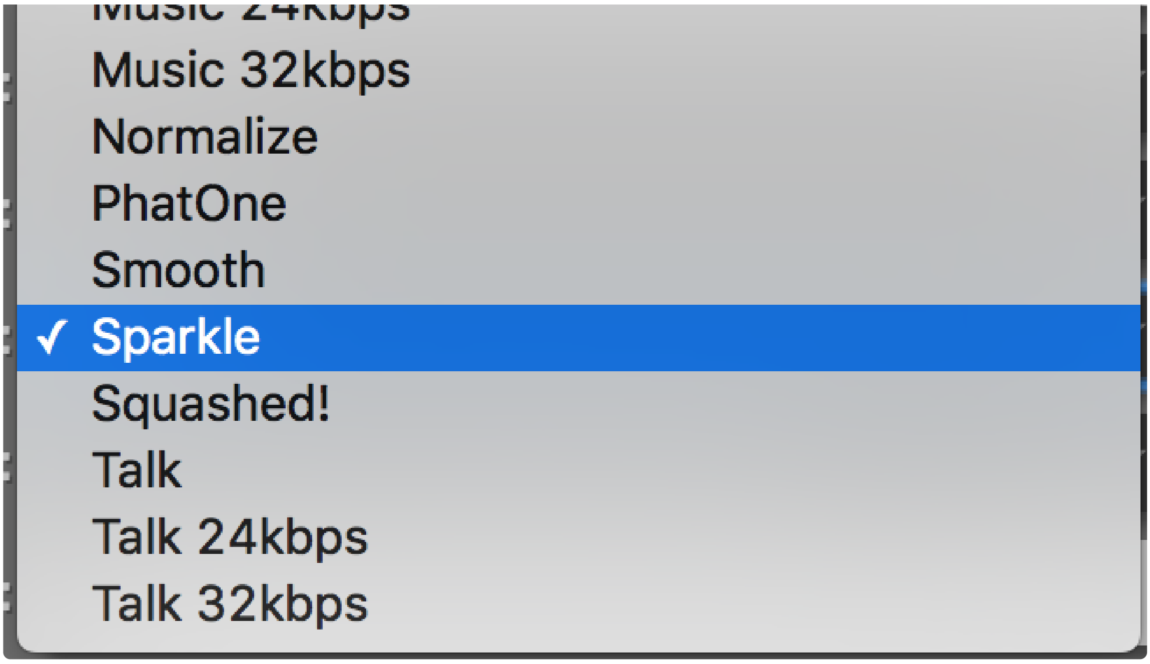
Processing



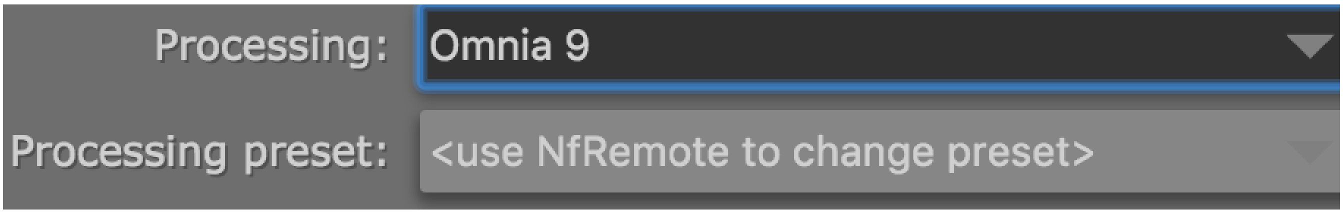
Selects between Omnia 3 and optional Omnia.9 processing (or no processing at all, should you simply want to route audio through this instance without processing). Up to 8 channels of Omnia.9 processing can be licensed per R/2. When you have reached the maximum number of Omnia.9 licenses available (or if Omnia.9 processing is not licensed) this option will not be displayed. See Chapter 16 for more information on Omnia.9 processing.

Processing preset





Selects the processing preset to be used for this instance. When Omnia 3 processing is selected, the list of available presets will appear. These presets are not format specific, though there are bitrate specific presets for “Music” and “Talk”. Experiment with various presets to find one that works best for your programming format and application. Presets can be copied and modified to create a sound customized exactly to your needs. See Chapter 9 to learn how to “fine tune” the Omnia 3 processing, and Chapter 13 for more information on Omnia 3 preset management.



If optional Omnia.9 processing is enabled, the preset must be selected via NfRemote software due to the powerful processing tools and test instrumentation available through Omnia.9. See Chapter 16 for more details.

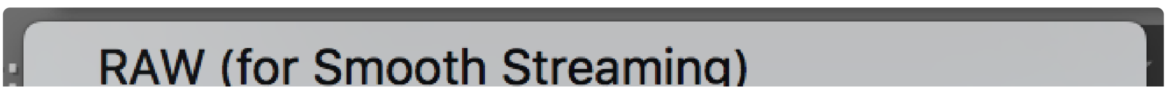
Fragment duration (milliseconds)

Configures the fragment length. Adaptive streaming splits audio into fragments which are then (usually by HTTP) downloaded by the client device. The longer the fragments, the more latency in the stream as the player will need to keep at least one fragment buffered before downloading the next one

Synchronize stream start to fragment duration aligned steps

This setting will synchronize fragment start between primary and backup servers. This option should be used together with an RTP source using absolute PTP time so adaptive streams will be started only on fragment length boundaries and fragments from primary and backup servers will be synchronized. If only one encoder server is being used, then this option should be left unchecked.

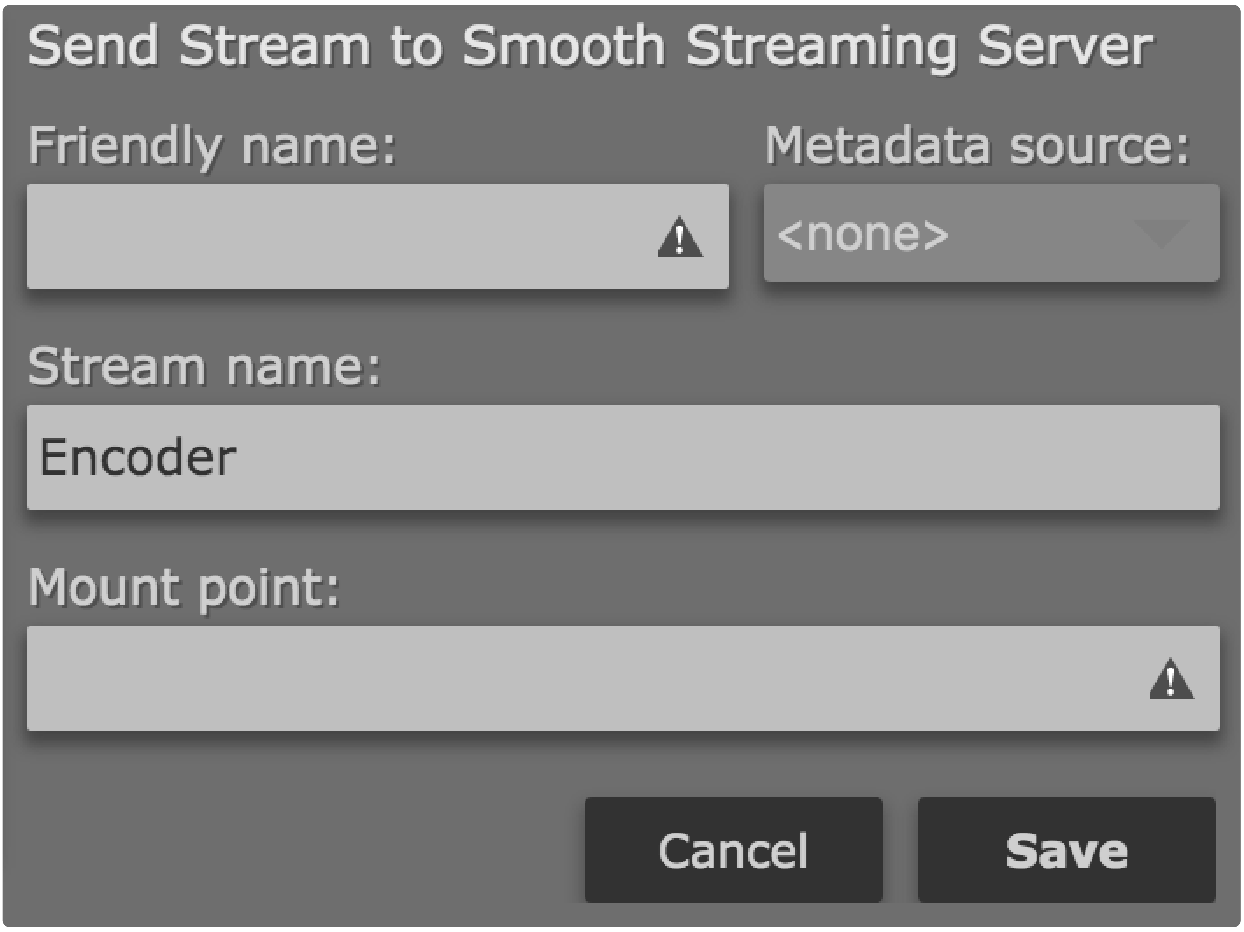
Transport Format



Select the desired transport format for this Adaptive AAC Encoding Instance. This will depend on the server platform and application. Multiple streams and bitrates can be defined in a single Adaptive AAC instance as necessary.

Note: ADIF and ADTS CRC should NOT be used with Smooth Streaming or HLS unless there are very specific server requirements.

RAW (for Smooth Streaming)



Smooth Streaming is an IIS Media Services extension that facilitates adaptive streaming to clients via HTTP. SDKs are available from Microsoft to enable Smooth Streaming over Silverlight and Windows Phone 7. It may also be ported to Apple iOS, Android and Linux

Friendly name

Specify an easy to remember "Friendly" name to identify this stream output in the list of streams that have been defined (i.e. Kxxx – Adaptive Stream).

Metadata source

Metadata is not yet available for Smooth Streaming but should be available for custom CDN servers in the

Stream Name

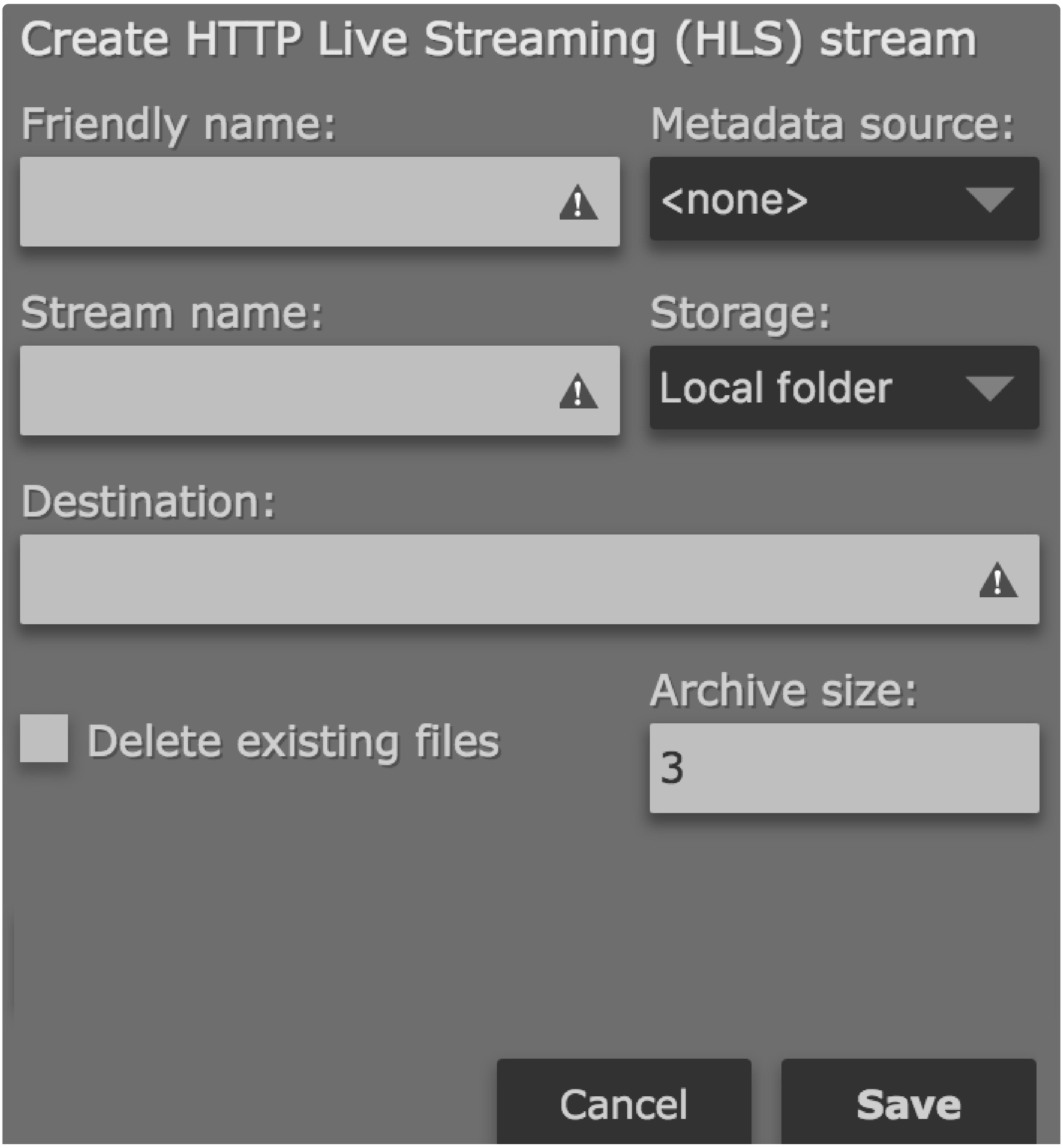
If multiple streams use the same mount point, then Stream Name can be used to separate them. The default Stream Name is “Encoder”.

Mount point

Each Smooth Streaming server may contain multiple broadcasts (or mount points) each containing a separate stream of content. Thus, 'mount point' is just a unique name that identifies a particular stream. The mount point needs to be configured properly in order for the stream to work. Mount point names should not contain any spaces or odd characters.

When finished, click “Save” to save and return to the “Adaptive AAC Encoder” screen or click “Cancel” to cancel any changes.

ADTS (for HLS)



Apple’s HTTP Live Streaming protocol allows you to create multiple bitrate adaptive streams, which allow the client’s device to request the needed bitrate stream depending on the available network bandwidth.

Friendly name

Specify an easy to remember “Friendly” name to identify this stream output in the list of streams that have been defined (i.e. Kxxx – HLS Stream).

Metadata source

Select a metadata input source to be used with this encoding instance (if desired). See Chapter 10 for more details.

Stream name

This name will be used in HLS filenames and folders. Please use only letters, numbers, dashes and underscores.

Storage

Select “Local Folder” or “FTP Server”. The “Local Folder” setting will make files available via an HTTP share on R/2 in the format: http://*r2address*/streams/HLS/*streamname/streamname.m3u8*

Destination

When set to “FTP Server”, this field specifies the FTP server where HLS file chunks should be uploaded. This field requires a full FTP path i.e. “ftp://user:password@hostname:21/folder/” to specify a username, password, and port, or “ftp://hostname/folder/” if no authentication or port number is needed. If the HLS stream is transferred to an FTP server, the server must have its own web service to serve the files to the listeners.

Delete existing files

Removes all files from the folder or FTP server when the stream is started. Use this option with caution as the files will be permanently deleted!

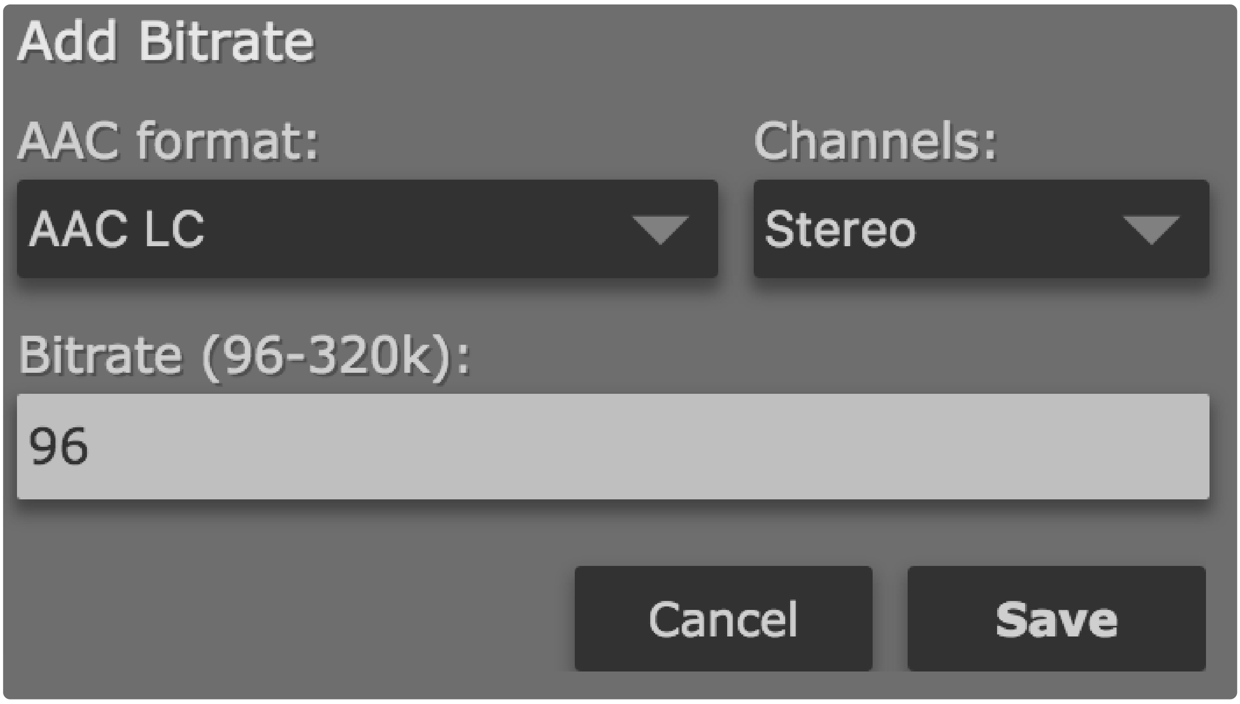
Archive size

Set number of fragments that need to be stored. The old fragments will be deleted. If this variable is set to 0, all the fragments will be stored.

When finished, click “Save” to save and return to the “Adaptive AAC Encoder” screen or click “Cancel” to cancel any changes.

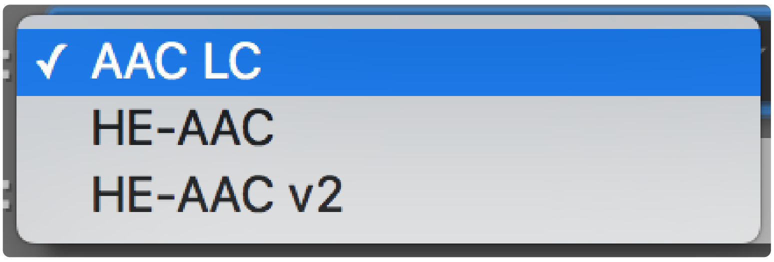
Configuring Available Bitrates

In addition to defining the other adaptive streaming parameters, you will need to configure the bitrates and AAC encoding formats contained in those streams.



Click “Add bitrate” to add an entry to the list.

AAC Format



There are several different “flavors” of AAC. The most common form (and the default for this instance) is AAC-LC (Advanced Audio Coding-Low Complexity). HE-AAC is “High Efficiency” AAC. It is designed for bitrates between 24 and 96 kbps. An “improved” version of HE-AAC (HE-AAC v2) was developed for lower bitrates between 14 and 56 kbps. HE-AAC v2 also offers improved stereo performance at lower bitrates. It is worth noting while HE-AAC v2 is widely supported, it may not be compatible with some players.

Note: If in doubt, choose AAC-LC. This is the most common form of AAC encoding, though the HE (High Efficiency) variants are becoming more common for lower bitrate applications.

Channels

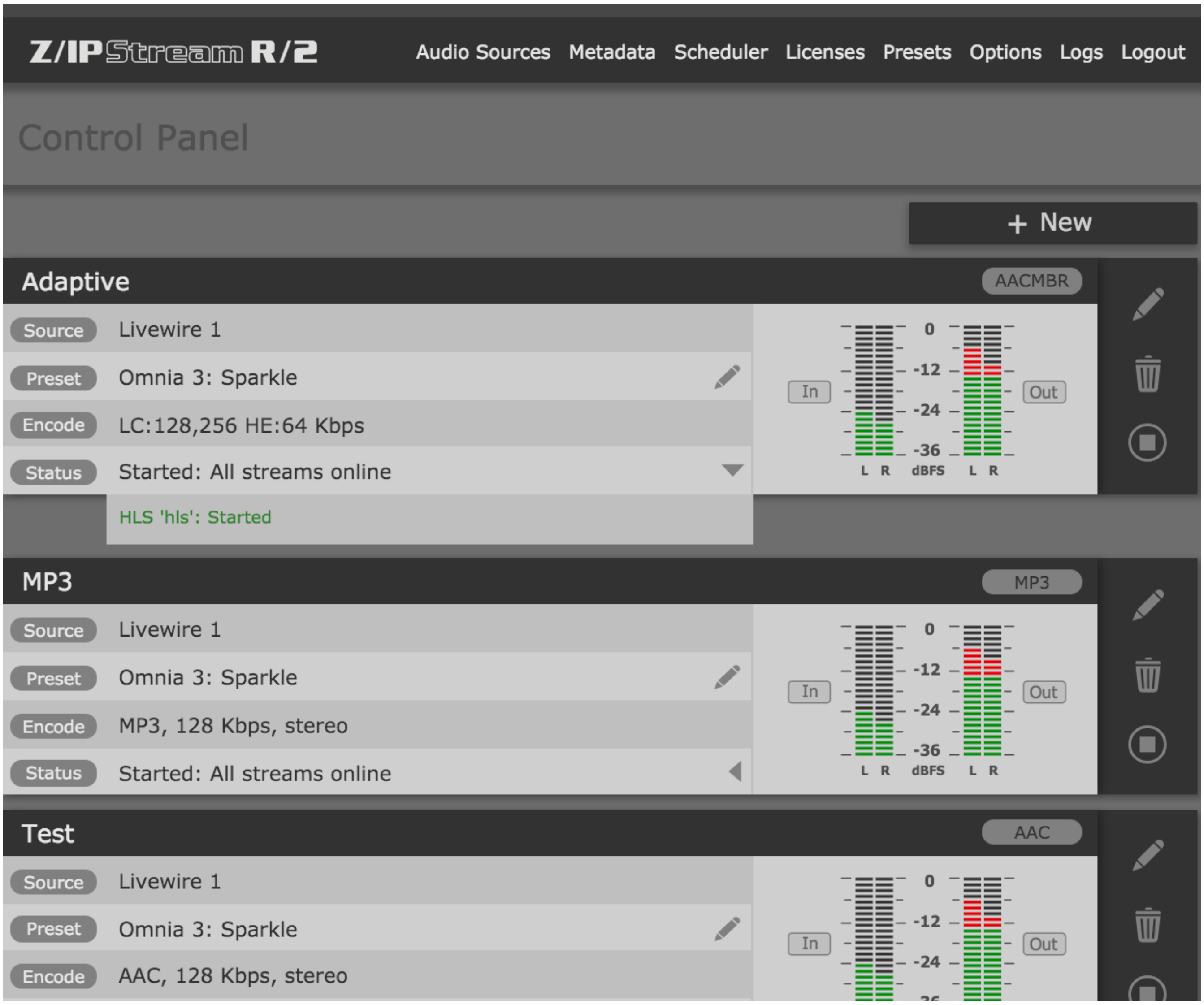
Select whether this stream will be encoded in Mono or Stereo. Mono should provide a mono sum of the incoming audio. This setting will be locked to stereo for HE-AAC v2.

Bitrate

Enter the bitrate for this stream. The valid bitrate range will change depending on which AAC format is selected.

When finished, click “Save” to save and return to the “Adaptive AAC Encoder” screen or click “Cancel” to cancel any changes.

Managing Processing and Encoding Instances



Click “Main” or the Z/IPStream R/2 logo to return to the main “Control Panel” screen. All processing and encoding instances that have been defined will be displayed, along with their configured parameters and status. Click the “triangle” on the “Status” line to display additional details about the individual streams contained within each instance.

Instance Type

The oval at the top right of each instance will display the type of instance that has been defined.

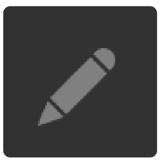
Indicates an audio processing instance

Indicates a standard AAC encoding instance

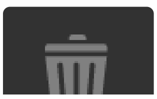
Indicates an MP3 encoding instance

Indicates a multiple bitrate (adaptive) AAC encoding instance

Instance Controls



Edit the parameters of an instance. Some parameters may not be changed once an instance has been defined, or while an instance is running.



Delete an instance. This is a permanent action. Deleted instances cannot be recovered and will need to be re-created.

Starts an instance that is currently stopped. Instances that are not currently running will be “greyed out”.

Stops an instance that is currently started.

Metering and Processing Control

I/O Metering is provided for each instance. Audio processing presets for each instance can be modified by clicking the “pencil” icon next to the preset name (see Chapter 9 for more details on Omnia 3 processing). Instances using Omnia.9 processing can be accessed via NfRemote for more detailed monitoring and control of the audio processing (see Chapter 16).

Omnia 3 Processing Setup

Overview

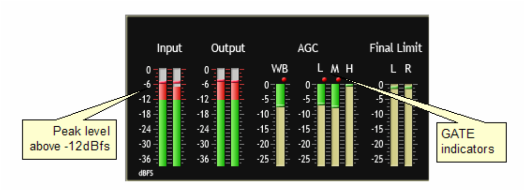
Standard audio processing for Z/IPStream R/2 is the Omnia 3-band processor. Omnia 3 is intuitive and easy to use. Applying audio will activate movement on the bar graphs, and this is confirmation that processing is occurring.

All audio processing adjustments are performed in the Preset Editor. A “preset” is simply a large table of values representing all of the control values for every control listed in the menu tree. When you choose a preset, the control values are loaded into the processing structure, reconfiguring Omnia 3.

We recommend exploring the provided factory presets. Then, once you find the preset that most closely matches the sound you’re after, you can customize it to your liking. More information about Omnia 3 preset management can be found in Chapter 13 of this manual.

Understanding the Omnia 3 Bar Graphs and Indicators

The bar graphs are capable of indicating more than just level information. The texture and density of the audio signal can be observed, based upon the dynamic action of the bar graphs, and peak-responding “pills”. Of interest are the “pills” at the end of the input and output meters, as they indicate peak level. The bar section represents the RMS average of the signal. Wide dynamic range will display a separation between the pills and the bar, whereas signal with little peak information will cause the pills to ride on the crest of the bar graph. The bar graphs can indicate up to 25 dB of gain reduction.



The bar graphs provide a wealth of information about the processing control in Omnia 3. The algorithms automatically adapt the style of compression/limiting control being employed on a moment-by-moment basis. This can be deduced if the metering is studied over time. During normal operation, the indicators will have a dynamic “bounce” that you will be able to get a feel for. Every now and then, you will see the processor react quickly and show a larger amount of gain reduction. Notice that the response is very fast. This action will recover very slowly, and return to “rest” with the main bar graph. This action will be easily noticed on material that is very dynamic in texture. More on this below.

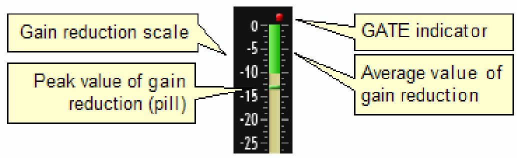
Another feature unique to Omnia 3 is processor “Hold”. During brief pauses in audio, the bar graphs will “freeze” and the GATE indicators will show the processor has entered the Hold mode of the algorithm. This is most easily noticed when there is a “dry” voice being processed.

The Input/Output level meters will change from green to red whenever the level exceeds –12dBfs. The pills also change color on the Input/Output displays. In the dynamic processing section, the pills maintain the same color.

Note: The input and output levels are displayed relative to 0dB full scale (0dBfs). The 0 indicator on the input/output bar graphs means that every available bit of signal level is being used at that time! There is nothing more in the level department, except to create distortion… nasty sounding distortion!

Interpreting the Gain Reduction Meter Displays

Through careful observation of the processing bar graphs, significant information can be acquired and analyzed about the audio signal on a moment-by-moment basis. The bar shows the average value of the gain reduction, while the floating pill indicates the peak value of gain reduction.



Since the processing algorithms in Omnia 3 ‘adapt’ to the audio, it is capable of performing many different processing functions at different times. The processing display bar graphs may appear to indicate different kinds of operation with different program material. This behavior is primarily based on dynamic range differences in the applied program material. For program material that is already processed or lacking dynamic range, the bar graphs will indicate differently than with material that possesses a high degree of dynamic range. This is because audio signals that lack dynamic range, naturally or by previous processing, will possess a lower peak to average ratio. Conversely, audio signals with a wide dynamic range possess a higher peak to average ratio. The Omnia 3 adapts differently to each case.

For program material that has low dynamic range (or high RMS and low peak levels), there will be more activity in the WB-AGC sections and less activity in the Multiband Processors. This behavior is due to the WB-AGC response to high RMS energy, while the Multiband section is reacting to lower peak energy. Sometimes the Multiband section may not indicate any action at all. For Omnia 3, this is completely normal! Examples of this behavior might be seen with heavily processed commercials or music, or with music passages of sustained level. A good example is the Aerosmith album “Pump.” This recording has very little dynamic range. Try any up-tempo selection from this recording and you will see the Wideband-AGC section make initial adjustments along with small amounts of limiting. Once adjusted, the Wideband-AGC section will stay deep in gain reduction, although with little movement, and action within the limiters will be minimal.

When the programming has wide dynamic range (low RMS and high peak levels), the opposite will occur. The Multiband section becomes active, while the WB-AGC section will appear not to respond as much. During these events, the Multiband sections could be working aggressively, while the WB-AGC section indicates little activity. Good examples of programming able to cause this behavior: vocal passages, live voice, classical music, and passages with high transient levels. Good test examples: almost anything classical, as well as almost anything by Steely Dan!

Also keep in mind that the WB-AGC section is designed to operate much more slowly than the Multiband sections, primarily because of the nature of each function. Remember that the WB-AGC section operates on the program’s RMS energy. During gain calculations, the incoming program’s “average” level is established, and gain adjustments, if needed, are made based on those calculations. This is why the WB-AGC sections will appear to move slower as it makes changes over relatively long time periods.

The intent of the Multiband section is to normalize the spectral balance and provide control of the peak levels. Peak energy must be detected and adjusted in a quick and accurate manner while, at the same time, not interfering with the sonic integrity of the audio signal. For this reason, the Multiband sections operate faster, with special background instructions to govern their behavior, and strictly on an as-needed basis.

Since the processing displays are capable of providing a wide range of information, we do not recommend setting up the system based on any specified meter indications. Instead, we recommend setting up the processing by using your ears to judge the sound. We’ve provided the meters only to analyze the signals and aid you in adjusting the specific parameters needed to achieve your desired sonic results. Like a speedometer, the meters are a guide, not the road!

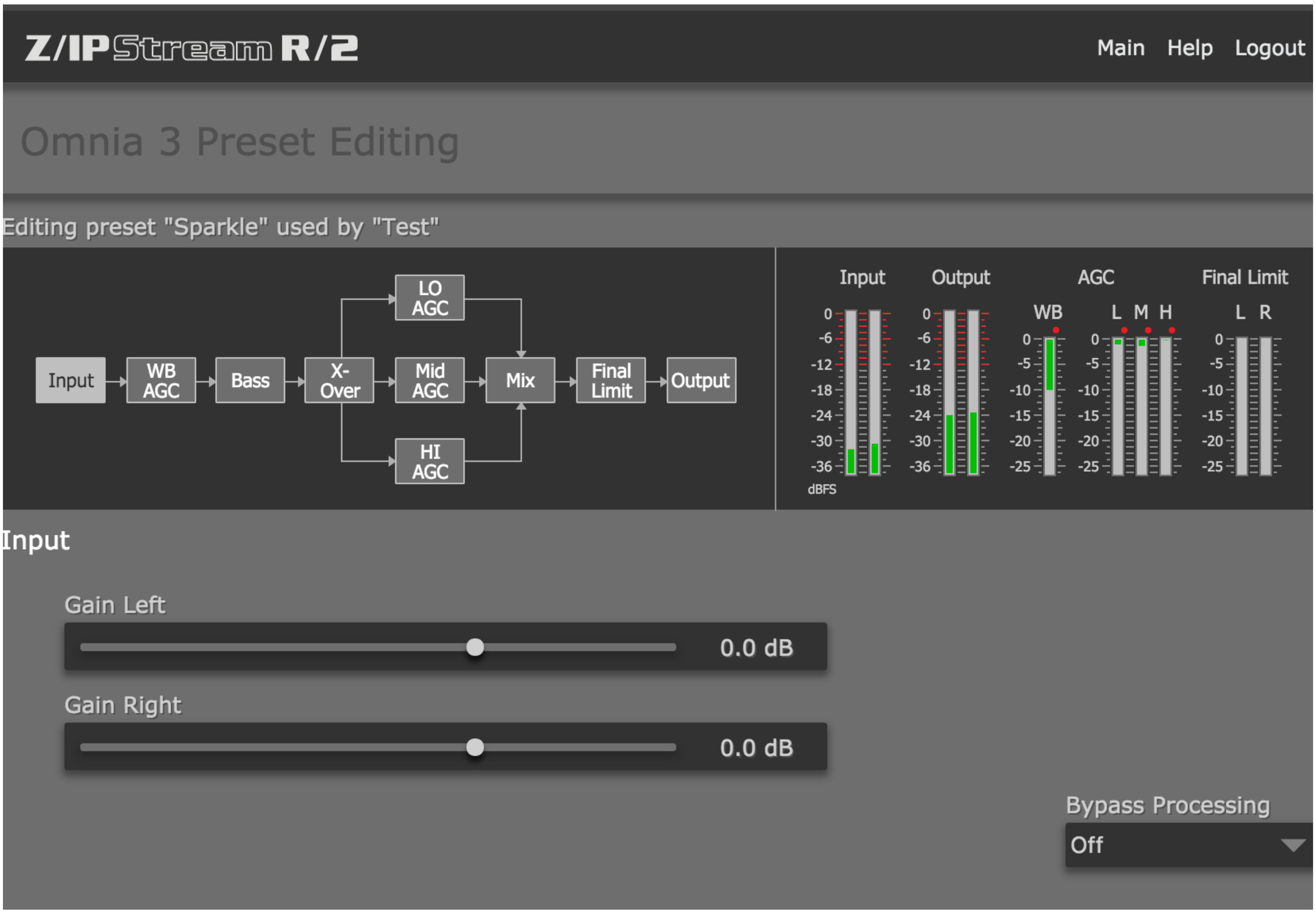
The red ticks that appear above the WB (Wideband AGC), L (Low-Band AGC), M (Mid- Band AGC), and H (High-Band AGC) bar graphs are the GATE indicators. They display GATE activity. The GATE function is described above in the section titled “Understanding the Bar graphs and Indicators”

Editing Presets

Once a at least processing or processing/encoding instance has been defined, you are ready to begin the editing process. From the main “Control Panel” screen, click on the pencil icon to the right of the preset name in the desired instance, and you will be presented with a block diagram of the Omnia 3's audio path. Clicking on any of the blocks takes you to the controls available for that section. Once you've made your adjustments, click **Save** or **Cancel**, depending on whether or not you like what you hear.

Input

Clicking on the Input block displays the input section sub-menu. There are adjustments for the individual Left/Right channels. Drag the slider-bar to adjust the respective level. There is a 30dB range of adjustment between –20dB and +10dB.



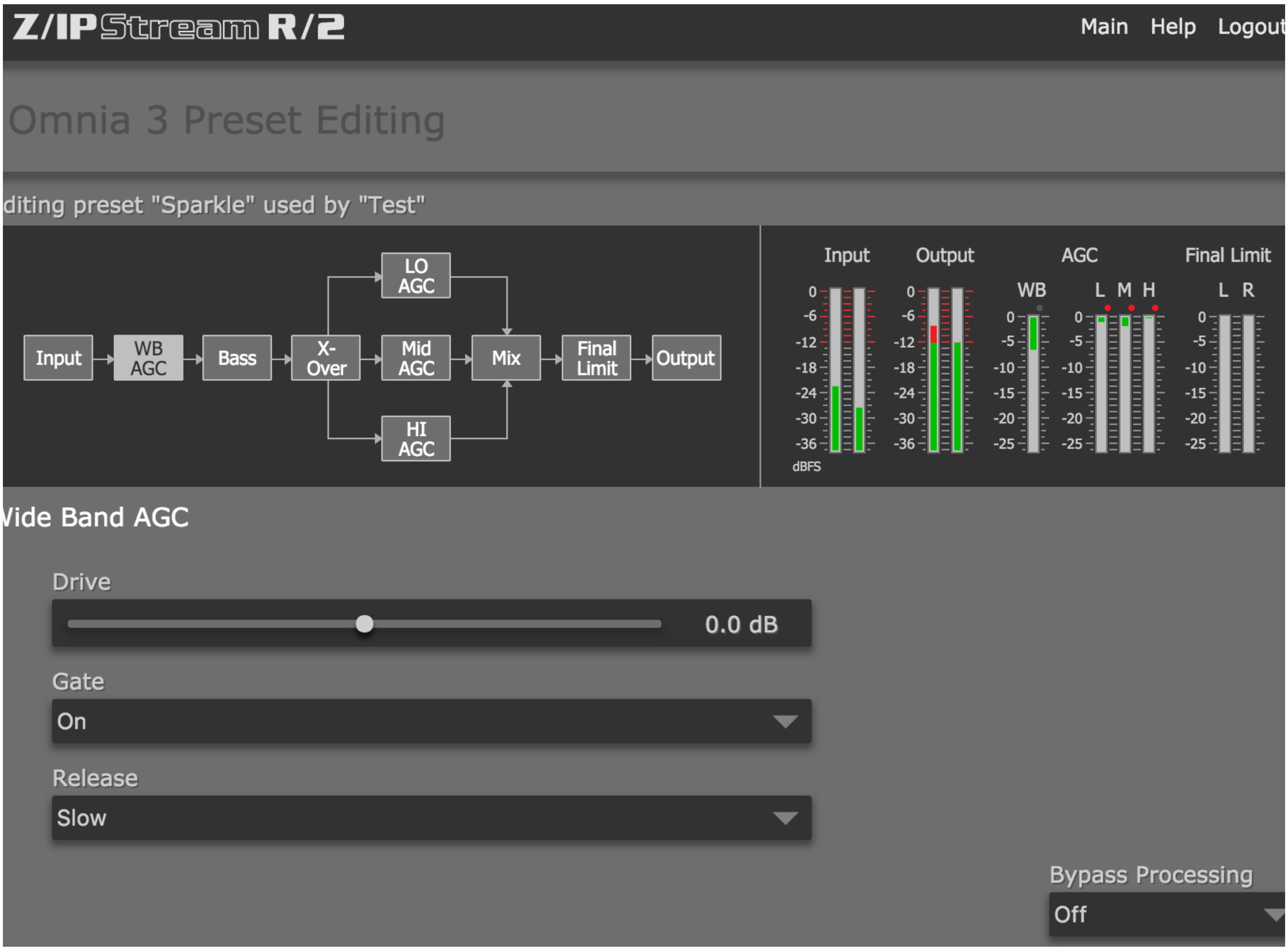
Using normal program audio, a correct gain setting results in peak indications (the“bouncing balls”) hitting -12 dBFS (where the level meters turn red) or a little higher. This corresponds to system headroom of 12 dB. You may adjust for more headroom if you wish (lower indications), but setting the input level for less headroom (higher meter indications) is strongly frowned upon.

Wide Band AGC

A very flexible wideband leveler section provides smooth, transparent control of the input program. This is achieved through two significant Omnia innovations: a dual referenced release gate and a hidden, intelligent “makeup” gain algorithm. The dual gate reference is a unique process that correlates the dynamics of the audio input signal to a “rolling” reference level, and from that information makes conditional decisions affecting the character of the release function.

A hidden, Make-Up Gain, control signal determines when the amplitude of the input program suddenly falls to a reduced level. It then adjusts the side chain gain in order to “fill in” the softer program passages so that the average level is increased. This allows the AGC function to operate with slower time constants, while significantly increasing the average audio level. These slower overall time constants yield lower intermodulation distortion, contributing to Omnia's trademark sound.

Using Classical music for an example, the orchestra often plays forte, and then enters abruptly into a quiet passage. Conventional AGC algorithms would hold the softer passage down until it was able to slowly recover at the static release time setting, making such passages nearly inaudible at normal listening levels. Omnia 3's makeup gain function allows a hidden, faster time constant to provide quick recovery, but only during the softer passage. As soon as the orchestra starts to play louder, the "makeup" time constant yields control back to the primary AGC circuit, returning gain to the previous platform level. This sophistication preserves the dynamic integrity of the signal while greatly enhancing the listenability of the program.



This is the first processing section within the Omnia 3, so the controls in this section affect the overall signal, its density, and hence affect every other processing section following it. The job of the Wideband AGC is to erase long-term audio level fluctuations, while doing so in a very unobtrusive manner. This sub-menu provides a few parameters that tailor the action of this section.

Drive

The sets the audio level that enters the WB-AGC. Increasing the Drive will produce more compression. This control is calibrated between –6dB and +6dB, and adjusts the signal level into the Wideband AGC. It should be adjusted to net approximately 12dB of gain reduction with typical programming. Too little gain reduction can defeat the "leveling" function of the Wideband AGC. Too much gain reduction has little additional benefit. Nominal gain reduction values for the Wideband AGC are between 10 and 15dB.

Gate

When set to ON, it will freeze the gain at the last level of processing action that occurred before the audio signal fell below the threshold of operation. This control helps to minimize “pumping”, and the increase of background noise during pauses in programming.

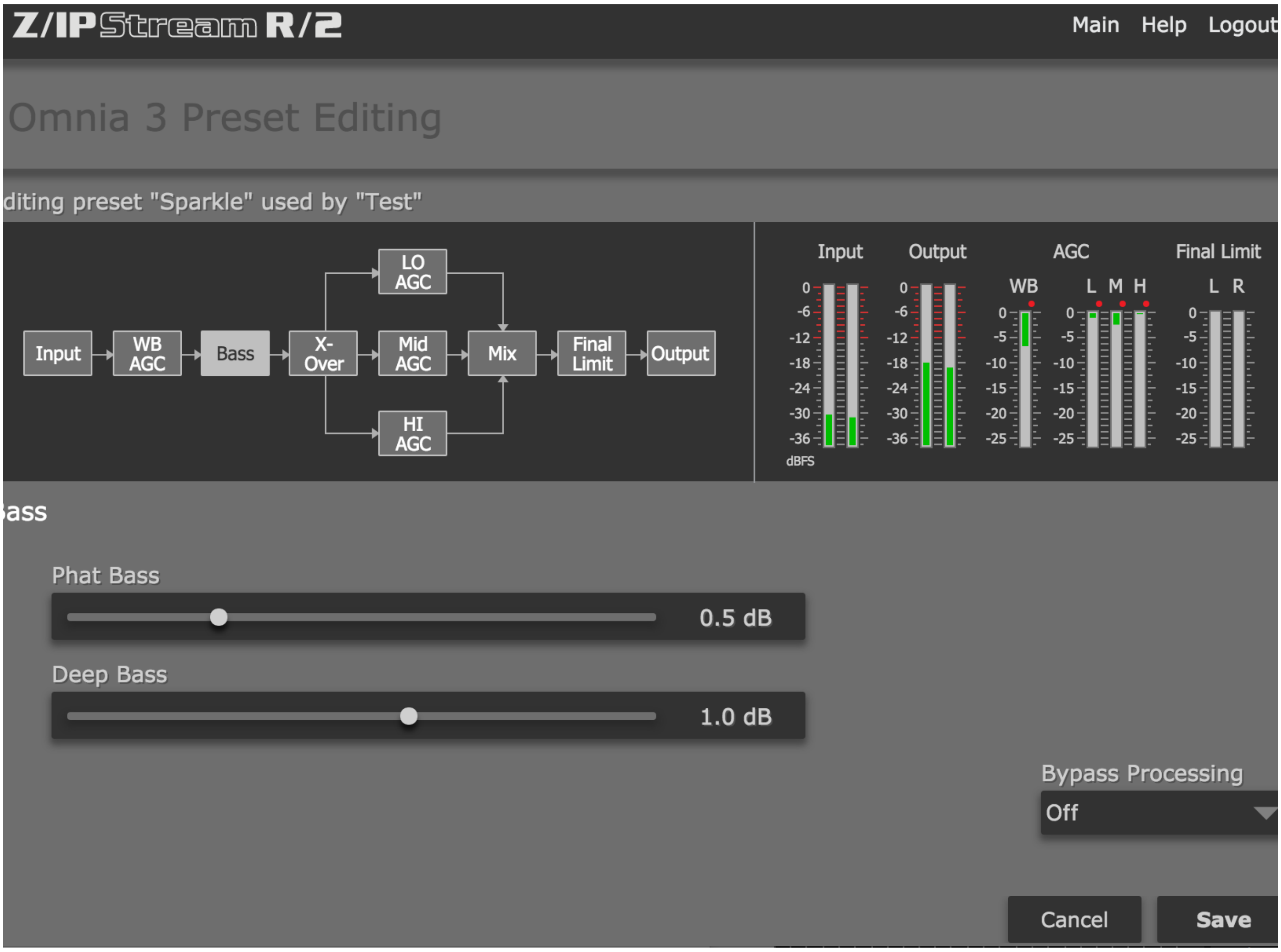
Release

Controls the speed of recovery for any given amount of gain reduction. Faster action yields less dynamic range and the presence of more density to the audio. The Release control sets the time constants in relative terms using Slow, Medium, and Fast. Because the time constants are program-dependent, calibrating these controls in absolute time values (ms/dB) would be useless—therefore we use simple terminology.

Clicking Save after adjusting any of the parameters in this section will save the change and close the sub-menu.

Bass

This section contains the Bass EQ controls. In the the unique Omnia 3 signal processing architecture, these controls are placed in their optimal position — just before the multiband processor sections. Each of the Deep Bass and Phat Bass controls can boost the level up to 12 dB. Be careful here not to overdrive the following sections or over emphasize these lower frequency ranges. When used properly this specialized low frequency enhancement tools can deliver the thunderous bass and warmth that the Omnia is known for, and it can do so without making the sound muddy.



Phat Bass

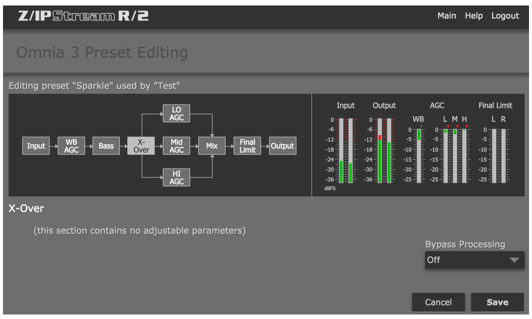
A unique enhancement that adds filtered harmonics of the lower registers to the upper bass frequencies. The algorithm extracts information contained in the initial attack to do its work, and low frequency texture is therefore emphasized. Older recordings sound fuller (or phatter) with the added illusion of loudness. Phat Bass EQ also helps devices with small speakers sound like they have more bass than they actually do. A little of this effect goes a long way so be careful not to apply too much boost. Be sure to listen on several different types of both fixed and mobile media devices to arrive at a setting that works best for all of them.

Deep Bass

For those who demand thunderous bass, the Omnia 3 has it! Up to 6 dB of “thunder” can be dialed in. And we're not talking about a simple bass boost, but a rather sophisticated concept that takes into consideration the time alignment of the low frequencies as they pass through the entire Omnia system. It allows a loud, clean low end, with absolutely no sacrifice to the overall loudness of your signal. We won't tell you where the control is in the signal chain. But we will tell you that Deep Bass function is a shelf boost at 90 Hz, utilizing a phase compensated 12dB/octave slope to emulate the EQ function.

X-over

This section contains no adjustable parameters, but here's a bit of information about why we did what we did. Most multiband audio processors make a compromise in the crossover area in order to provide a flat dynamic frequency response. This is done to avoid audible peaks or dips in the recombined frequency response as the individual band gains change during processing. In an effort to minimize this problem, most processors use a fixed phase-offset in the crossovers. While such treatment helps to minimize frequency response peaks and dips, it also results in a loss of phase linearity, usually causing an increase in “smearing”, as well as reducing musical clarity.



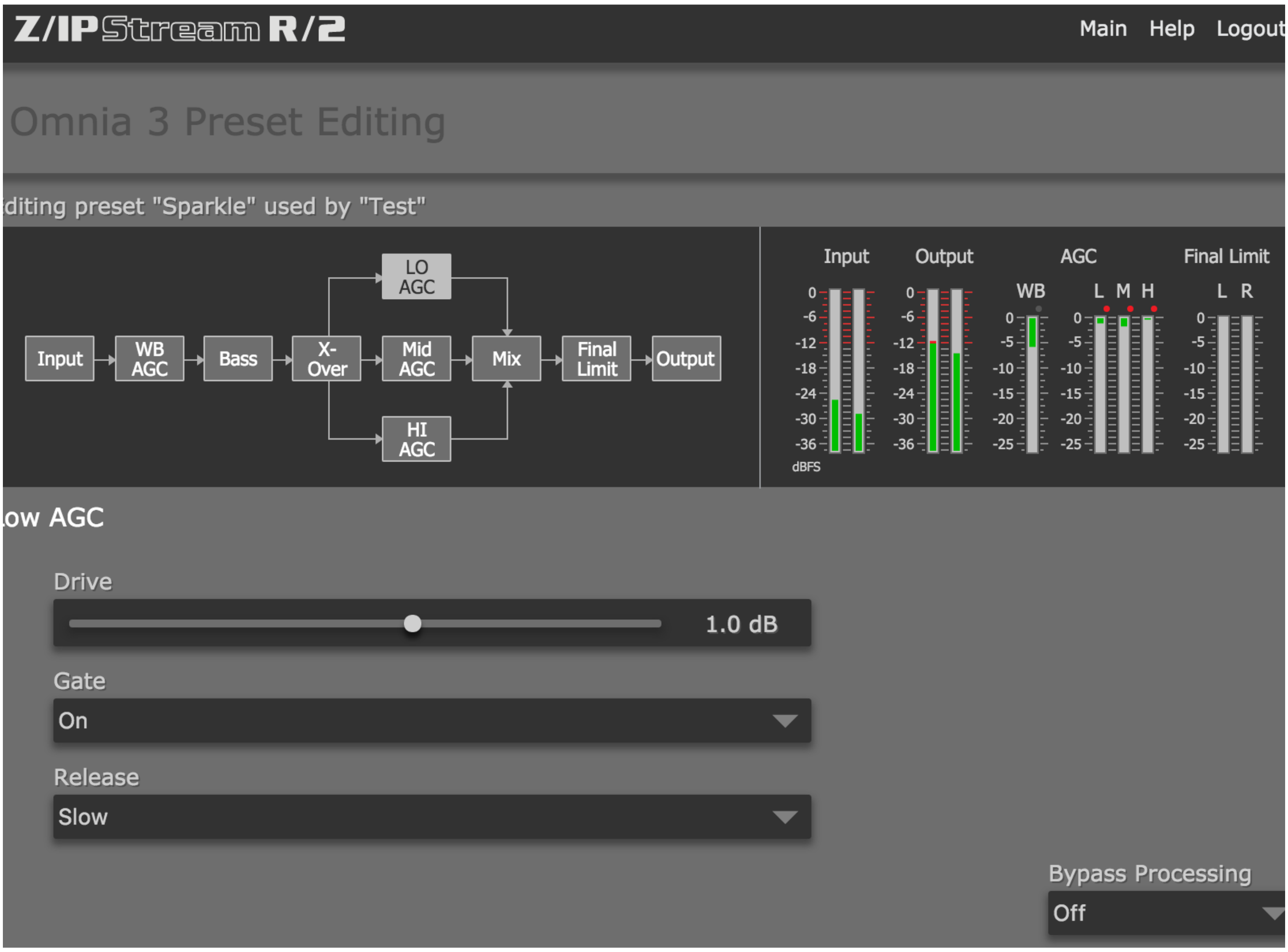
In Omnia 3’s implementation, the crossover network is carefully time-aligned so that the recombined spectrum remains flat, regardless of the amount of gain control being applied within any band. This true phase linear response assures that harmonic overtones are not displaced in time. The result: the truer, more natural, and more musical Omnia sound.

Multiband AGC Sections

Similar in character to the wideband AGC described above, but expanded to three bands, this section has the ability to significantly add power and loudness to the audio while tailoring and pre-processing it for delivery to the final limiter section. The system uses different algorithms for each of the bands: The low and mid bands use a feedback configuration, which produces a larger, warmer sound on lower frequencies. The high band utilizes a feedforward design that maintains a more open, natural and musical texture on higher frequencies.

As with any multiband processor, improper adjustment can exaggerate noise at high frequencies, particularly with older recordings that possess a good deal of tape hiss. This can especially occur if the highest AGC band is driven so that significant gain reduction occurs, and the band’s gating control is set to the lower end of the scale. During audio segments that have little or no HF content, the AGC will try to increase its gain in an effort to restore treble balance. Even during periods when the gate function is in the default Freeze mode, the problem might still exist with some programming.

Since the user parameters are the same for all three bands, only the LF Band will be detailed here.



Drive

Sets the audio level that enters the AGC. Increasing the Drive will produce more compression. This control is calibrated between –6dB and +6dB, and adjusts the signal level into the AGC. It should be adjusted to net approximately 12dB of gain reduction with typical programming. Too little gain reduction can defeat the “leveling” function of the Wideband AGC. Too much gain reduction has little additional benefit. Nominal gain reduction values for the AGCs are between 10 and 15dB.

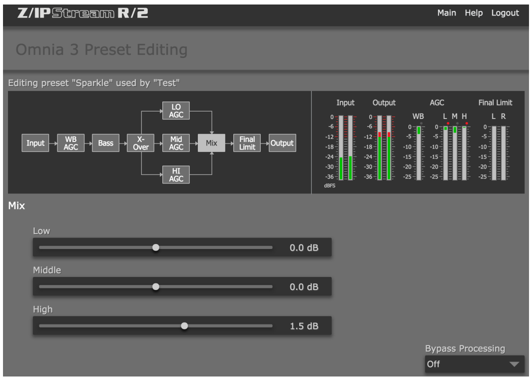
Release

Controls the speed of recovery for any given amount of gain reduction. Faster action yields less dynamic range and the presence of more density to the audio. The Release control sets the time constants in relative terms using Slow, Medium, and Fast. Because the time constants are program-dependent, calibrating these controls in absolute time values (ms/dB) would be useless – therefore we use simple terminology.

Gate

When set to ON, it will freeze the gain at the last level of processing action that occurred before the audio signal fell below the threshold of operation. This control helps to minimize “pumping”, and the increase of background noise during pauses in programming.

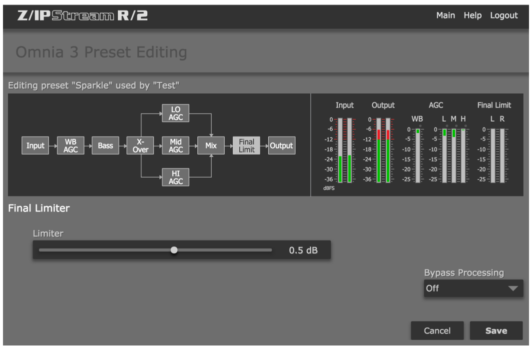
Mix



This is where the three multiband processors are mixed together. Use care in adjusting this section, as too much level from one particular band may cause an excessive amount of emphasis to that range of frequencies. Such adjustment may also drive the final limiter bands in that range of frequencies too hard, causing the sound to become unnatural, dense, and squashed. As explained earlier, if your desire is to “EQ” the sound, you should perform that function using the drive levels in the multiband section. The mixer is primarily intended for minor “EQ” changes to the overall sound. There is a +/-6dB range of adjustment for each band.

Final Limiter

Omnia 3 employs a lookahead limiter to provide absolute and precision peak control. This limiter has been designed to minimize processing side-affects like IMD, which are usually associated with limiters of this type. Using an innovative design that cancels intermodulation products before they develop, allows this limiter to sound extremely transparent. The limiting function is derived using numerous control signals that monitor one another.



This type of peak controller is used instead of a clipper because it does not generate the same high levels of THD as a Clipper does. THD causes added difficulties in a coded audio system, as the harmonics generated from the clipping action, create added artifacts in the encoder. These are especially annoying at high frequencies.

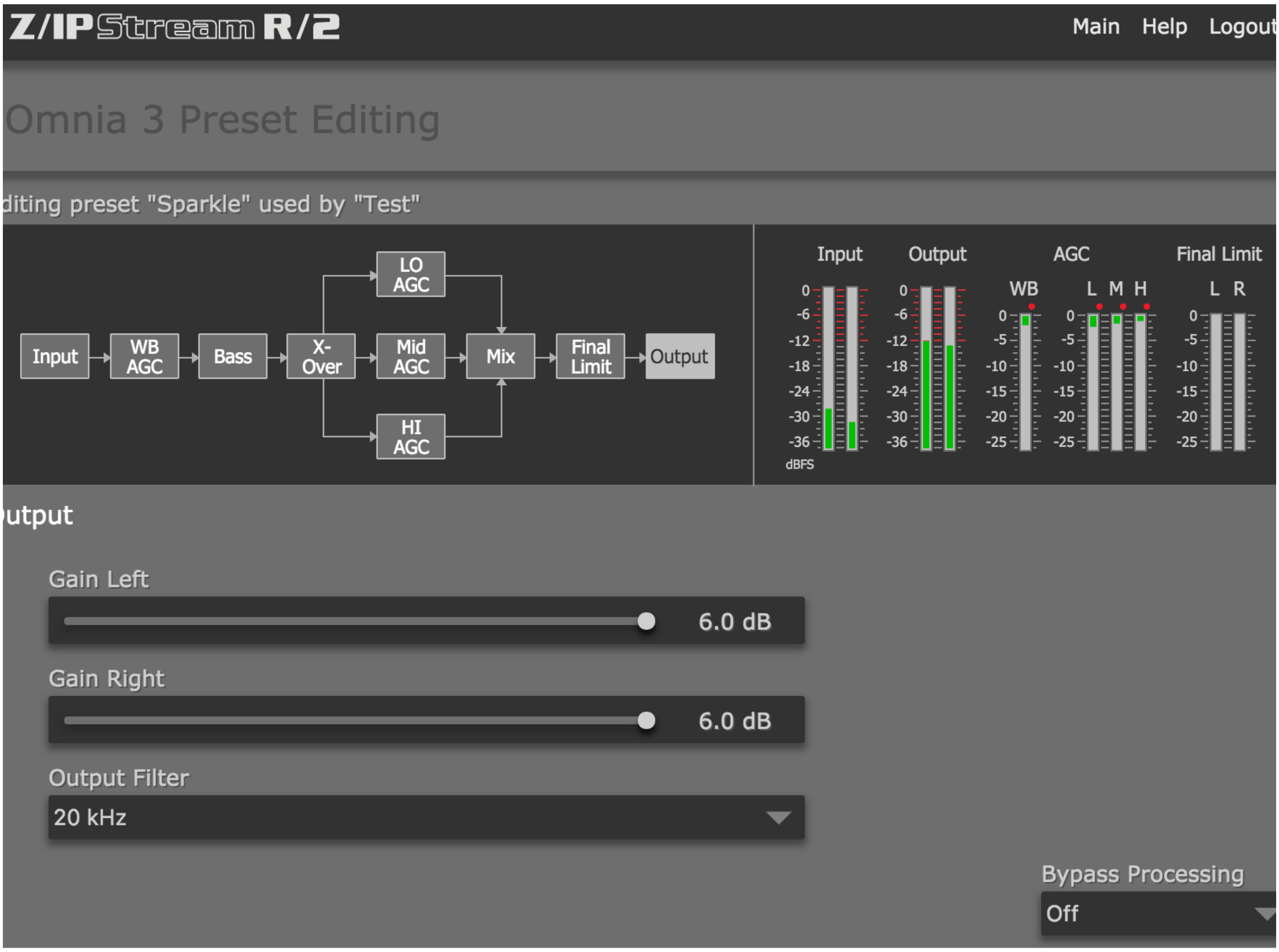
On the other hand, a lookahead limiter, yields extremely low levels of THD, although it will create some IMD component, and this allows the audio coder to operate with minimal sonic artifacts. There are tradeoffs in how each of these peak controllers sound when they are set to produce added loudness. When a clipper is pushed, the audio may appear edgier. This is from the added harmonic content. In contrast, the lookahead limiter will appear busier, or dense as the action of the control signal may be heard, when more limiting is applied.

Limiter Drive

This is where the loudness versus quality is most evident! This limiter's adjustment range is +/-3 dB, in 0.1dB steps. It is advisable to make minor changes, primarily as the "loudness fine tuner". Be careful, there is a lot of power here! A little bit goes a long way!

Output

The peak output level adjustment is done using the individual Left/Right output controls. The control range is from –26dB to +6dB of gain. This level is normally set to the maximum input level, or just below it, of the device that the Omnia 3 is connected to.

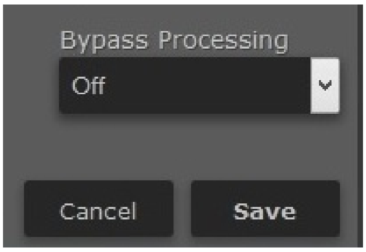


Output Filter

For webcasting applications, especially at lower bitrates, it is sometimes desirable to reduce the audio bandwidth. This is particularly useful if you are using a low cost sound card to feed and external encoder as the filtering in many sound cards in inadequate.

A pull-down menu provides eight different filter response curves that can reduce the spectrum down to 4kHz.

Bypass



The processing can be bypassed simply by clicking on the BYPASS button located in the lower right hand corner of the processing panels. This is a simple and fast method to compare the before and after affects of the processing.

To save changes to the current processing preset and return to the main “Control Panel” screen, click “Save”. Click “Cancel” to return without saving changes.

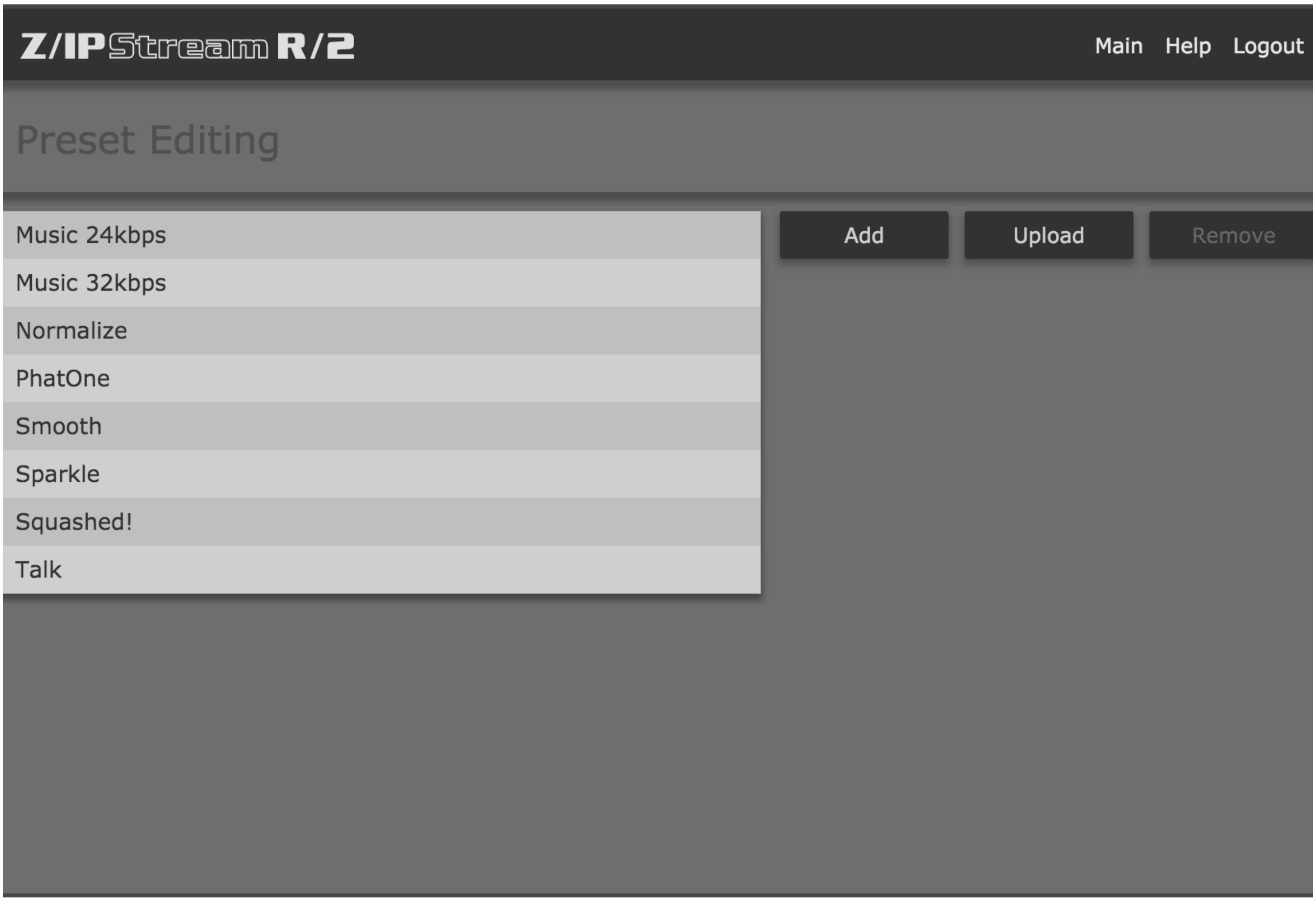
Omnia 3 Preset Management

Overview

R/2 is equipped with Omnia 3 processing on all audio channels. The “Presets” screen allows creating new presets and managing existing Omnia 3 presets on the system. More information on editing Omnia 3 processing and presets can be found in Chapter 9.

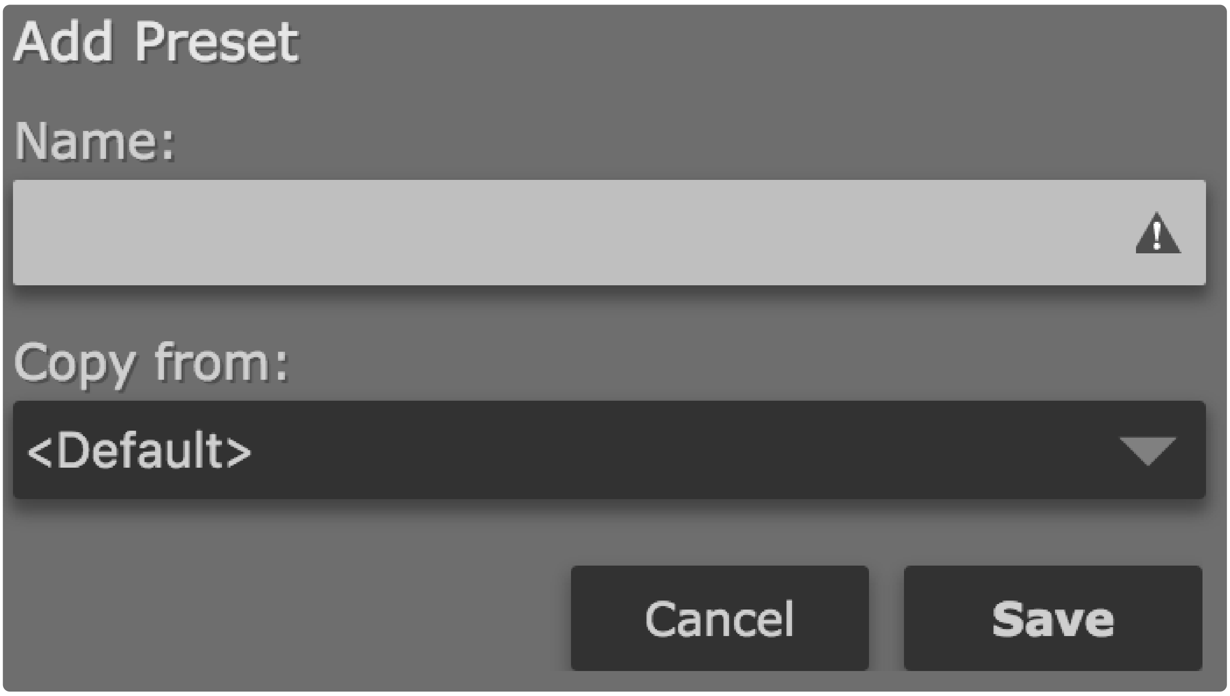
Note: Omnia.9 presets are managed through NfRemote software (see Chapter 16).

Preset Editing



Click “Add” to create a new preset based on an existing factory preset, or click “Upload” to upload a compatible preset file. See Chapter 9 for more details on modifying Omnia 3 processing.

Add Preset



Name

Enter a “Friendly Name” for this preset (i.e. Kxxx – HOT HOT HOT) so it can be easily located in the list of presets.

Copy from

Select a factory preset as a starting point for this new preset.

Click “Cancel” to return to the presets screen without saving changes, or “Save” to create the new preset and return to the presets screen.

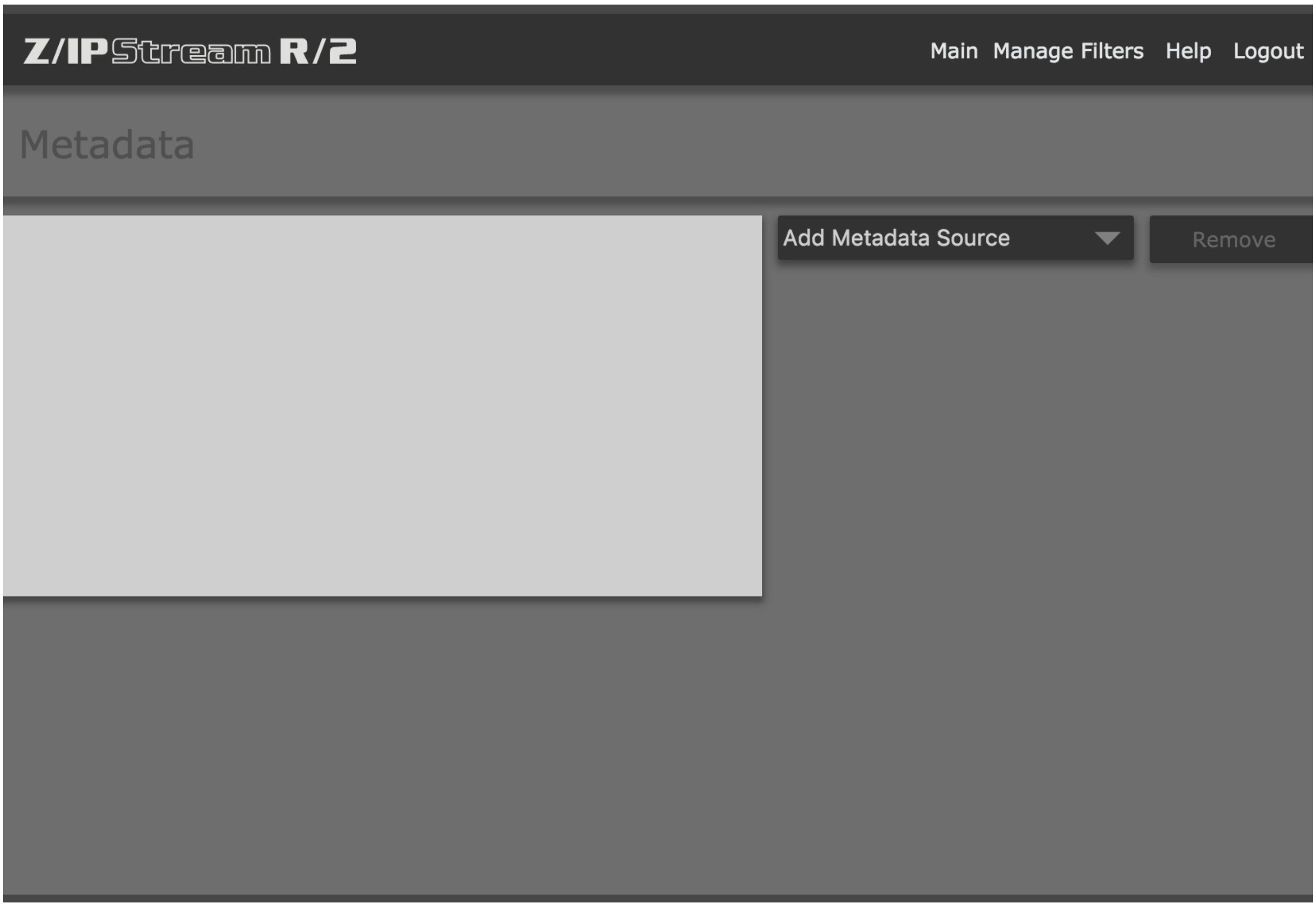
Metadata Sources and Filters

Overview

The metadata subsystem gives you the option to tag your audio stream with “now playing” text, album covers and other graphic information.

The metadata is received from an external system over TCP/IP or UDP, then passed to a metadata filter for parsing. RS-232 serial metadata will be supported in a future version. Once the filter detects a complete record, it sends the tagging information to the Stream Sender which includes it in the outgoing stream data.

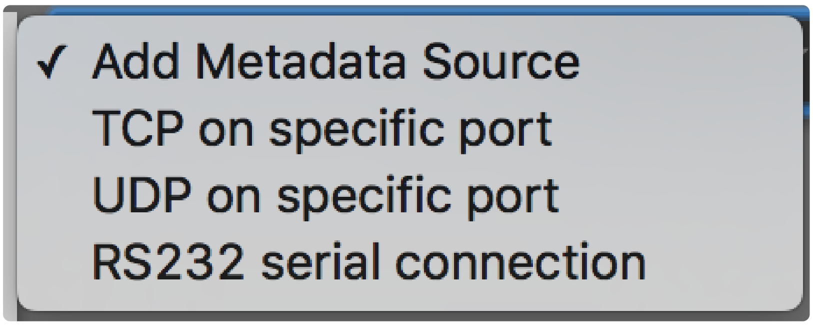
Metadata Sources



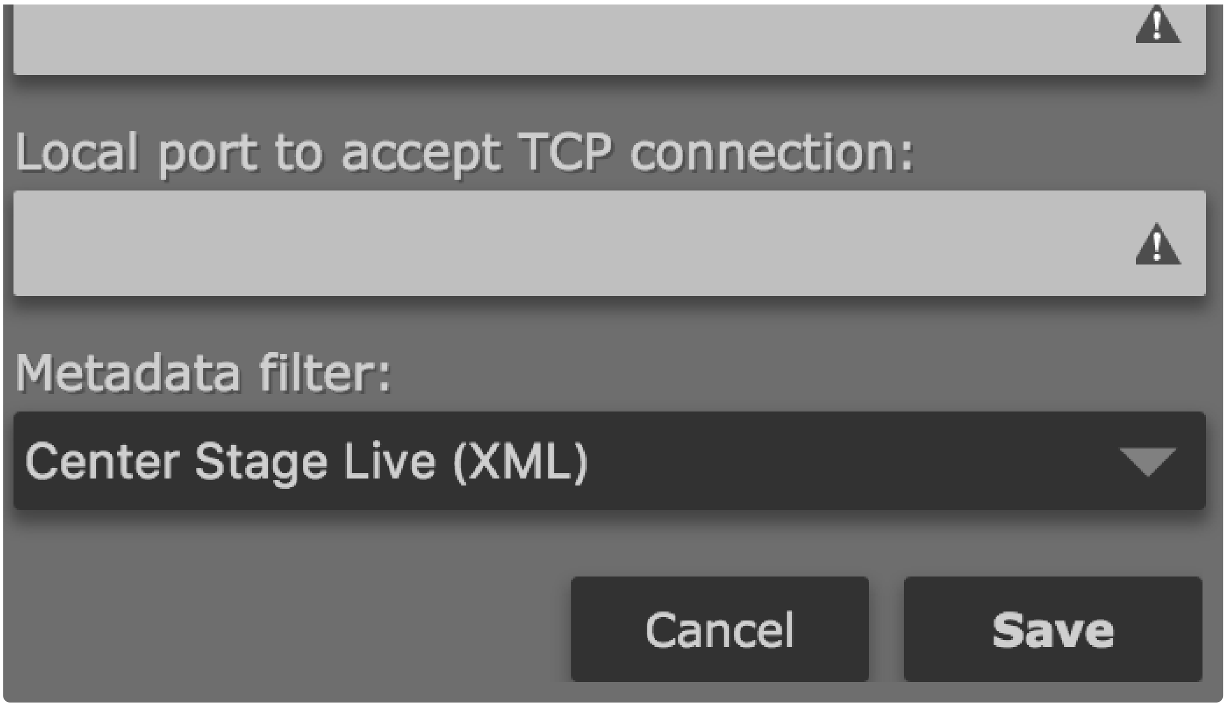
From the main “Control Panel” window, click on “Metadata”. The Metadata window allows you to view currently defined metadata sources, define new ones, and manage metadata “filters”.

After you define metadata sources, they will appear in the metadata dropdown list in the encoder configuration dialog. Make sure to create the metadata source first, or go back and edit any of your encoders to add metadata support to the streams.

To add a new metadata source, click on “Add Metadata Source”. Metadata via TCP, or UDP is supported. RS-232 serial is not currently implemented.



TCP on specific port



This selection accepts metadata via a specified TCP port and processes it according to the selected Metadata filter.

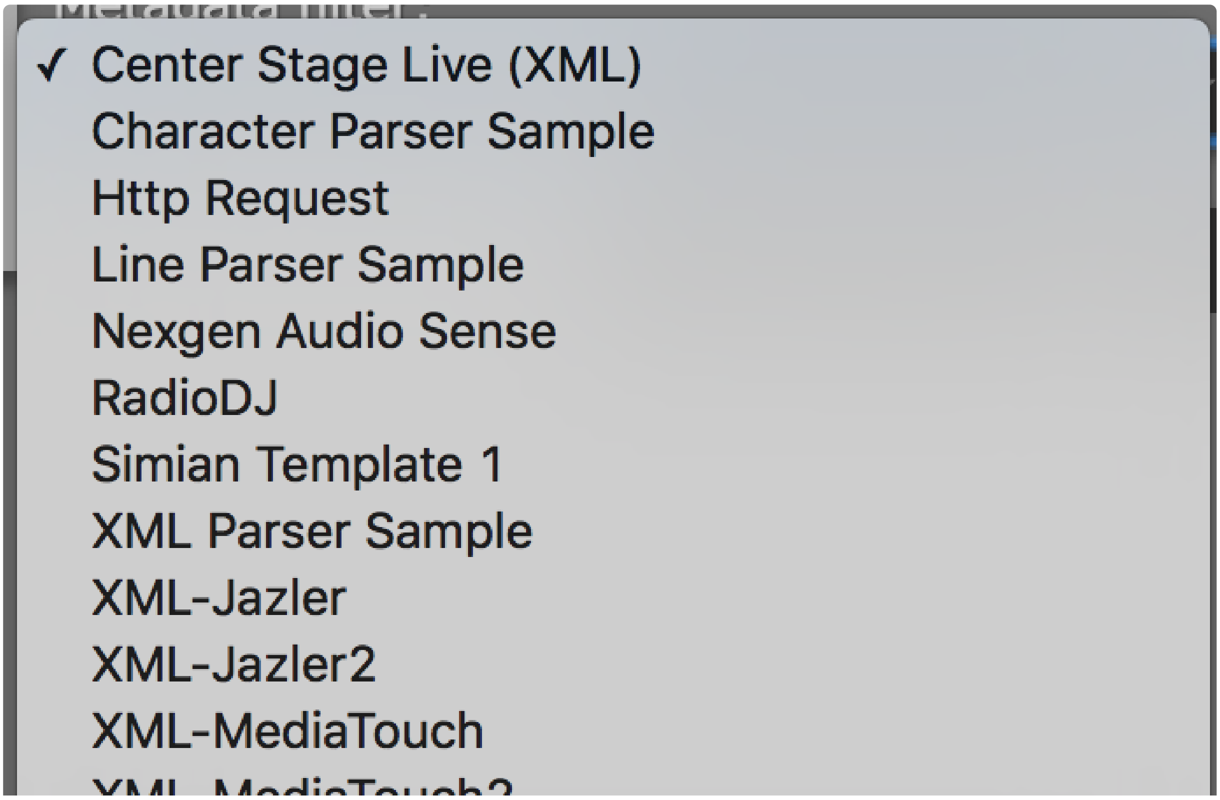
Friendly name

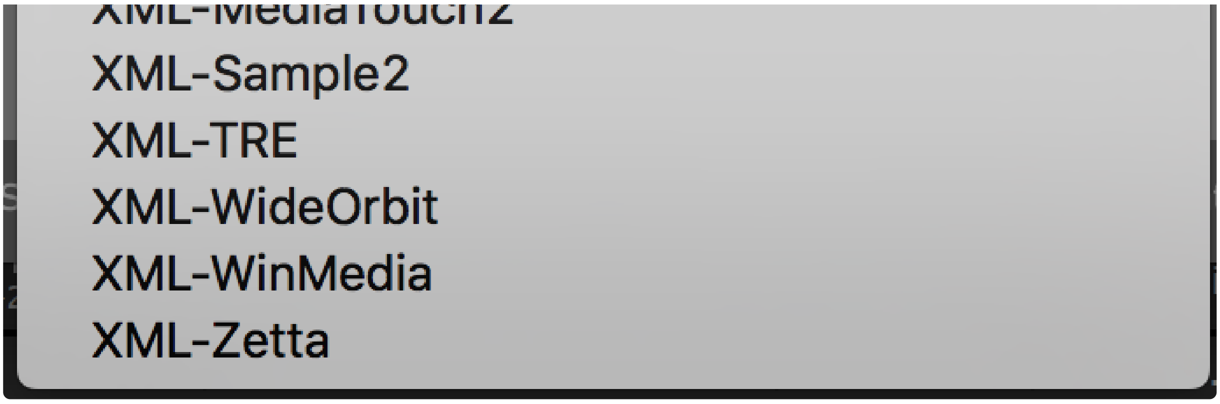
Enter an easily identifiable “friendly” name for this metadata source (i.e. Kxxx Automation Metadata).

Local port to accept TCP connection

Enter the TCP port number where your automation or metadata management system will connect to send metadata to R/2.

Metadata filter

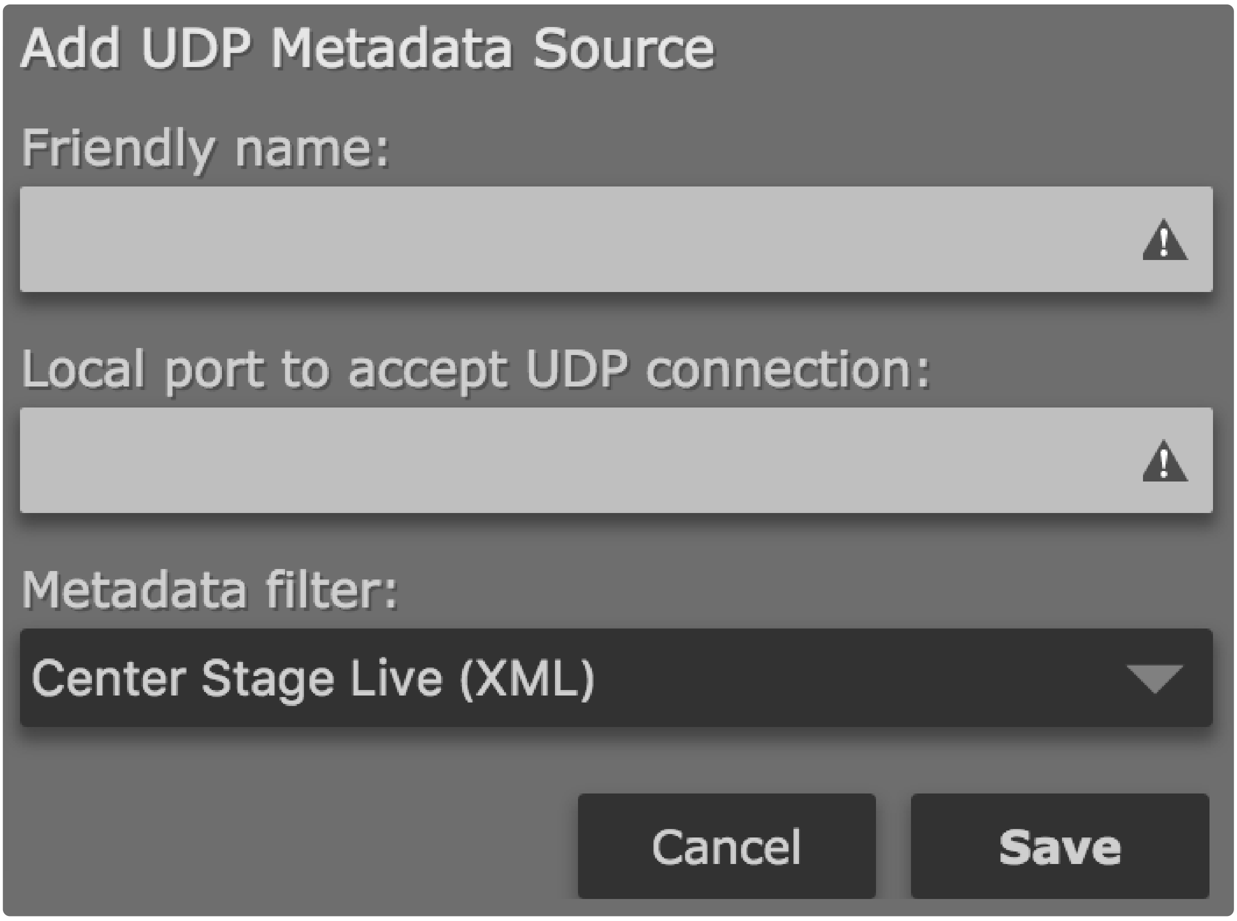




Select from the list of available “metadata filters”. The factory supplied metadata filters should cover a number of common applications. Contact technical support for additional information on creating or modifying metadata filters.

Click Save to save this entry, or Cancel to return to the Metadata screen.

UDP on specific port



This selection accepts metadata via a specified UDP port and processes it according to the selected Metadata filter.

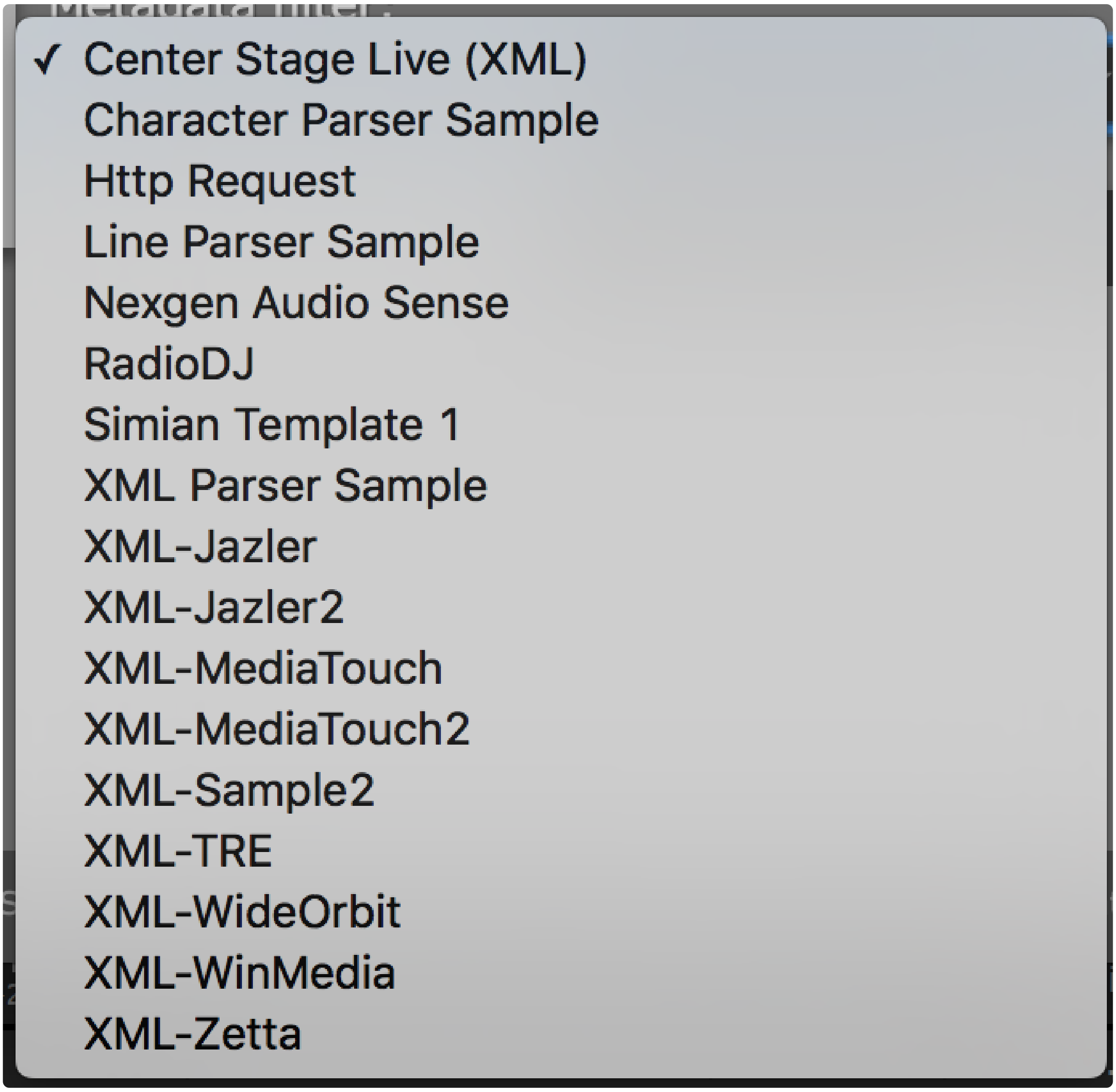
Friendly name

Enter an easily identifiable “friendly” name for this metadata source (i.e. Kxxx Automation Metadata).

Local port to accept UDP connection

Enter the UDP port number where your automation or metadata management system will connect to send metadata to R/2.

Metadata filter



Select from the list of available “metadata filters”. The factory supplied metadata filters should cover a number of common applications. Contact technical support for additional information on creating or modifying metadata filters.

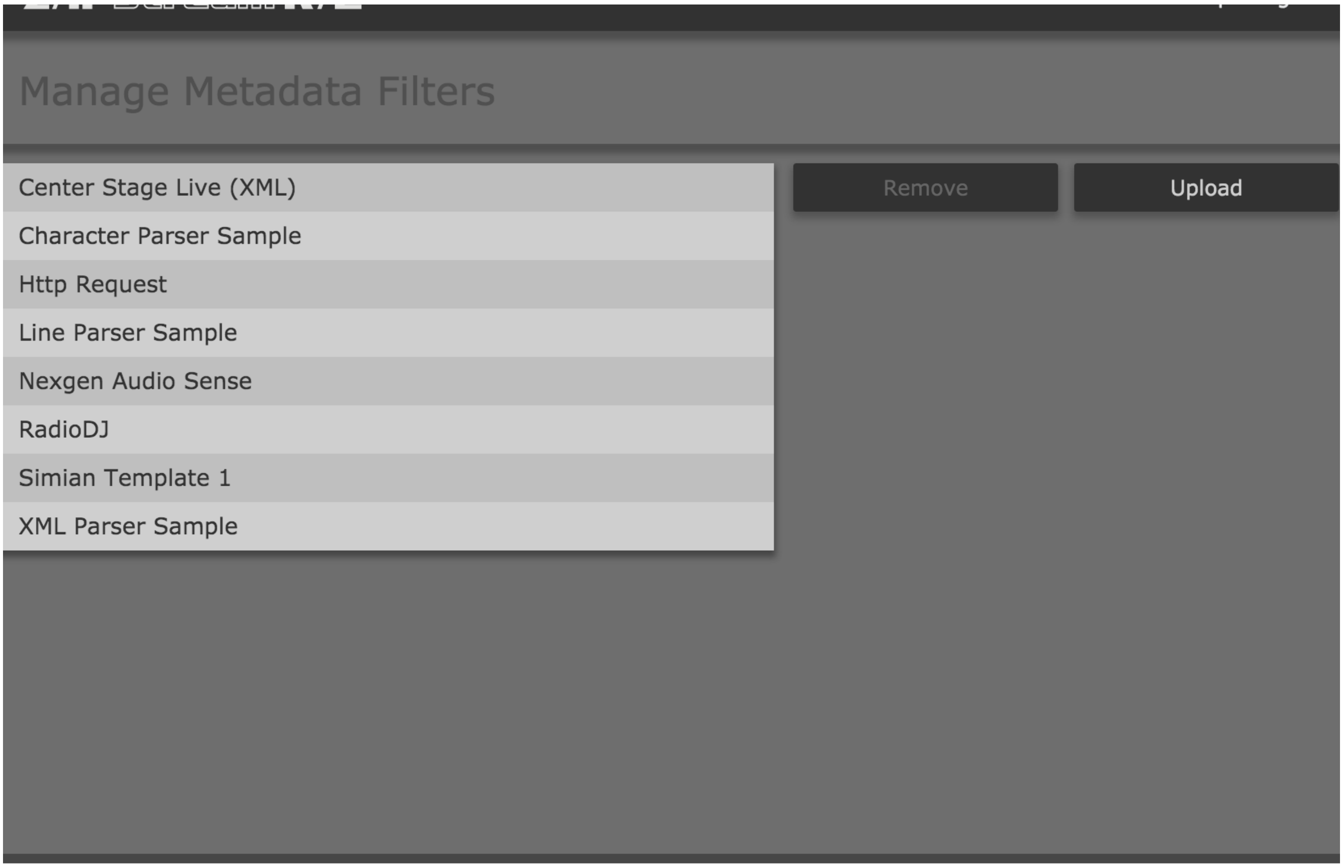
Click Save to save this entry, or Cancel to return to the Metadata screen.

Add Serial Metadata source

RS-232 serial metadata is not currently implemented in R/2.

Click Save to save this entry, or Cancel to return to the Metadata screen.

Manage Filters



R/2 is extremely flexible and can accept stream metadata from a number of different sources using metadata “filters”. These filters define the format of the incoming data and reformat it accordingly for stream now playing information. The “Manage Filters” screen allows you to delete existing filters, or upload new metadata filters to R/2.

> **TECH NOTE:** Metadata “filters” are small mini-programs that translate the data received from an external source to tags that the stream sender can understand. Each filter is a separate file stored on R/2. These mini-programs are written using the Lua programming language. Technical Support can assist with developing custom metadata filters. You can find out more about the Lua programming language at http://www.lua.org.

Event Scheduler

Overview

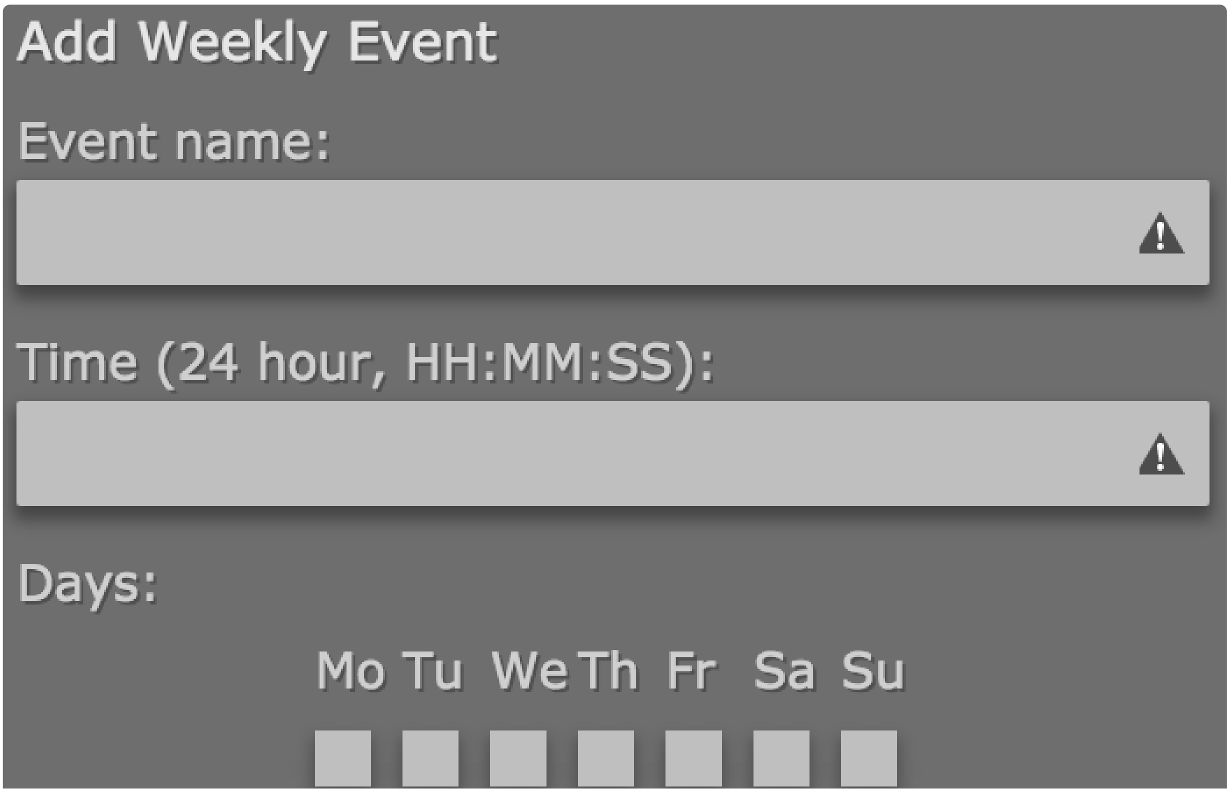
The scheduler is able to start, stop audio processor instances or encoders and change an instance’s processing preset at specified times. This is useful if you want to encode and stream only certain shows. If you need to process one show differently than another, you can use the scheduler to change audio processing presets.

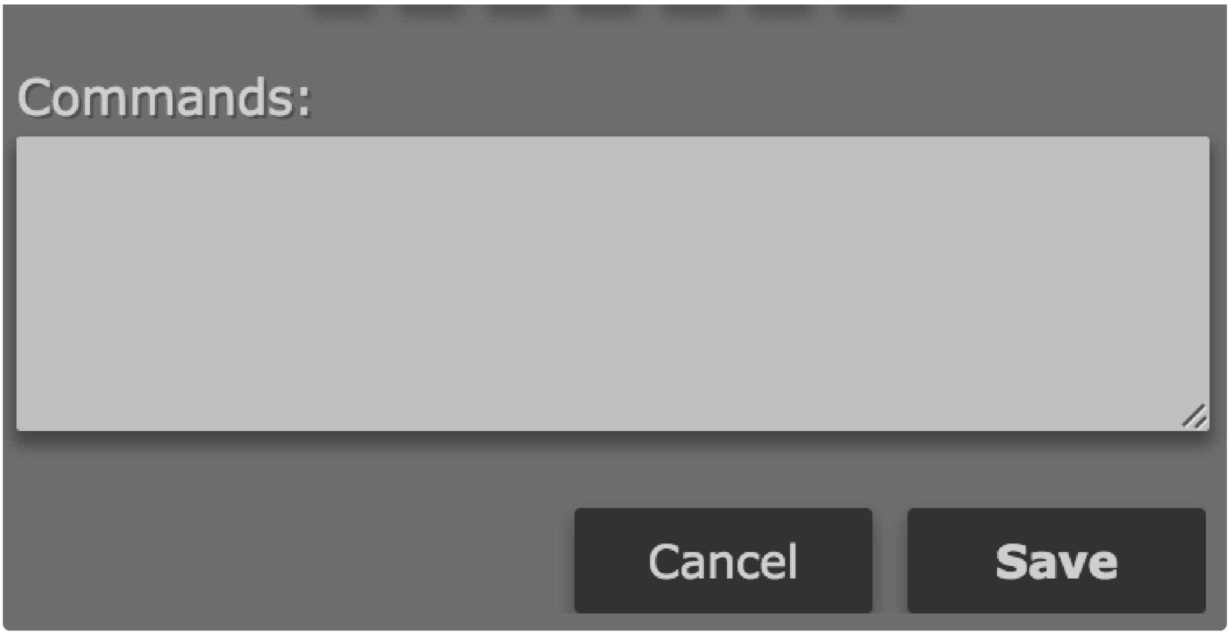
Schedule Manager

Click “Scheduler” from the main “Control Panel” screen to access the Schedule Manager.

The Schedule Events list displays all the events you have created so far. When you select an event from the list, the event commands are displayed in the Commands area. Click the “Add...” button to create a new event.

Add Weekly Event





Event Name

Enter a name for the event. Keep the names descriptive but use only letters, digits, underscore and space characters. Do not use other special characters.

Time (24 hour, HH:MM:SS)

Enter the time for the event (in 24-hour, HH:MM:SS format)

Days

Click the check boxes for the days of the week when the event should be active.

Commands

The scheduler works by executing commands you enter in the Commands text box. As of program version 1.01.29, R/2 recognizes the following commands:

InstanceStart("instance_name")

Use this command to start the instance named instance_name.

InstanceStop("instance_name")

This command will stop the instance named instance_name.

InstanceLoadPreset("instance_name","preset_name")

Use this command to load the preset named preset_name into the instance named instance_name.

You may add multiple commands by entering each command on a separate line. For example, you could start an instance then load a certain preset for it. Or you could start or stop multiple instances at the same time.

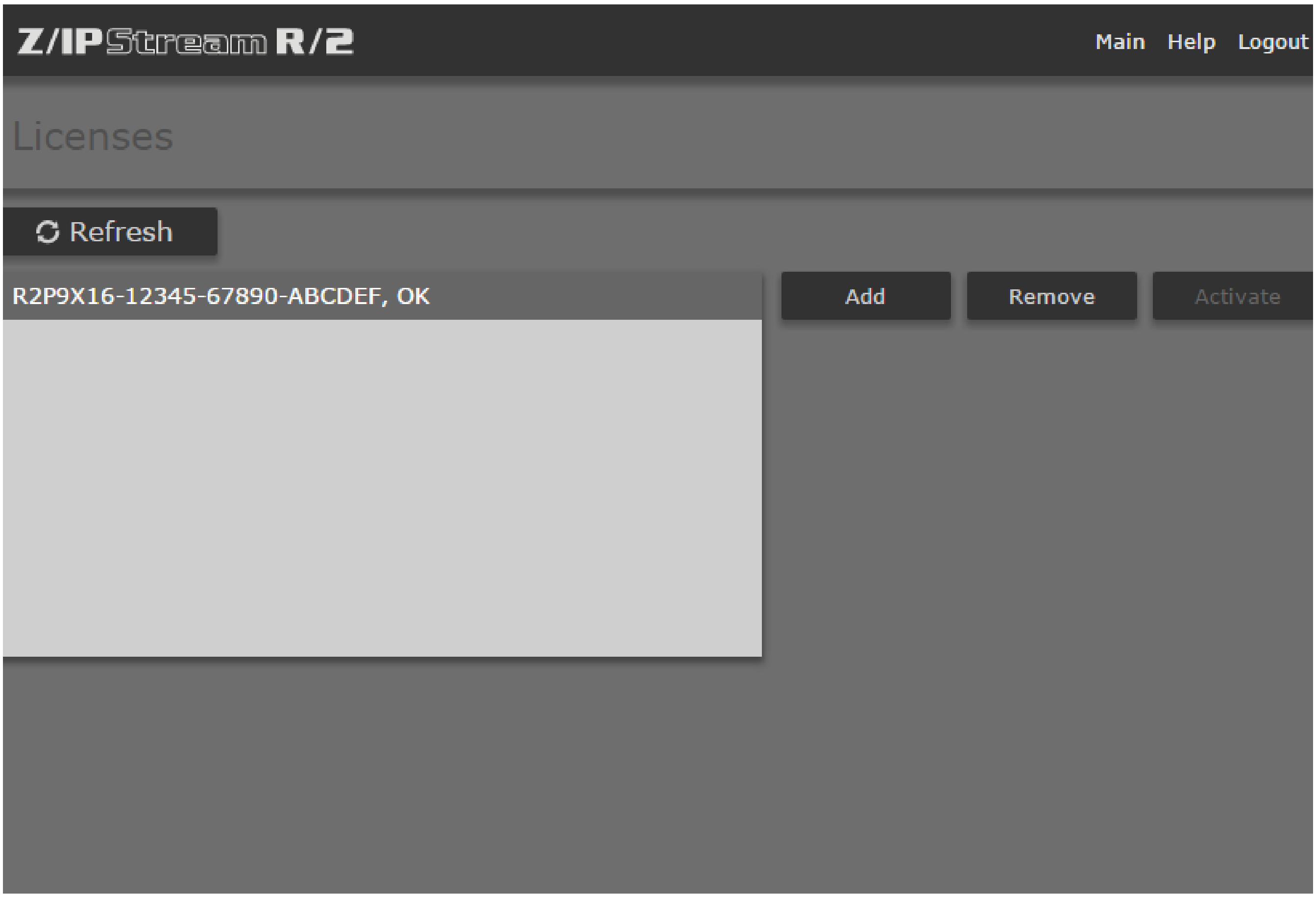
Click "Cancel" to cancel and return to the scheduler window, or click "Save" to save changes and return. The newly added event should appear in the list.

License Management

Overview

The licenses determine the number of audio inputs that can be created in R/2, and the type of audio processing available. R/2 should come licensed from the factory for the configuration that was ordered, but in some cases it may be necessary to enter additional license keys. Examples would be adding additional audio channels, or enabling Omnia.9 processing.

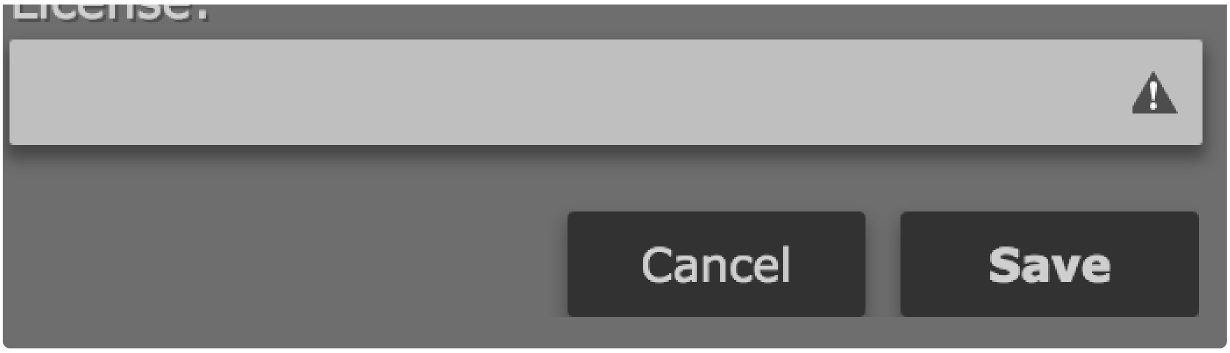
Licenses



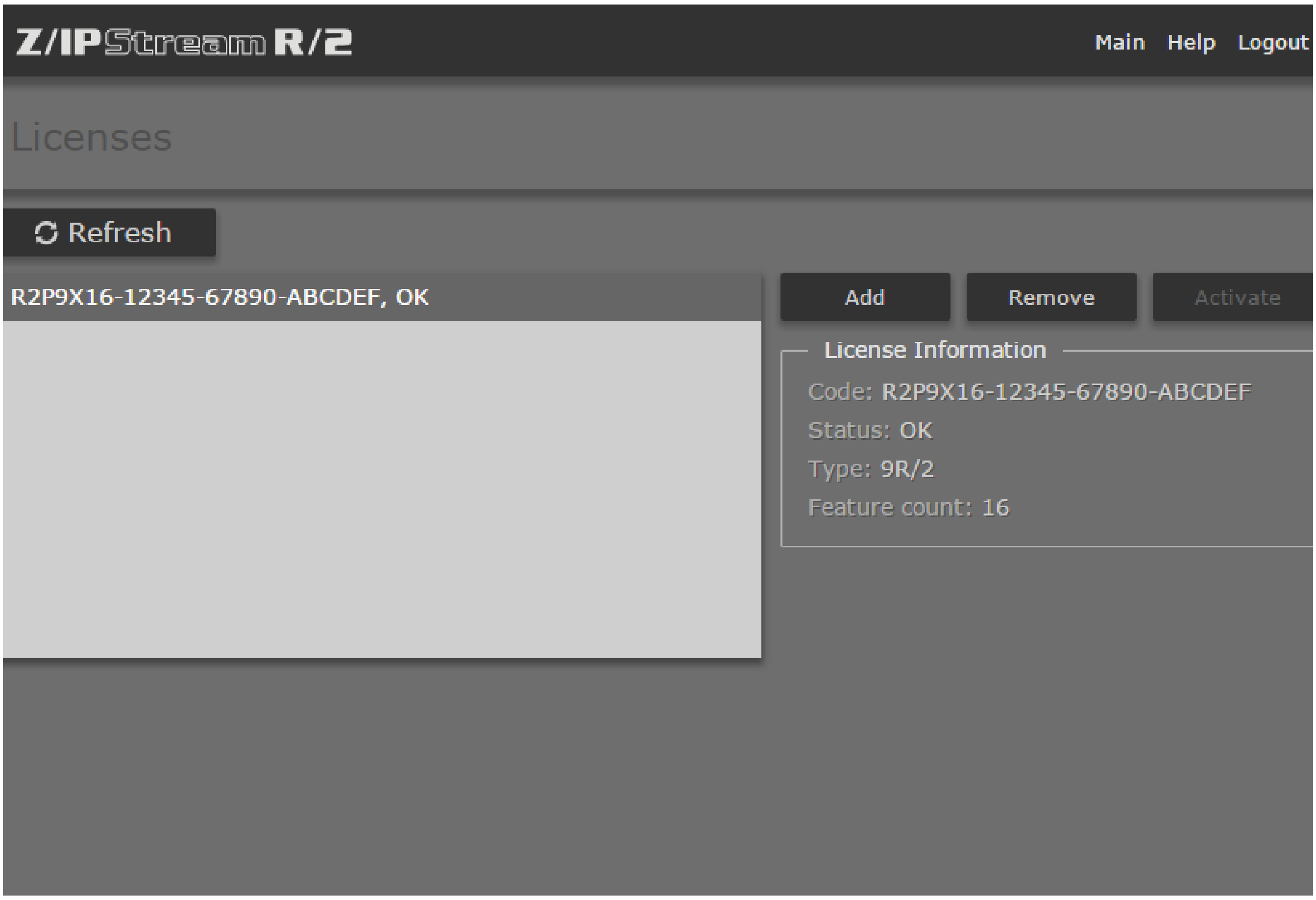
Displays the current license key(s) on the system. Click "Add" to add a new license key.

NOTE: Licenses are verified with a Telos licensing server. Make sure R/2 is connected to the Internet when you enter the license codes.

Add…



Enter the license key. Click “Save” to save or “Cancel” to return to the Licenses screen without adding the license key.



The code you entered will appear in the Licenses list on the left. Next to the license code, you will see additional text describing the state of the license. The text will initially say “Pending activation”. After 10-15 seconds it should change to “OK”. To reload the license status, use the “Refresh” button. Add each license code you have received individually. If all license codes have an “OK” status, then they are active and the additional features (audio channels or Omnia.9 processing) may be used. See Chapter 7 for configuring additional audio sources, Chapter 8 for configuring processing/encoding instances (including Omnia.9 if licensed) and Chapter 16 for configuring Omnia.9 processing.

If you see an error message next to the license code, please check the license string to make sure it matches the one you received. If the error is “Activation failed”, or similar, then contact customer support at +1 (216) 622-0247 or support@telosalliance.com to learn how to activate your license manually.

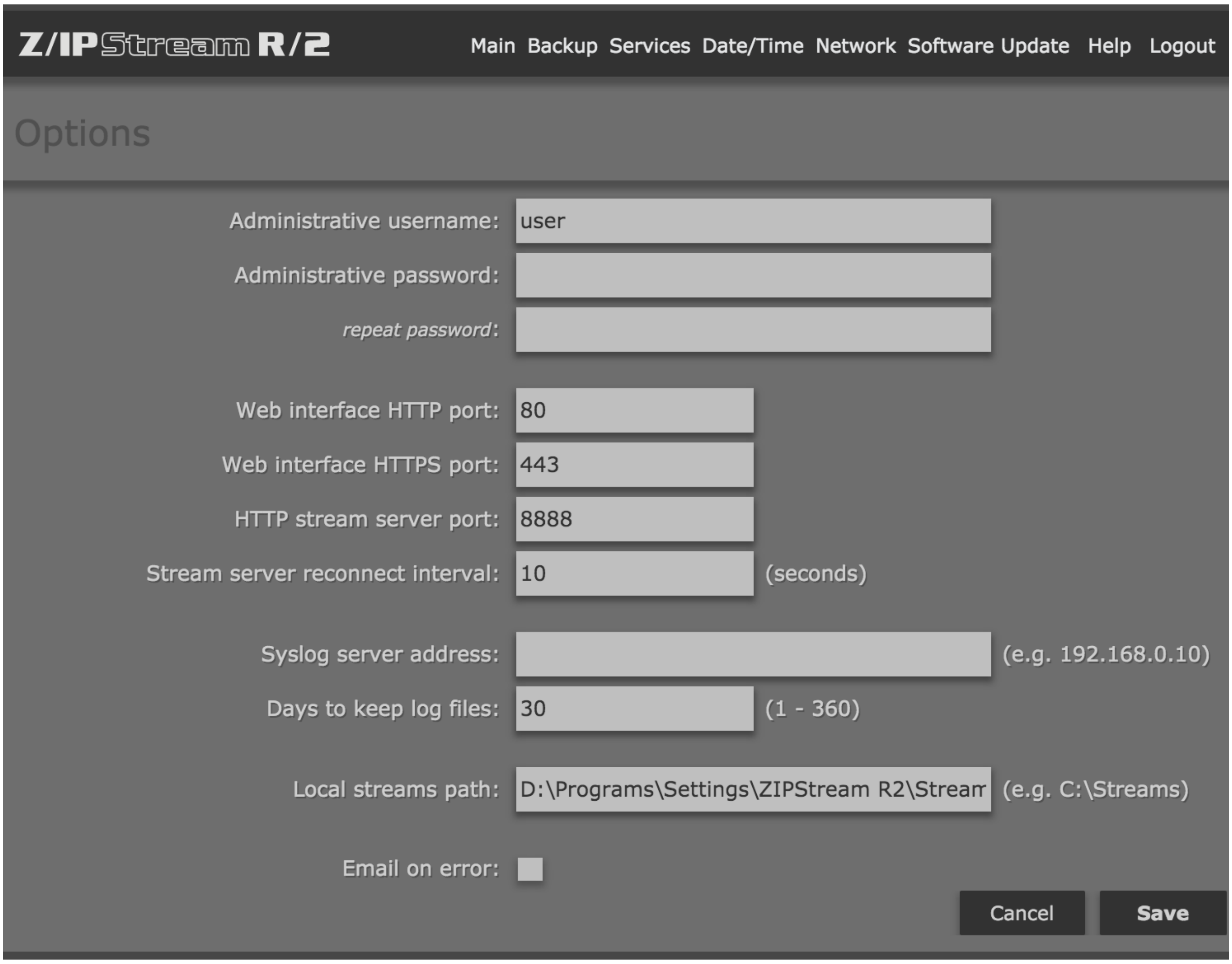
System Options

Overview

This chapter will cover global configuration options including:

- Setting administrative username and password
- Modifying web interface and stream server ports
- Configuring logging
- E-Mail alerts for error conditions
- Backup and restore of system configuration
- Management of services
- Date and Time/NTP
- Network interface configuration
- Software update

Options



Administrative username

The default administrative login is “user”. Although only one administrative login username is supported, this username can be changed if desired.

Administrative password

By default, this password is blank. It is highly recommended that this password be changed, particularly if the R/2 web interface will be accessible from the public Internet. Enter the password again in the “repeat password” field.

Web interface HTTP port

Modifies the port for the internal web server for security, port conflicts, or other reasons. Port 80 is the default, as with any other standard HTTP server.

Web interface HTTPS port

Modifies the port for the internal HTTPS (secure) web server for security, port conflicts, or other reasons. Port 443 is the default, as with any other standard HTTPS server.

HTTP stream server port

Modifies the port for the internal “test” HTTP stream server. The default is port 8888.

Note: Normally the default values for these ports are sufficient. Only change them if you have a specific reason to do so. Changes to port values will take effect after restarting R/2 or the R/2 service.

Stream server reconnect interval

Sets the interval for reconnect attempts to the stream server. The default value of 10 seconds should be sufficient in most cases.

Syslog server address

Syslog is a protocol designed for central logging of various events occurring on equipment within a network. If you wish to log R/2 events to a syslog server, enter the IP address for that server here.

Days to keep log files

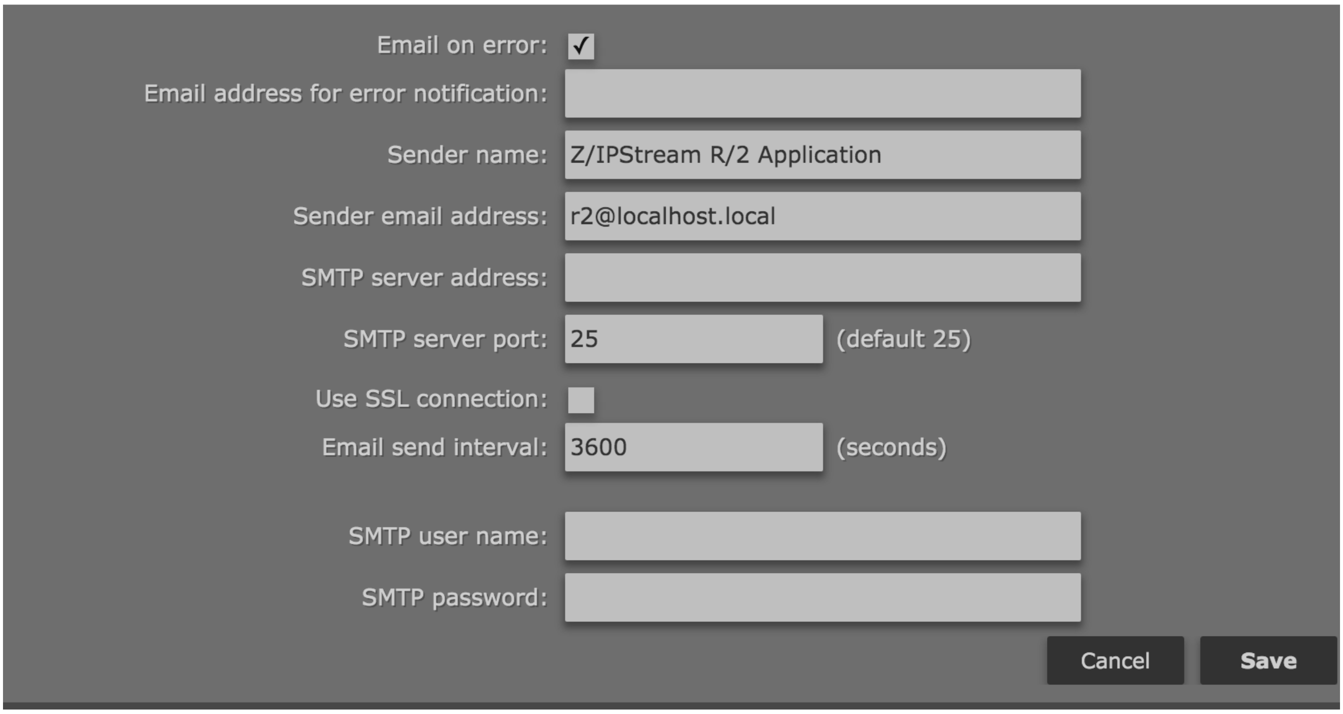
Specify the length of time that logs should be kept on R/2, up to 360 days. The default is 30 days.

Local streams path

Specifies where HLS and other streaming files should be generated when stored locally. There is no local access to the R/2 file system, and generally this value should not be changed from the default. See the HLS streaming configuration examples from Chapter 8 for more details.

Email on error

When this box is checked, it enables additional settings for configuring e-mail alerts in the event of an error condition.



Email address for error notification

Enter an email address in this field. While emails may be sent directly to e-mail recipients, they can also be sent as text messages to mobile phones. If you are not familiar with how to do this, see the tutorial at http://sms411.net/how-to-send-email-to-a-phone/.

Sender name

The default is 'Z/IPStream R/2 application'. It will be useful to change this, if you have more than one R/2 running in your facility).

Sender email address

Enter the account username.

SMTP server address

You may enter either an IP address in the ###.###.###.### numeric format, or a name address (e.g. smtp.gmail.com). In order to resolve the name address, a working DNS server must be already configured on R/2 (see network configuration details later in this chapter).

SMTP server port

The default is 25, but you may change it if necessary. Many providers are now blocking SMTP on port 25. An alternative is port 587, which is also supported by most SMTP servers for non-secure SMTP traffic. Ideally, you will want to point to an SMTP server on your own network (or your ISP’s network) to prevent issues with relaying.

Use SSL connection

Checking this box will ensure that your SMTP communications are encrypted and secure from hackers but it will work only if your email service provider can accept SSL connections. A number of providers have switched from open SMTP servers (port 25) to SSL and authentication in order to cut down on SPAM.

Email send interval

Determines how frequently Z/IPStream sends you email alerts about a problem. The default is 3600 seconds (one hour). We suggest not setting it lower. You want to have a value large enough to give you enough time to resolve the situation before the problem repeats. Otherwise, in case of a persistent problem, you may end up having your email flooded with too many messages.

SMTP username

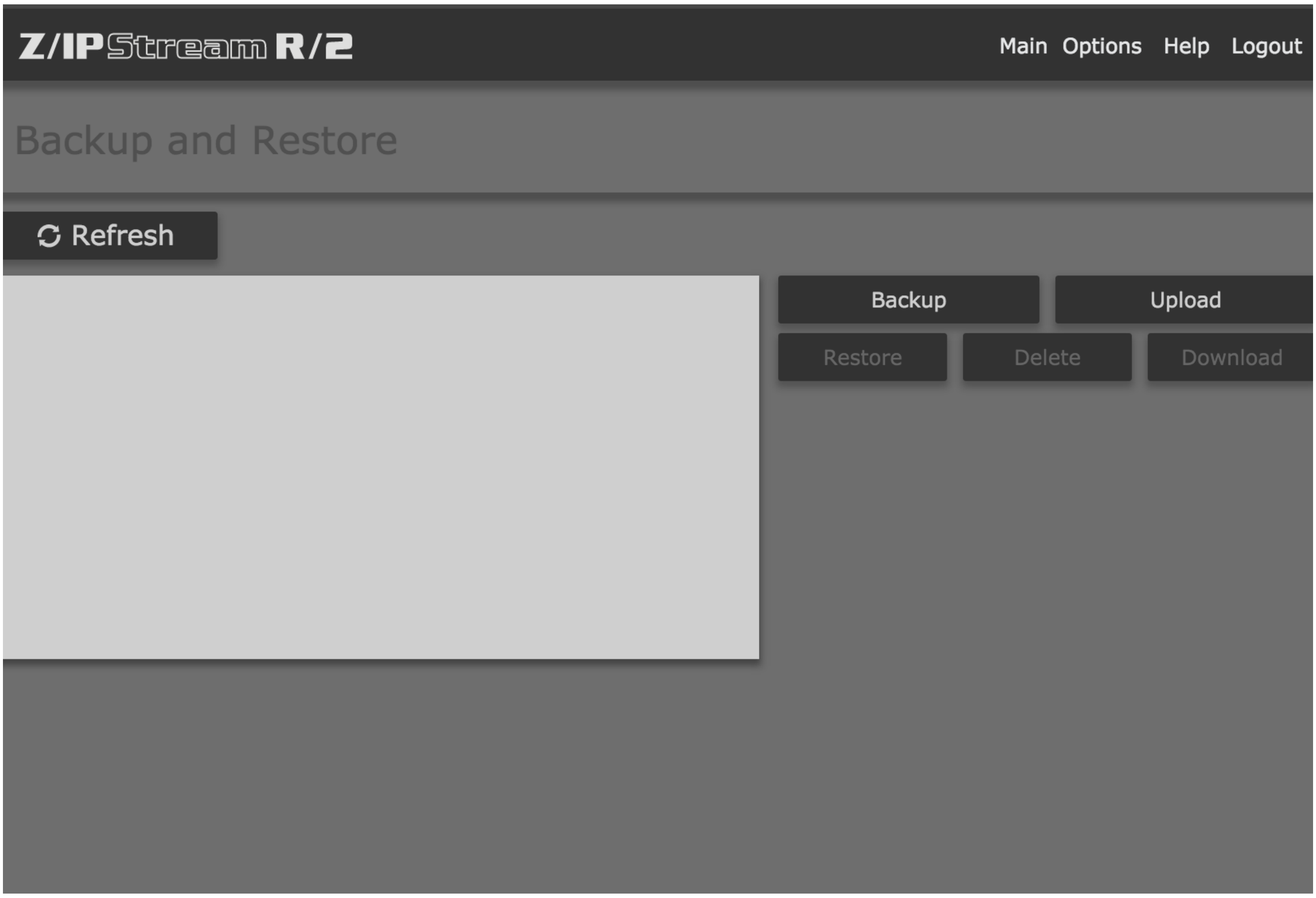
Enter the user name for your SMTP server here (if applicable).

SMTP password

Enter the password for your SMTP server here (if applicable).

Click the Save button to make the changes permanent. You will then be returned to the Control Panel page.

Backup and Restore



From the “Options” screen, click on “Backup” to access the Backup and Restore screen. This allows you to perform and restore complete backups of the R/2 configuration (including presets).

Refresh

Refreshes the displayed list of backups currently on the system.

Backup

Performs a backup of the R/2 configuration.

Upload

Uploads a previously downloaded backup.

Restore

Restores the selected backup from the list.

Delete

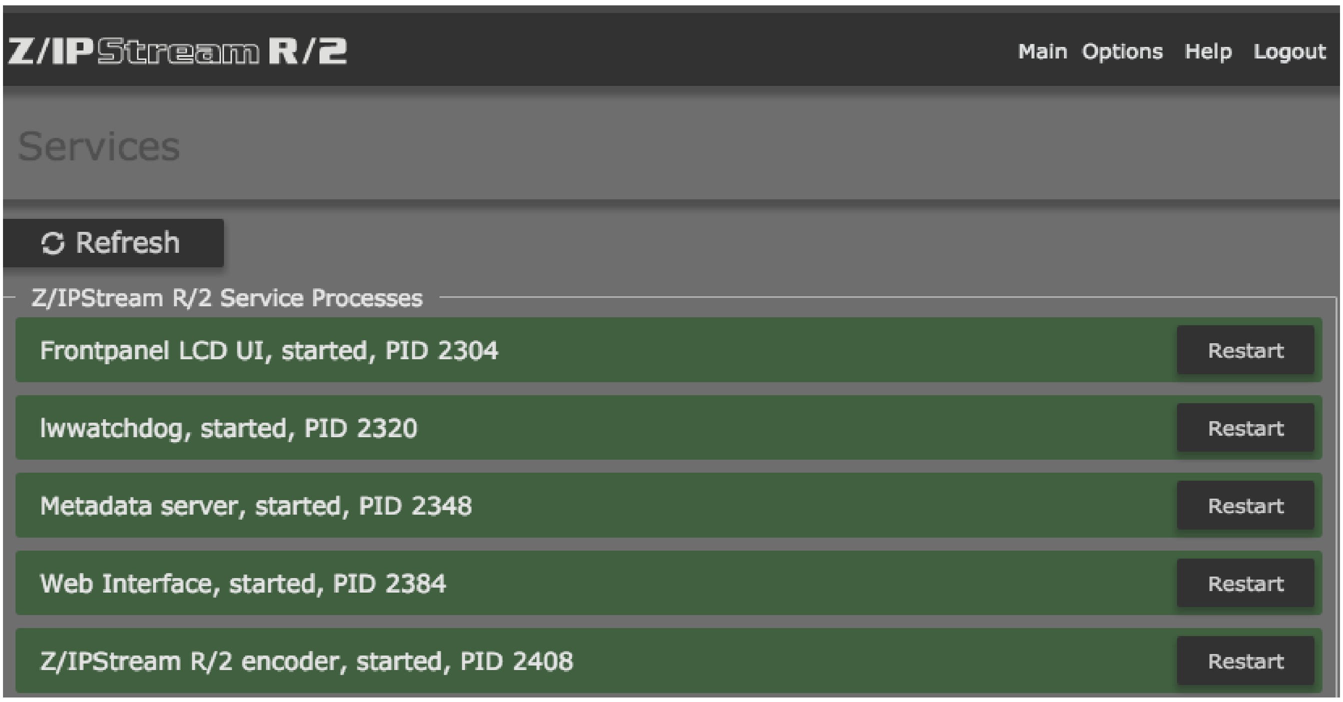
Deletes the selected backup from the list.

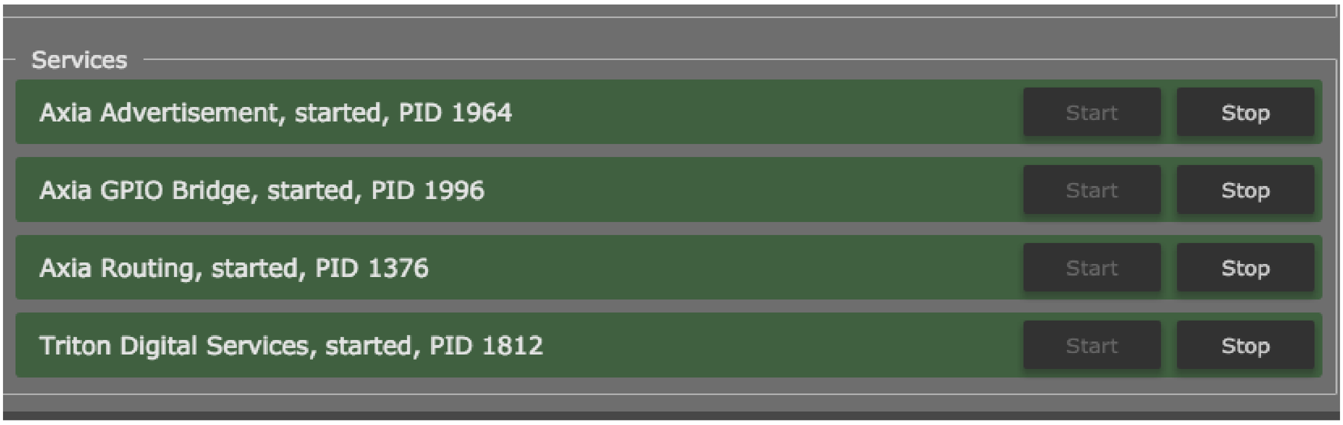
Download

Downloads the selected backup from the list.

Click “Options” to return to the system options screen.

Services





The various software components of R/2 run as “Services” on the system. While it should not be necessary under normal circumstances, this interface will allow each of the components to be restarted, stopped, or started as a troubleshooting measure. Note that stopping and restarting these services WILL temporarily disrupt operation of R/2 in various ways depending on which service is restarted.

Refresh

Updates the currently displayed status of these services. Green means go. A service will turn RED if it is not currently running. Contact support if a service is not running and will not restart.

Restart

Restarts the associated service.

Stop

Stops the associated service if started.

Start

Starts the associated service if stopped.

Service Descriptions

Frontpanel LCD UI

Service for the R/2 front panel interface. Restart in case of issues with the front panel.

Metadata server

Manages metadata services on R/2. Restart to troubleshoot metadata issues (i.e. if metadata stops working for some reason but there had been no other known changes).

Web Interface

Restarts the web interface service. This will log you out of R/2.

Z/IPStream R/2 encoder

Restarts the main R/2 encoder service. This is an alternative to completely rebooting R/2 and will restart all

R/2 functions.

Axia Advertisement/GPIO Bridge/Routing

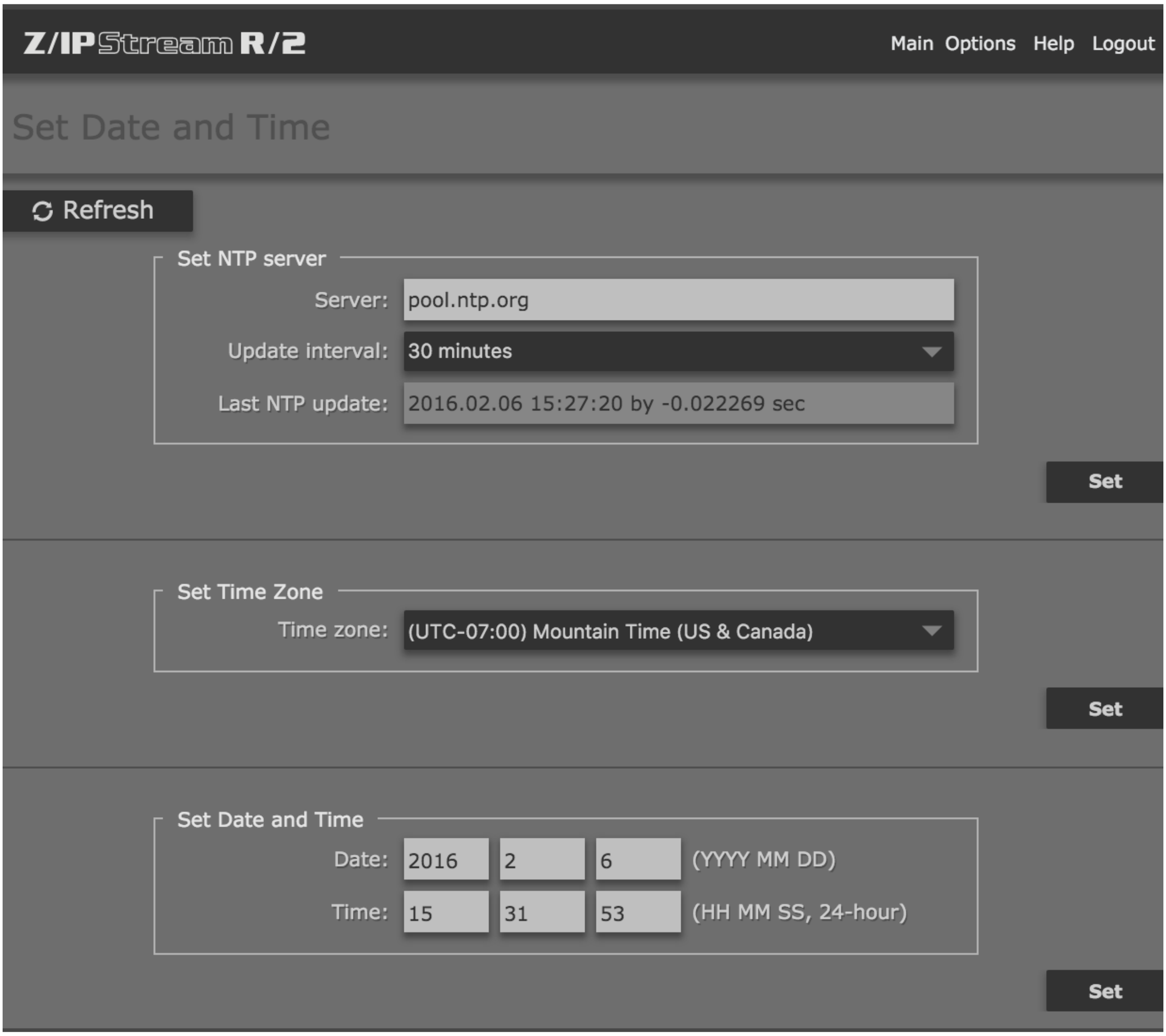
Livewire driver functions. GPIO Bridge and Routing are both dependent on Axia Advertisement. Restart these services in case of issues with the Livewire Driver.

Triton Digital Services

Controls the service that connects to Triton Digital Services.

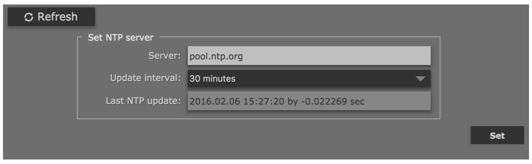
Click “Options” to return to the system options screen.

Date and Time



Provides various parameters related to setting the date and time on the unit. Typically, the system time would be set via NTP (Network Time Protocol). While time can also be set manually, it is critical for logging and other purposes that the R/2 have an accurate time reference.

Set NTP Server



Refresh

Refreshes all values displayed on the “Set Date and Time” screen

Server

Specify an NTP server for R/2 to query. The default is pool.ntp.org, a public NTP time server pool. If your organization uses an internal NTP server, specify that address here.

Update interval

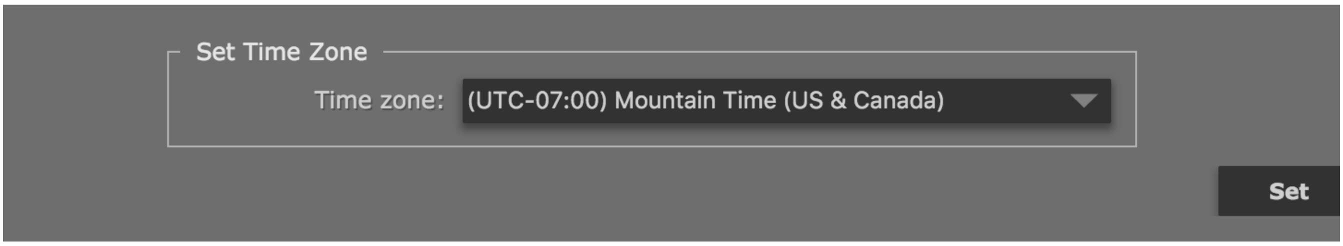
Specify the frequency of NTP queries. The default of 30 minutes is generally sufficient.

Last NTP update

Displays the last time the NTP server was successfully queried and the amount of correction. If everything is working properly, the correction value should be fairly small and the last update should be within the specified interval.

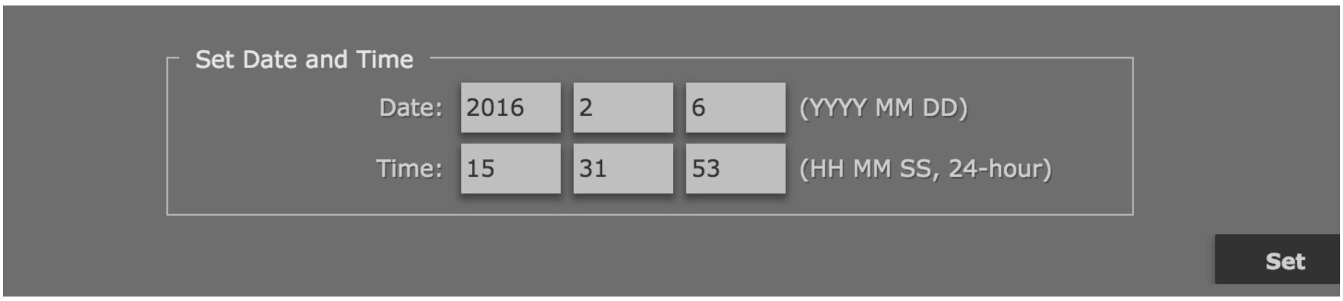
Click “Set” to confirm any changes.

Set Time Zone



Specify the local time zone to be used by R/2. Click “Set” to confirm.

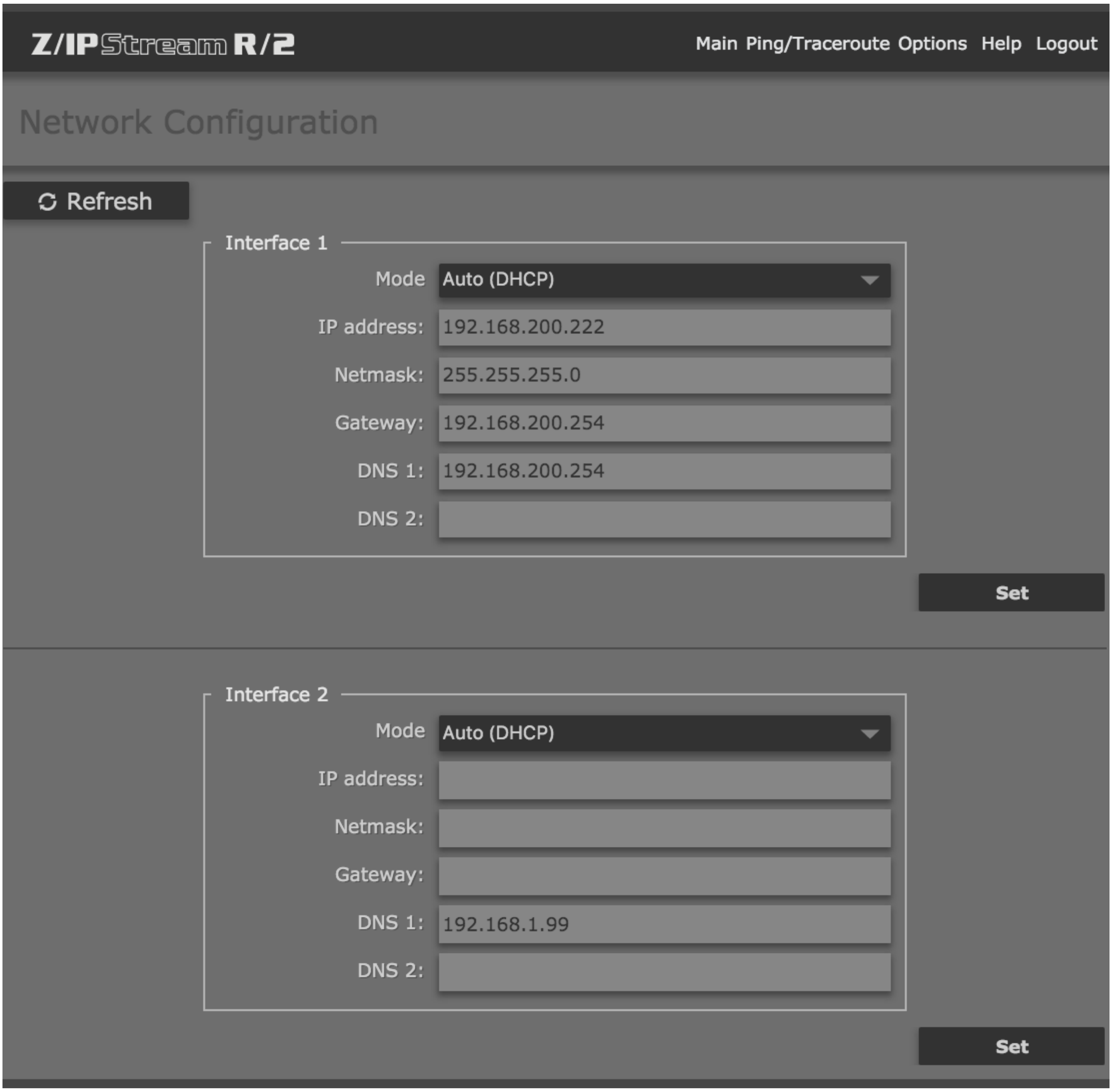
Set Date and Time



Allows the time and date on R/2 to be set manually. Click “Set” to confirm.

Click “Options” to return to the system options screen.

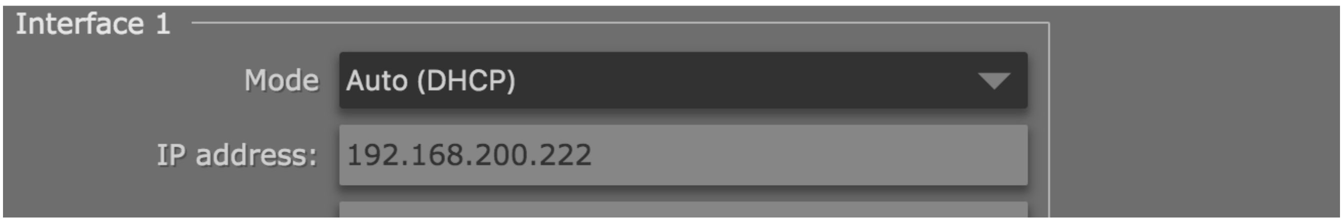
Network Configuration



While the initial configuration of IP addresses is done from the front panel of R/2, the Network Configuration screen allows reconfiguration of network interfaces, as well as specifying DNS servers for each interface.

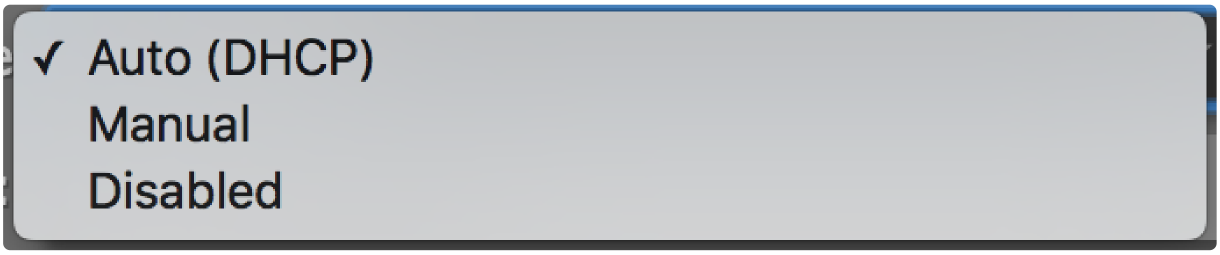
Note: Improper configuration of network parameters could disrupt connectivity to R/2. The front panel will always be capable of reconfiguring network interfaces, but in the case of remotely located units, it will require operator intervention at the front panel of the unit to restore connectivity.

Interface 1/Interface 2



Configures parameters for NIC 1 or NIC 2

Mode



Set to “Auto (DHCP)” to obtain an address via DHCP (Dynamic Host Configuration Protocol), “Manual” to specify IP address parameters manually, or “Disabled” to disable the interface entirely.

IP address

Displays current IP address if configured for “Auto (DHCP)”, or allows entry of a static IP address when configured for “Manual”.

Netmask

Displays current subnet mask if configured for “Auto (DHCP)”, or allows entry of a subnet mask directly when configured for “Manual”.

Gateway

Displays current network gateway (router) if configured for “Auto (DHCP)”, or allows entry of a gateway address directly when configured for “Manual”.

DNS 1

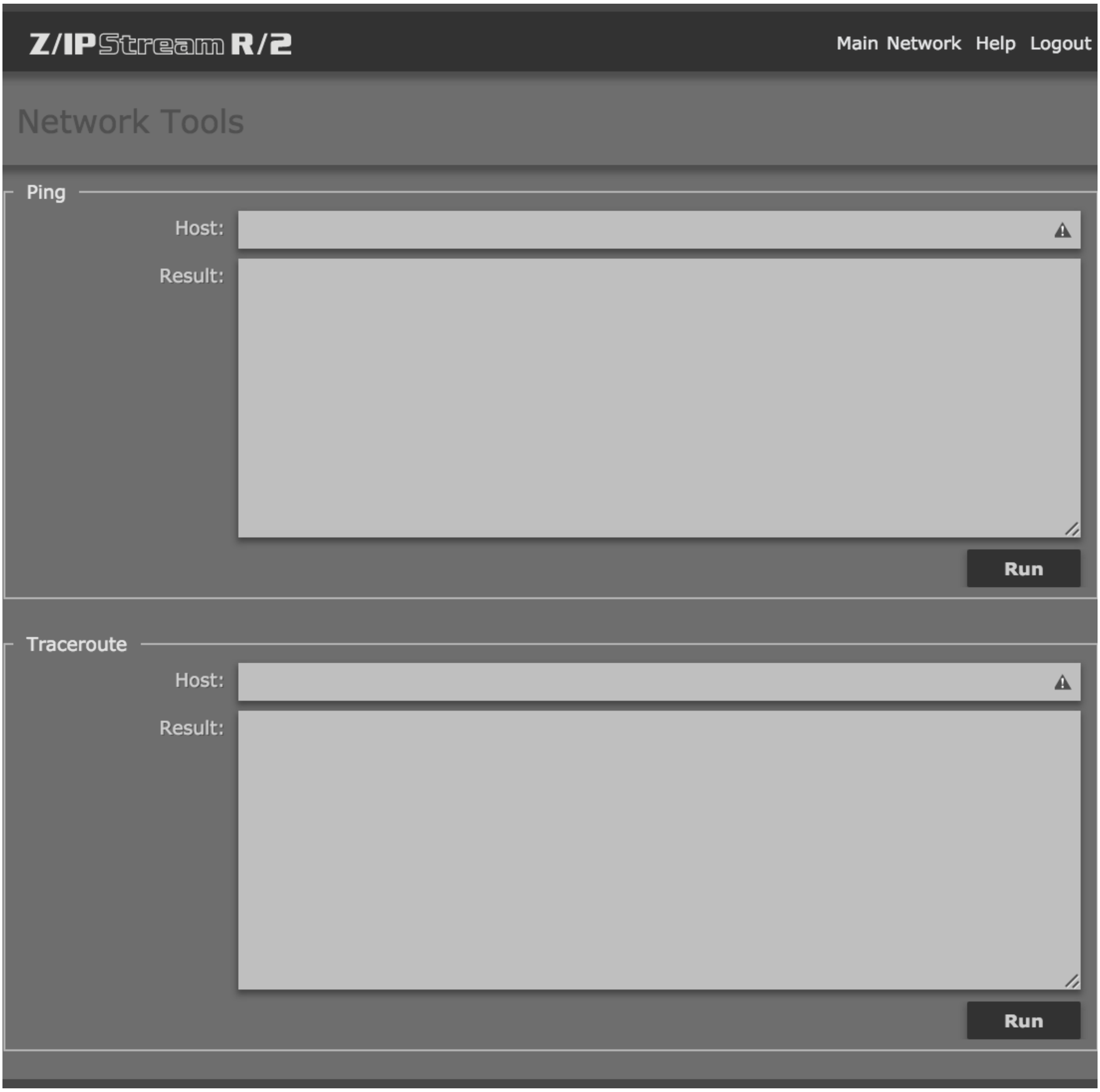
Displays current DNS (domain name) server if configured for “Auto (DHCP)”, or allows entry of a DNS server directly when configured for “Manual”. Note that this value cannot be entered via the front panel and must be entered through this configuration page.

DNS 2

Displays current secondary DNS (domain name) server if configured for “Auto (DHCP)”, or allows entry of a secondary DNS server directly when configured for “Manual”. Note that this value cannot be entered via the front panel and must be entered through this configuration page.

Click “Set” to confirm the values, then click “Options” to return to the main options page, or click “Ping/Traceroute” to access network diagnostic tools.

Network Tools

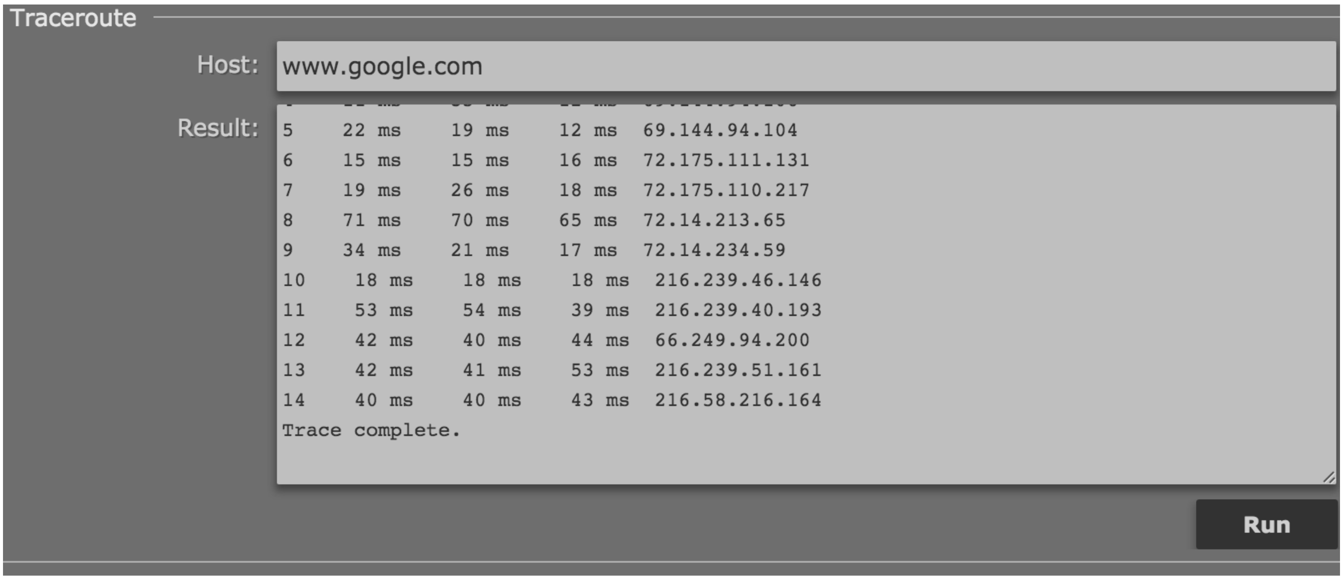


From the “Network Configuration” screen, click “Ping/Traceroute” to access the “Network Tools” interface. This screen provides ping and traceroute tools for network diagnostic purposes. These tools work like the equivalent tools on any other system…Ping checks the ability to access a specific host and displays the amount of time (latency) it takes to access that host, while traceroute traces the network path to a given host address across various routes. Both of these tools can be extremely valuable in troubleshooting network connectivity issues.

Ping

Enter the address of the desired host to ping, and click “Run”. The results will be displayed in the window.

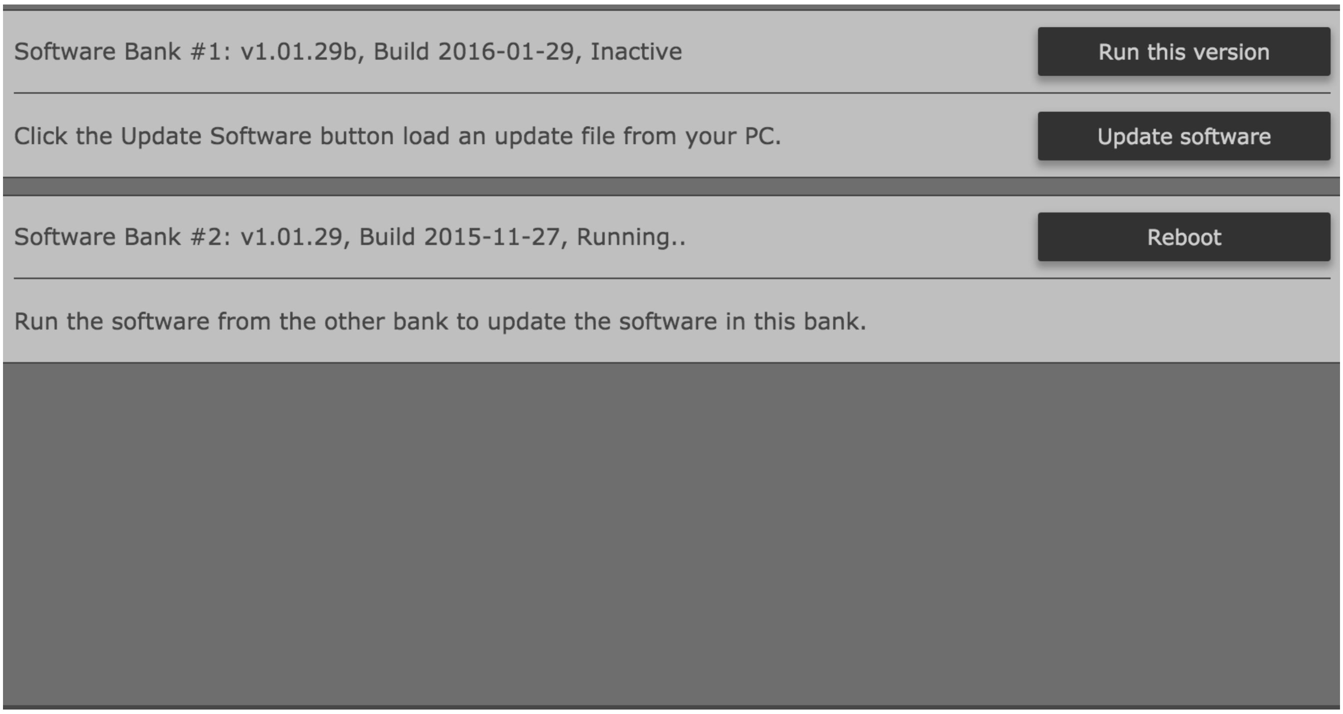
Traceroute



Enter the address of the desired host to traceroute, and click “Run”. The results will be displayed in the window.

Click “Options” to return to the main options screen.

Software Update



Provides a mechanism for updating software and rebooting the R/2. There are two software “banks” on R/2, only one of which is active at any given time, while the other is available to receive a software update.

Click “Update software” to upload new software to R/2, then click “Run this version” to run the newly updated version.

Click “Reboot” to reboot the R/2.

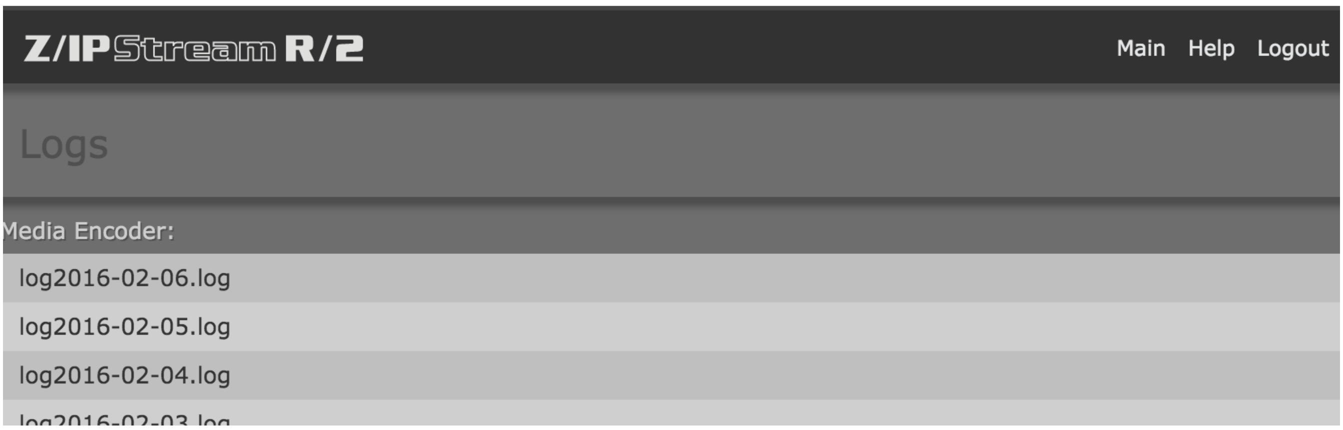
Click “Options” to return to the main options screen, or “Main” to return to the main control panel screen.

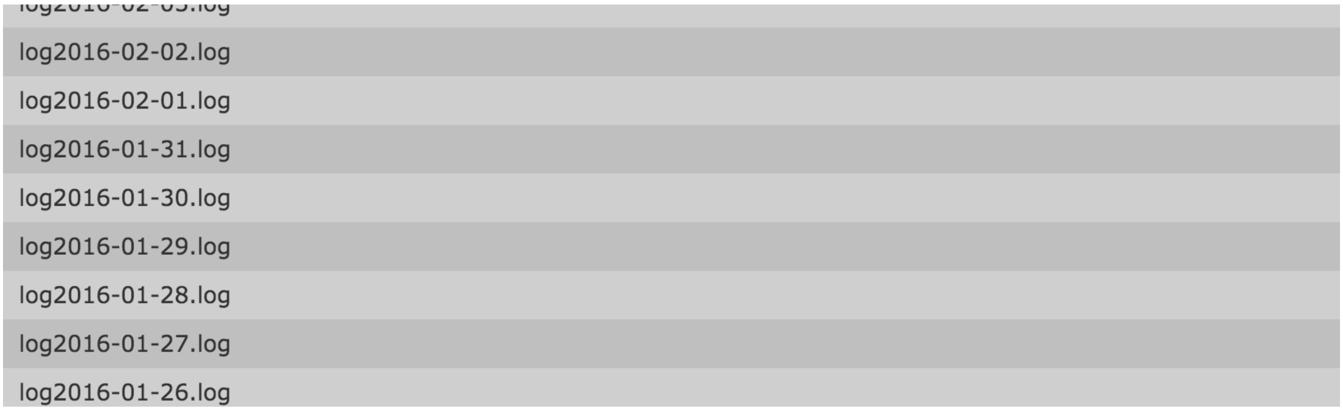
Logging

Overview

The R/2 contains a comprehensive logging system, designed to provide assistance in troubleshooting a variety of conditions. These logs can be extremely valuable both to the user, and to technical support.

Media Encoder Logs





_Screen%20Shots/15-01-Logs.png

Contains diagnostic logging information for the various media encoders, audio interfaces, and other routines such as NTP.

22:41:17.656: Info: NtpTimeSync: Clock adjusted by -0.027328 sec

22:41:22.510: Info: WaveIn: wave0: t=27703 data=4887552 rate=176426.812981 Bps, rate2=44115.944122

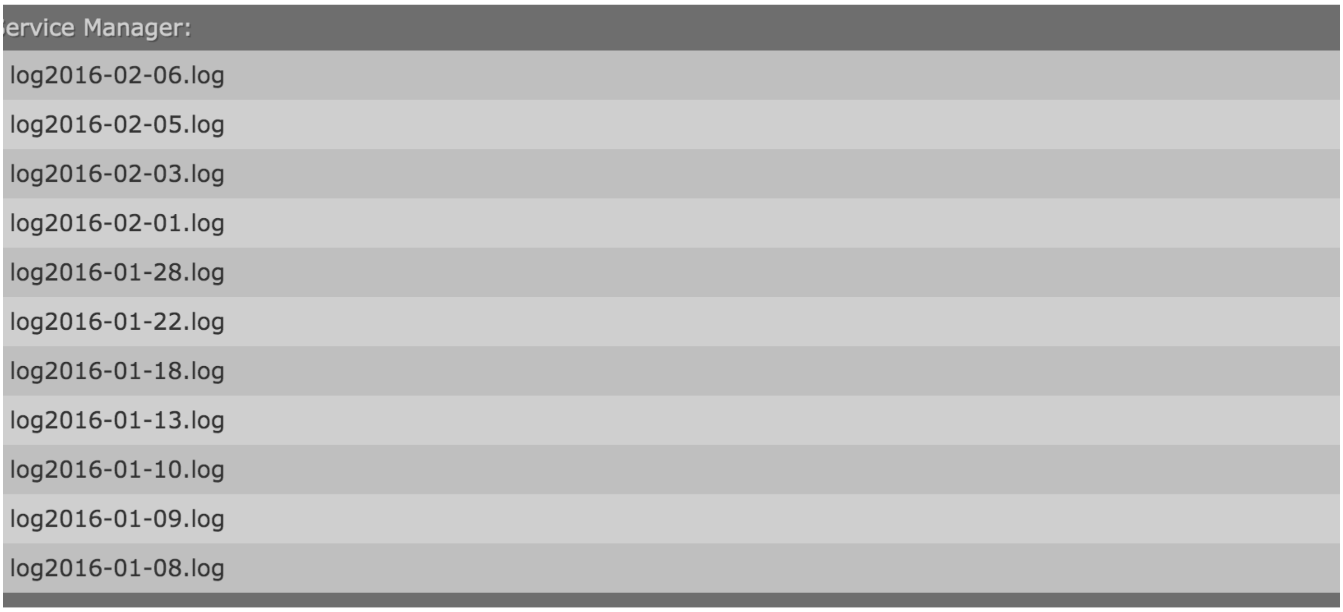
22:41:22.510: Info: WaveIn: wave7: t=54295579 data=9577504768 rate=176395.665069 Bps, rate2=44098.919988

22:42:19.389: Info: RestAPI: GET /v2/options.json

22:42:19.517: Info: RestAPI: GET /v2/network_adapters.json

22:44:22.515: Info: WaveIn: wave0: t=27703 data=4887552 rate=176426.812981 Bps, rate2=44115.944122

Service Manager Logs



Contains diagnostic information about the various services running in the background within R/2 (such as the web interface, etc.)

22:27:22.423: Info: ProcManager('WebIfc'): info('> GET /v2/network_adapters.json')

22:27:27.427: Info: ProcManager('WebIfc'): info('> GET /v2/ipv4.json')

22:27:33.433: Info: ProcManager('WebIfc'): info('> GET /v2/options.json')

22:27:33.433: Info: ProcManager('WebIfc'): info('> GET /v2/network_adapters.json')

22:29:19.535: Info: ProcManager('WebIfc'): info('> GET /v2/version.json')

22:42:20.251: Info: ProcManager('WebIfc'): info('> GET /v2/options.json')

22:42:20.251: Info: ProcManager('WebIfc'): info('> GET /v2/network_adapters.json')

22:56:21.055: Info: ProcManager('WebIfc'): info('> GET /v2/processes/proc1/logs.json')

22:56:22.056: Info: ProcManager('WebIfc'): info('> GET /v2/logs_common.json')

22:56:27.061: Info: ProcManager('WebIfc'): info('[L]> log2016-02-17.log - OK')

Click “Main” to return to the main control panel window.

Omnia.9 Processing Setup

Overview

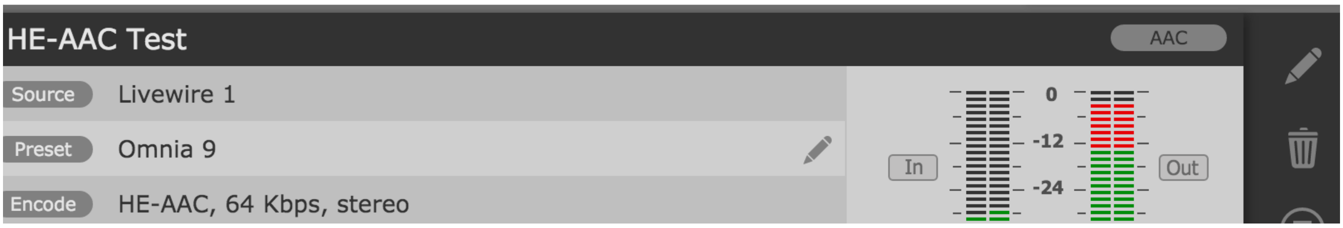
The Omnia.9 audio processing is an option available for Z/IPStream R/2. It features a wide array of unique tools including our exclusive “Undo” technology and 6-band parametric EQ.

Due to the wide range of parameters and processing options, Omnia.9 provides access to the settings through its own custom application, called NfRemote. As the name suggests, NfRemote can connect, monitor and manage the settings from another computer, either on the local network or over the Internet.

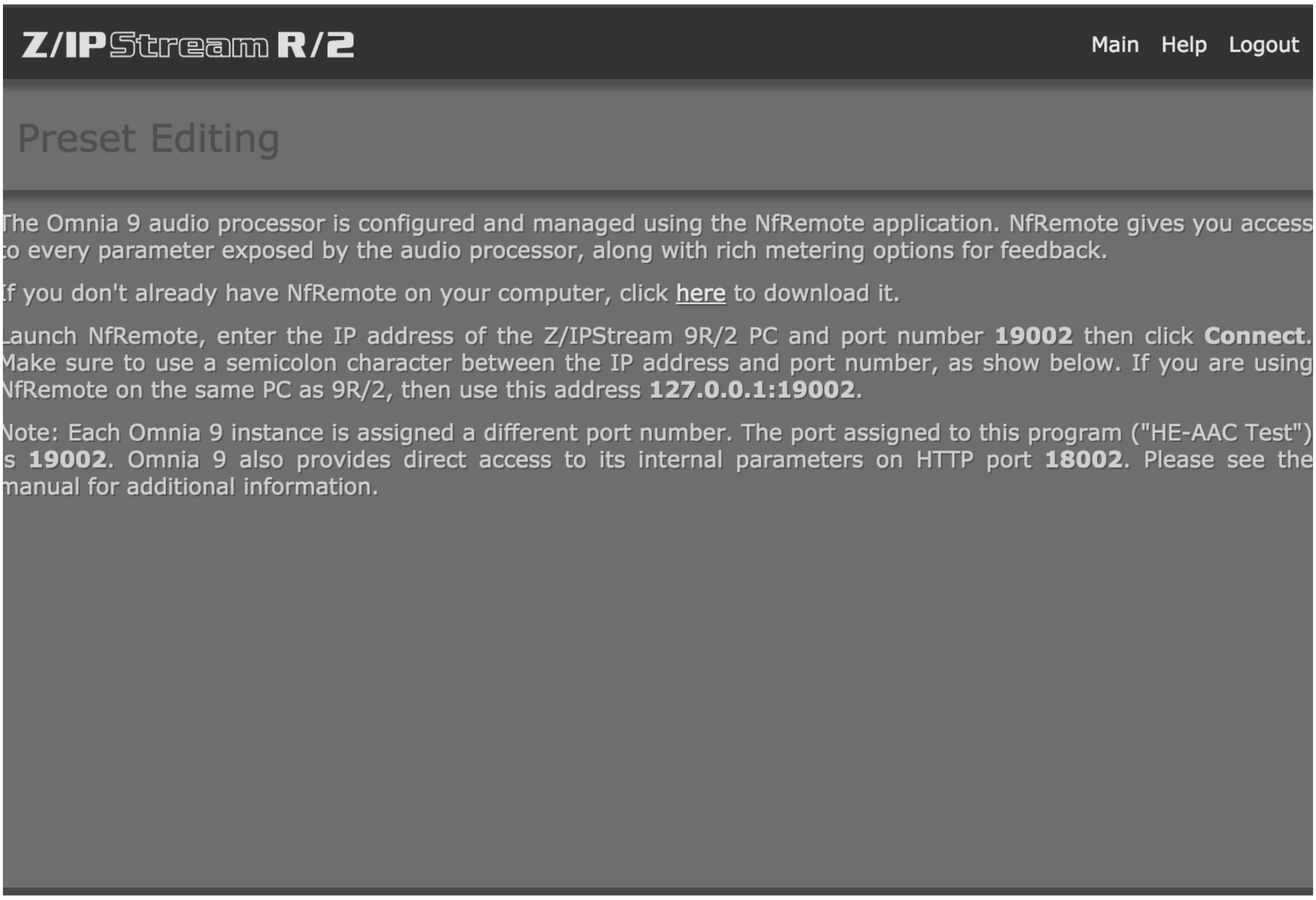
In the sections that follow, we’ll describe the main concepts and all available parameters used to configure the Omnia.9 audio processor. Since the Omnia.9 audio processing engine used with R/2 is based on the hardware audio processor by the same name, the material in this chapter was borrowed from the hardware Omnia.9 manual. Because of this, while the descriptions of the parameters will be accurate, some of the screen images might be slightly different. You may wish to download the entire Omnia.9 manual from http://www.telosalliance.com/omnia/omnia9.

Accessing NfRemote and Connection Information

On the main page of R/2 application, find the instance using Omnia.9 processing that you wish to adjust. Make sure the instance is running and that you have audio present.



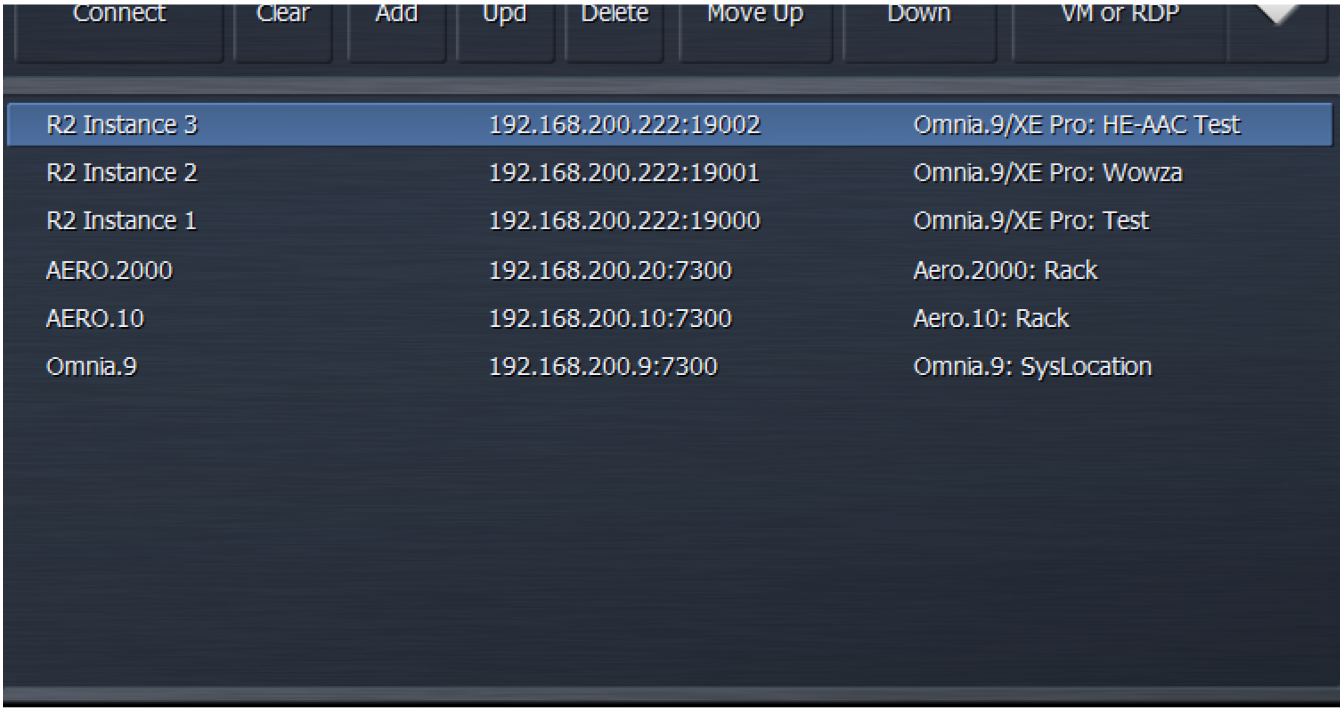
Click the pencil icon found on the Preset line of the instance. The page displayed next will allow you to download NfRemote to your PC if you don't already have it. The page will display information for connecting to this instance through NfRemote. Please note that each instance uses a separate port (i.e. the first instance is 19000, the next instance is 19001, and so on).



Using the NfRemote Client Software

Understanding the NfRemote Connection Window

NfRemote is designed to make remote connections as easy as possible. As you can see, it is possible to connect to any number of NfRemote capable processors individually or concurrently. NfRemote compatible processors include R/2 with Omnia.9, 9/XE software, 9X/2 software, Omnia.9 and Omnia.7 broadcast processors, as well as Linear Acoustic AERO.10/100/1000/2000/soft processors.



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 the audio processing in R/2, 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 between the client and the host PC.

Comment

Allows you to enter a "friendly" name for each device to which you plan to connect.

IP Address

Should contain the IP address of the Z/IPStream R/2.

Password

Should contain the same password established in the Z/IPStream R/2 application.

Audio Streaming

It is possible to stream audio from R/2 directly through 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 PCM

Enables PCM streaming even on connections that are too slow. This choice is best reserved for local networks with sufficient bandwidth to support it.

Connect

Will initiate a connection to the R/2 or other processor whose IP address and port is currently displayed.

Clear

Removes all information from the Comment, IP Address, and Password fields.

Add

Stores 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.

Update

Refreshes the list of saved connections.

Delete

Will remove the selected connection from the list.

Move Up/Down

Will move the selected connection up or down respectively in the list so that you can sort them in any order.

User Interface Pointing Device

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

Use this setting 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.

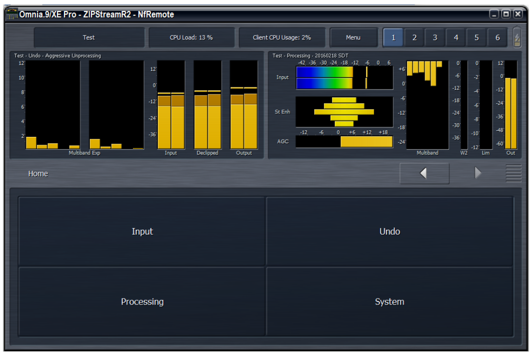
Touch Screen

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

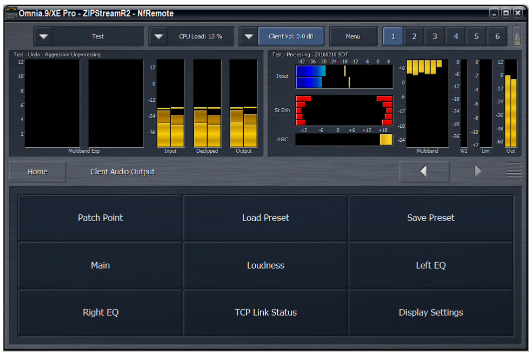
Operates just like Touch Screen but with a large cursor, making it ideal for situations when you are using a projector for demonstration or instructing a group on how to use NfRemote.

Client-Only Controls



The NfRemote interface looks nearly identical to Omnia 9’s window display, and will be described in another section, below. Therefore, only those controls or features unique to the client software will be explained here.

Client Audio



If bandwidth permits, or if Audio Streaming is forced, additional menu items related to client audio output will be available.

NfRemote Home Screen and User Interface

Opening NfRemote 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 is visible at the top of every screen. It shows the station currently being viewed, a “friendly” customizable location reference field (based on the instance name from R/2), the CPU load (host or client), client audio streaming volume, a Menu button, and 6 display preset buttons. The drop down arrows allow selection of what each field displays. “Client Volume” can also be used to control the volume when selected. Drag the control to adjust or double-click to bring up coarse and fine adjustment controls.

Menu

The “Menu” button has seven options: Displays Only, Tree Navigation, Folder Navigation, Expert Mode, Lock Position, Maximize, Full Screen, and Close Window.

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.

Tree Navigation

This is the “default” mode for NfRemote and presents the various control sections in a “tree” with an overview of where within the “tree” you currently are. Click one of the sections to navigate back “up” through the “tree”. For purposes of this manual (and simplicity) we will remain in this mode for the most part, though you may see some screens displayed in “tabbed” or “expert” mode. The controls are the same, even though the presentation and navigation is different. More information about these two modes can be found in the next section of this chapter.

Forward/Back arrows (Tree Navigation Mode)

Moves forward or backwards between the most recently selected control pages.

Tabbed Navigation

For “expert” navigation—Displays multiple “tabs” simultaneously for quicker navigation between various control pages.

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.

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.

Fullscreen

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.

Close Window

Selecting “Close Window” closes the application window and returns to the connection list.

Display Preset buttons 1 – 6

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 are doing to the audio signal.

Navigation Modes

The Omnia 9 User Interface (UI) is set up in “tiers”.

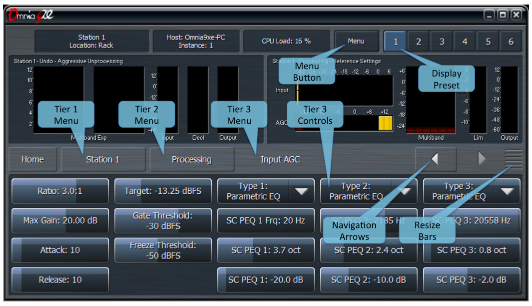
The first tier always contains the highest level (most general) menu options in the form of buttons. When selected, these buttons take you to a second tier of menu options, which are also buttons, but of a more specific nature. The third tier is made up of controls.

There are two navigation modes available from the Menu button: “Tabbed” and “Tree”. The default mode is “Tree”. 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 “Tree” 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 “Tabbed” 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.

"Tree" Navigation Mode

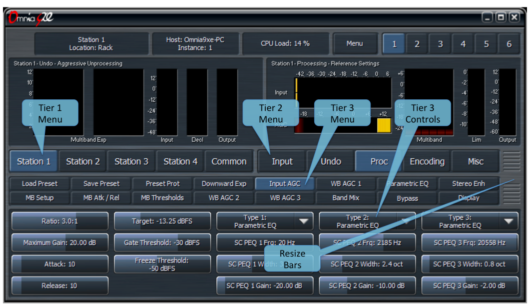


The “Tree” mode is the default. In this mode:

- 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 revisit 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 resize bars.

Tip: Resizing the window will dynamically re-size all controls and meters on the screen. Adjust to suit your preferences. The “resize bar” between controls and metering can be moved as well to devote more screen real estate to one or the other.

"Tabbed" Navigation Mode



Clicking on the Menu button and selecting “Tabbed Navigation” 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 the resize bars.

Control Types

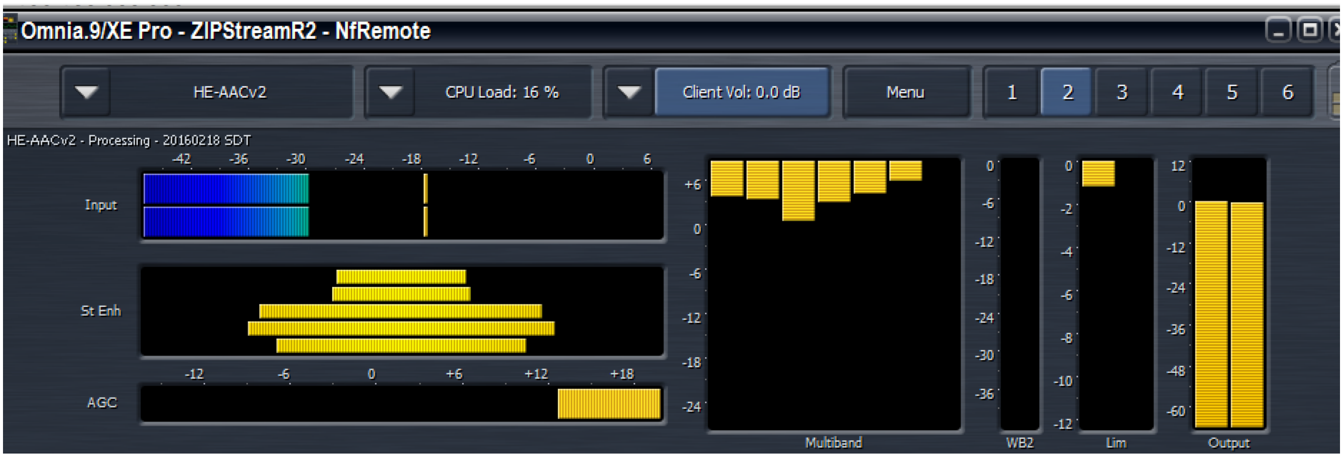
There are three basic types of controls in Omnia 9:

- **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 popup window with two large sliders, one for "coarse" adjustment and one for "fine."

Processing Metering

Understanding the Processing Meters Display

The processing meter display provides a tremendous amount of information about what's going on inside the Omnia.9 processing, 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 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.

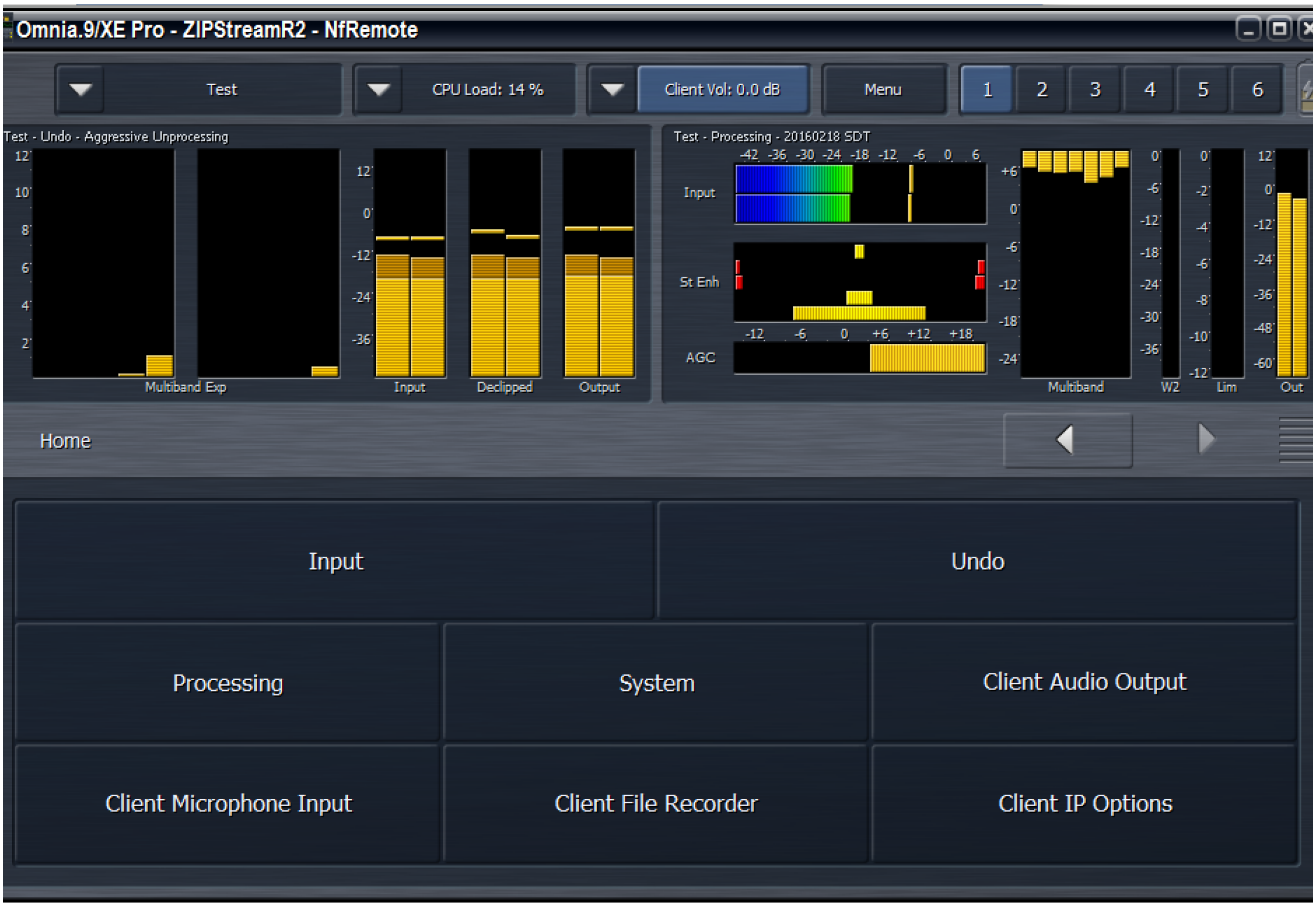
- The **Input AGC** levels are indicated by a bright yellow bar. The Wideband AGC1 levels are shown as a dark orange bar extending beyond the Input AGC meter. Wideband AGC2 levels (if this compressor is situated before the multiband section) are shown as a second, darker orange bar beyond the AGC1 meter.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

The Home Menu

The **Home Menu** represents the highest level menu available in the user interface. It includes sub-menus for Input, “Undo”, Processing, System, and (if available) “Client Audio” related options such as Client Audio Output, Client Microphone Input, Client File Recorder, and Client IP Options.

If you ever lose you way when navigating the menu system, going back to the Home Menu will get you

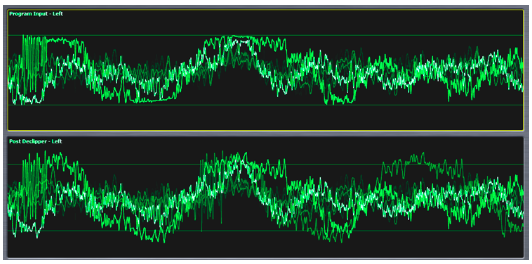


Making Audio Better—One Square Wave at a Time: Undo

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.

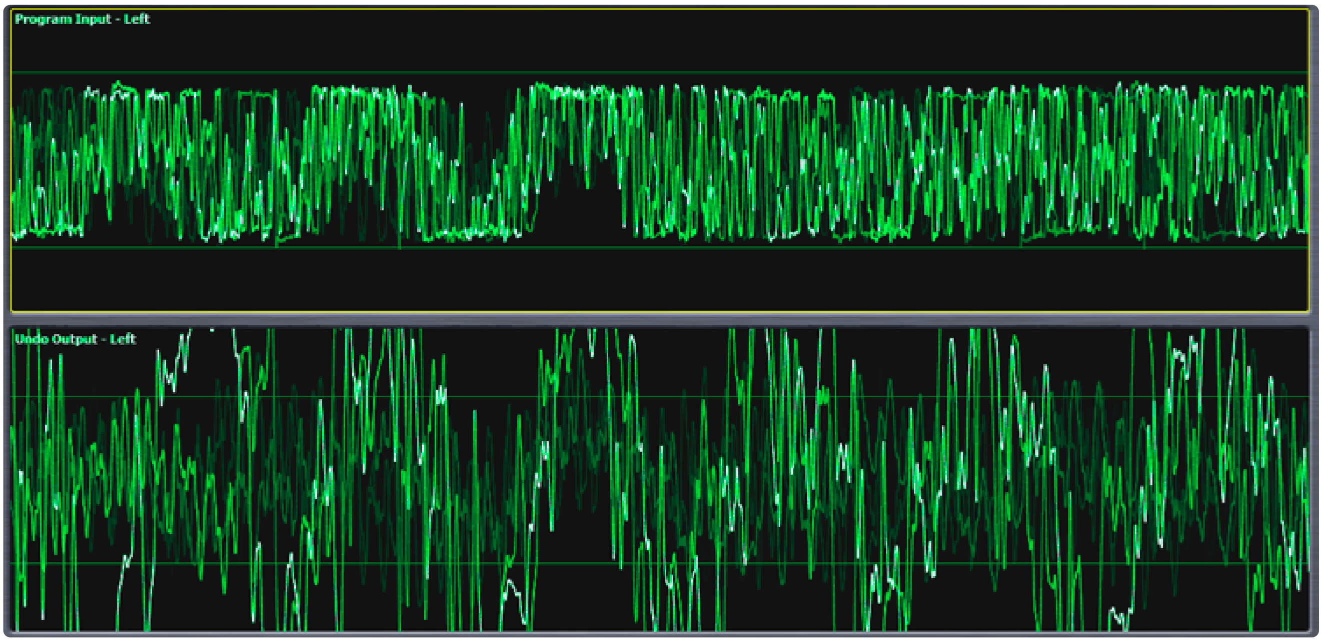


Take a look at the two waveforms below, captured using the oscilloscope built into Omnia.9. The top window shows the damaged source audio. The bottom window shows the same audio after it has passed through the de-clipper. Notice how the peaks that were clipped during mastering have been restored by the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

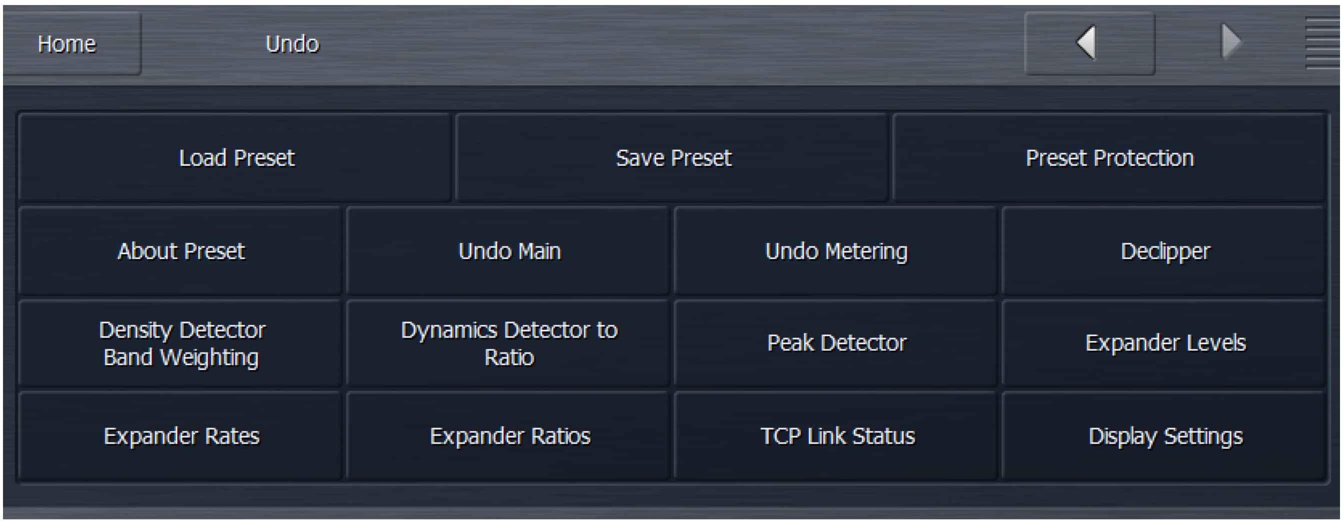
The second stage of Undo is a multi-band expander that is designed to restore dynamic range to audio that has been excessively compressed. Each of the five bands of the expander has a corresponding psychoacoustic dynamics detector and a peak detector that examine the peak levels of the incoming audio, 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

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. TCP Link Status and Display Settings menus are duplicated here for convenience.



A word of caution about Undo! In keeping with the “spirit” of Omnia.9, 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 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.

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.

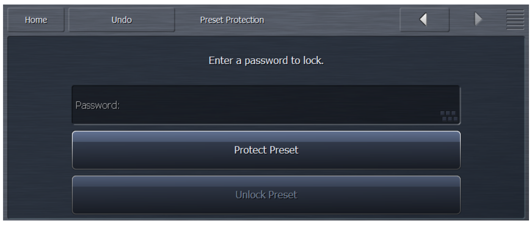
You can rename your current preset, regardless of whether or not is has been modified, by clicking in the field. Type in the new name, and select Save Preset. The new preset will now appear in the list.

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.

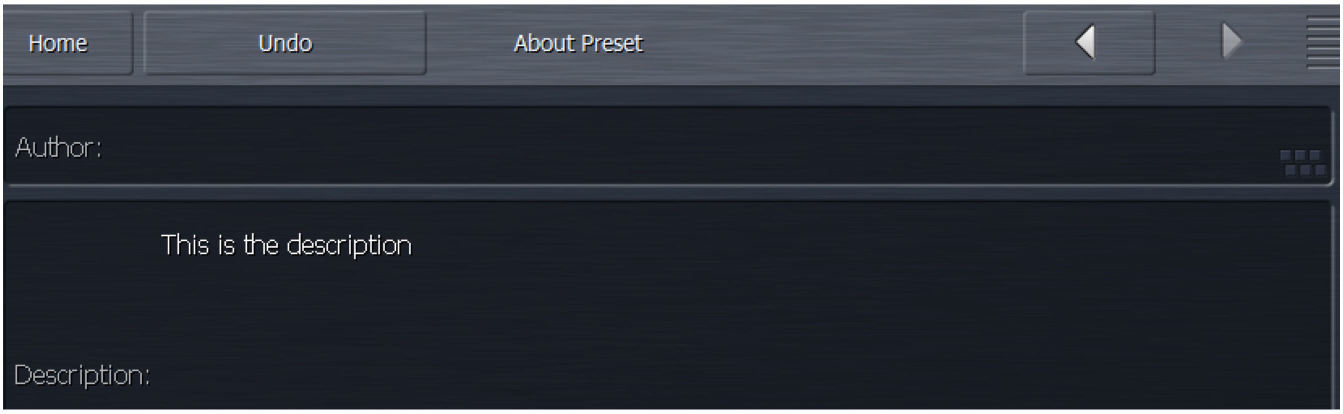
Note that that you cannot give a Custom Preset the same name as an existing Factory Preset.

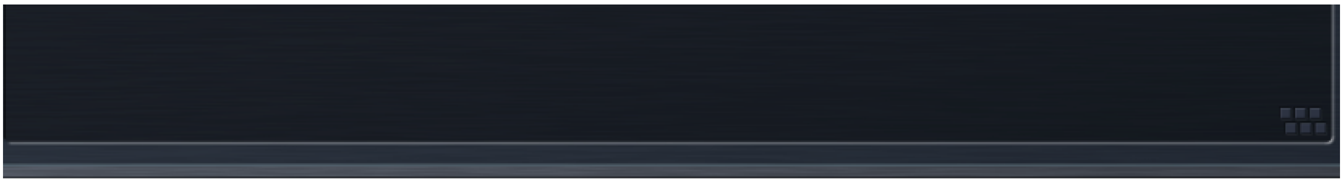
Preset Protection



Allows you to "protect" a preset from modification with a password. Enter the password and click "Protect Preset" to lock, or "Unlock Preset" to unlock an existing protected preset.

About Preset





Allows entry of an author and description for a preset. This could be used for describing changes made in a 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 de-clipper and multiband expander independently.

Power

Turns Undo completely on and off. Turning it off reduces latency and CPU load.

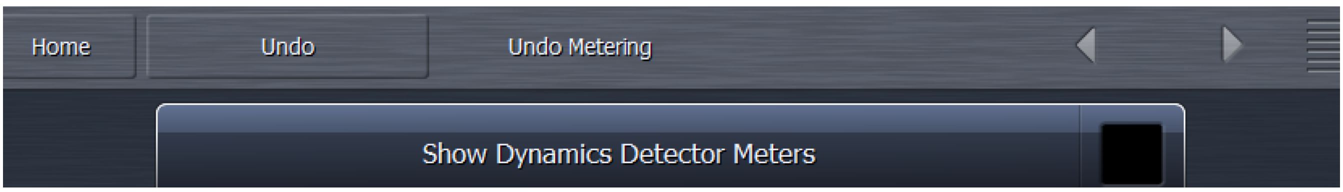
Bypass

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.

Enable De-clipper/Enable Expander

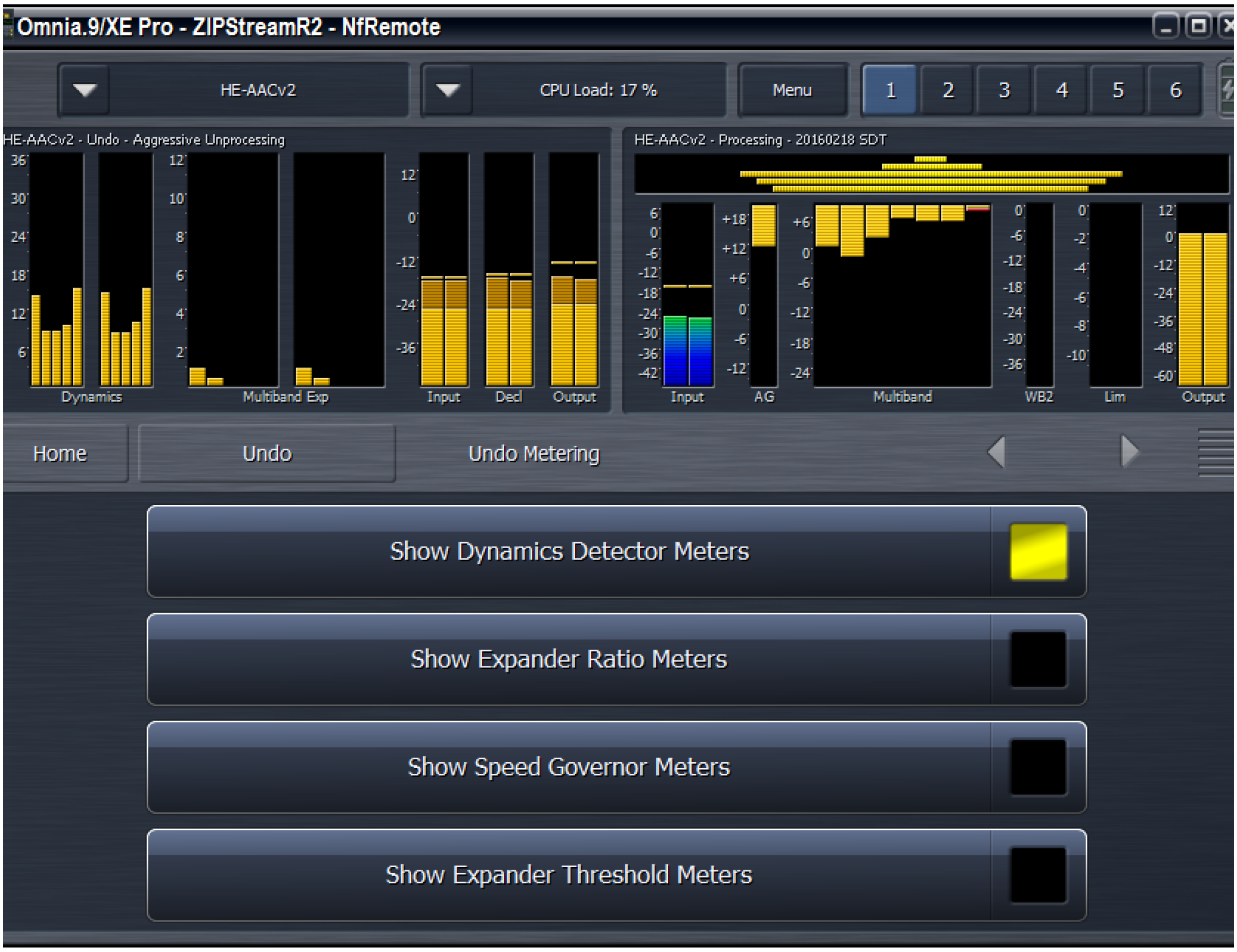
Turns the de-clipper and expander portions of Undo on and off.

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

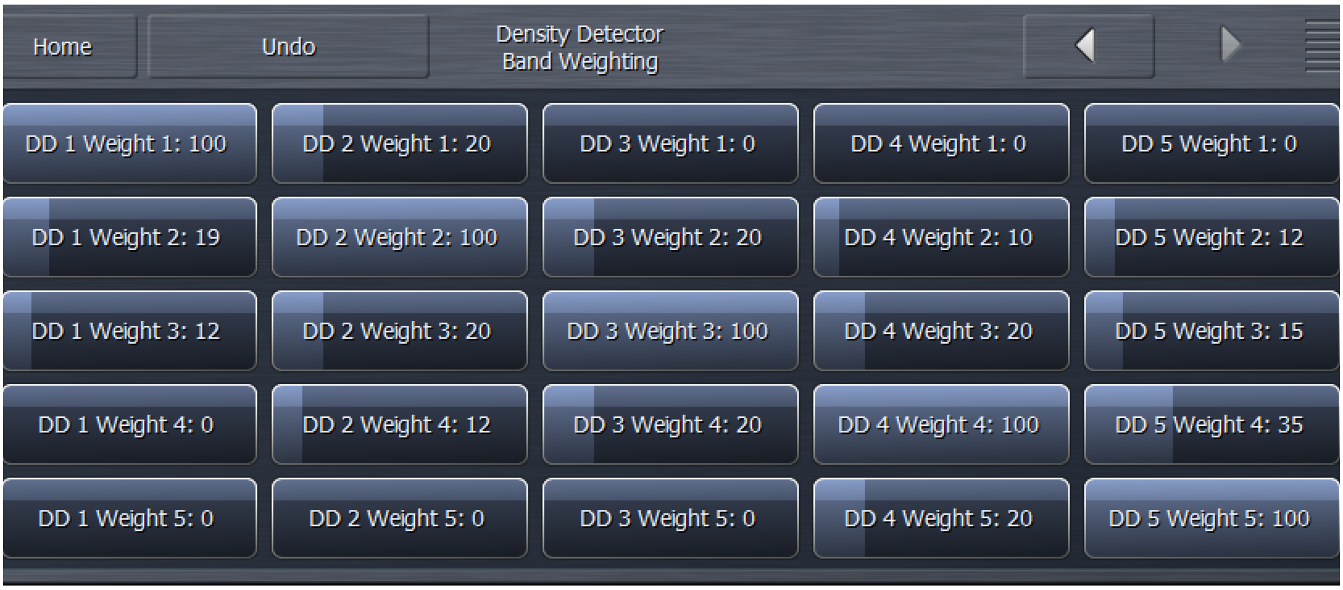
The **De-clipper** menu contains a single control to set the level of de-clipping in 5 stages. Lower settings offer more conservative de-clipping action, while higher settings offer more de-clipping.

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’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.



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.

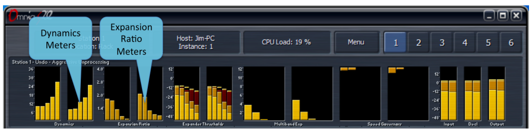
Dynamics Detector to Ratio Menu

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.

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.



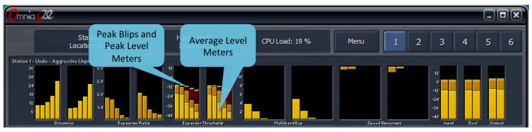
The detected amount of dynamics in each band is indicated by the Dynamics meters (enabled in the previously described “Undo Metering” menu). Taller bars indicate more inherent dynamics in the music, while shorter bars indicate less dynamics and are generally the inverse of the Expansion Ratio meters. The result of settings in the Dynamics Detector to Ratio menu can be observed on the Expansion Ratio meters in

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.

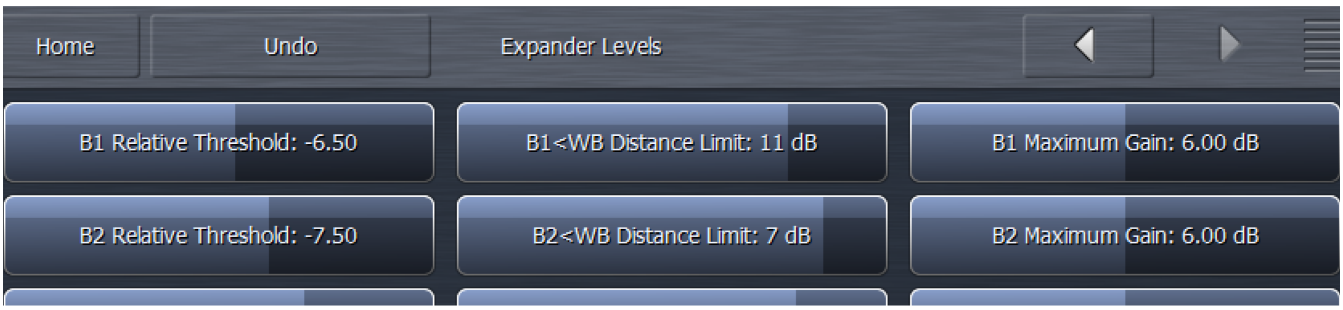
The **Release** speed controls determine how quickly Undo allows the peak values to fall back toward the actual audio levels. Faster times will cause the peak levels to fall more quickly.

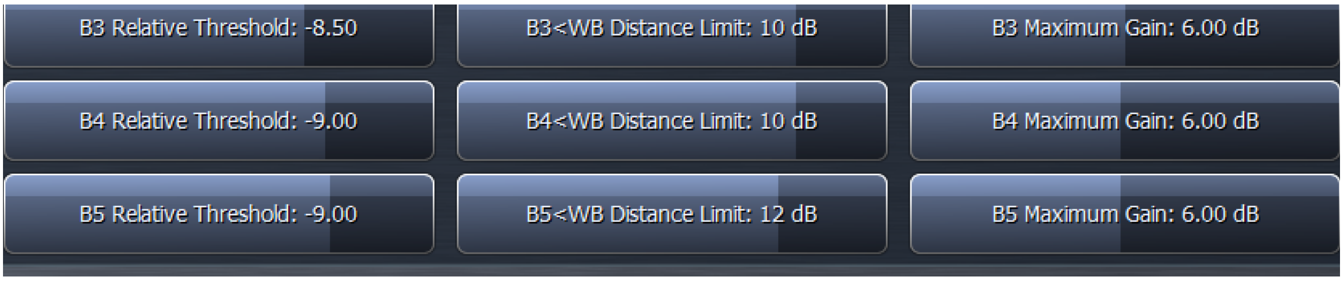


The effect of the Peak Detector settings is represented by the peak “blips” at the top of the dark orange peak level bars within the Expander Threshold meters in the Undo Meter window (enabled in the previously described “Undo Metering” menu). Average levels are indicated by the bright orange bars.

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.

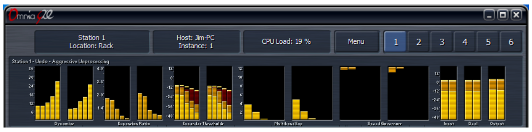




The **Relative Threshold** control for each band determines how far below the currently detected peak the expander threshold sits. Lower thresholds (moving the slider to the right) result in a greater peak-to-threshold distance and therefore provide more expansion. Higher thresholds (moving the slider 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 (enable the Expander Threshold meters through the "Undo Metering" menu).

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.

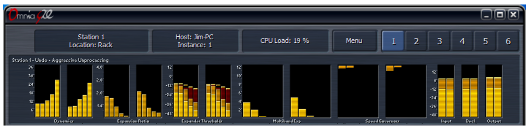
The **Expander Attack** speed control determines how quickly each band's expander will rise when the audio is in need of expansion. Faster settings will expand the audio more quickly Using the built-in oscilloscope to view the Undo Control Signal patch point is very helpful for adjusting attack and release 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 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.

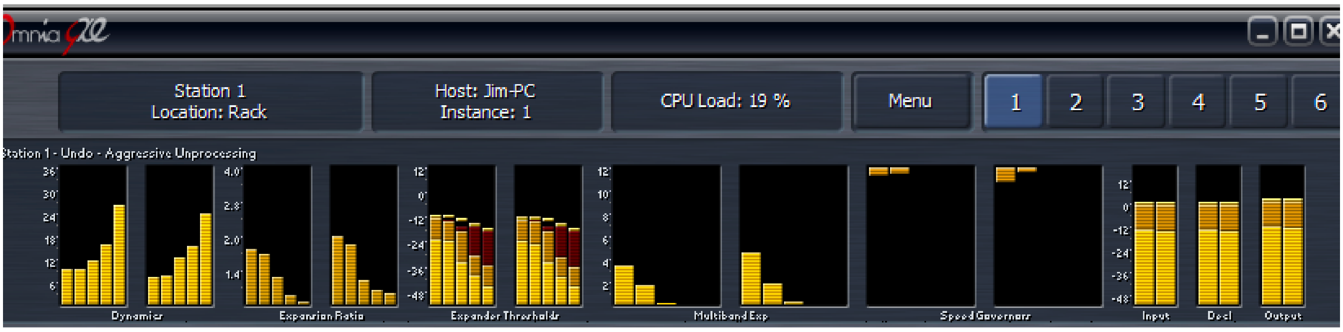


The effect of the expander attack and release speeds are indicated by the **Multiband Expander** meters in the Undo Meter window. The levels of the speed governors are indicated in the **Speed Governors** window (enable the Speed Governors meters through the "Undo Metering" menu).

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.

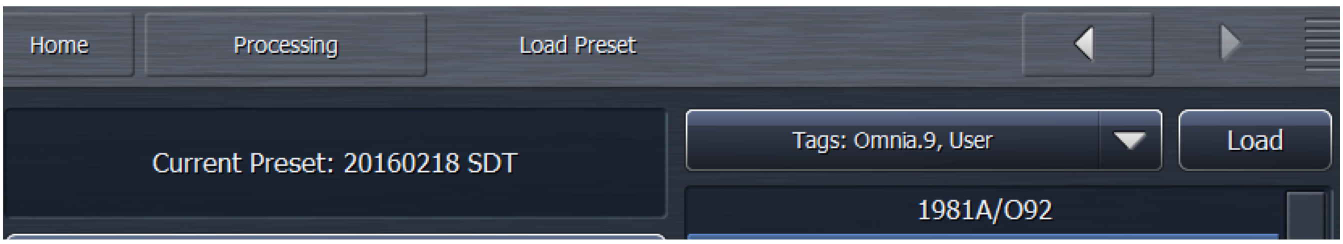


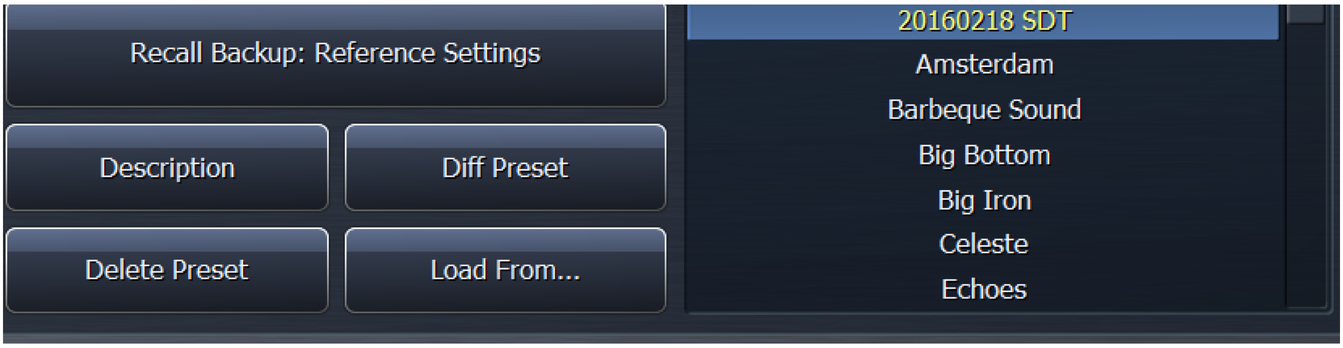
Processing Menu

The **Processing Menu** provides access to all sections of the audio processing core for the connected instance. 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.

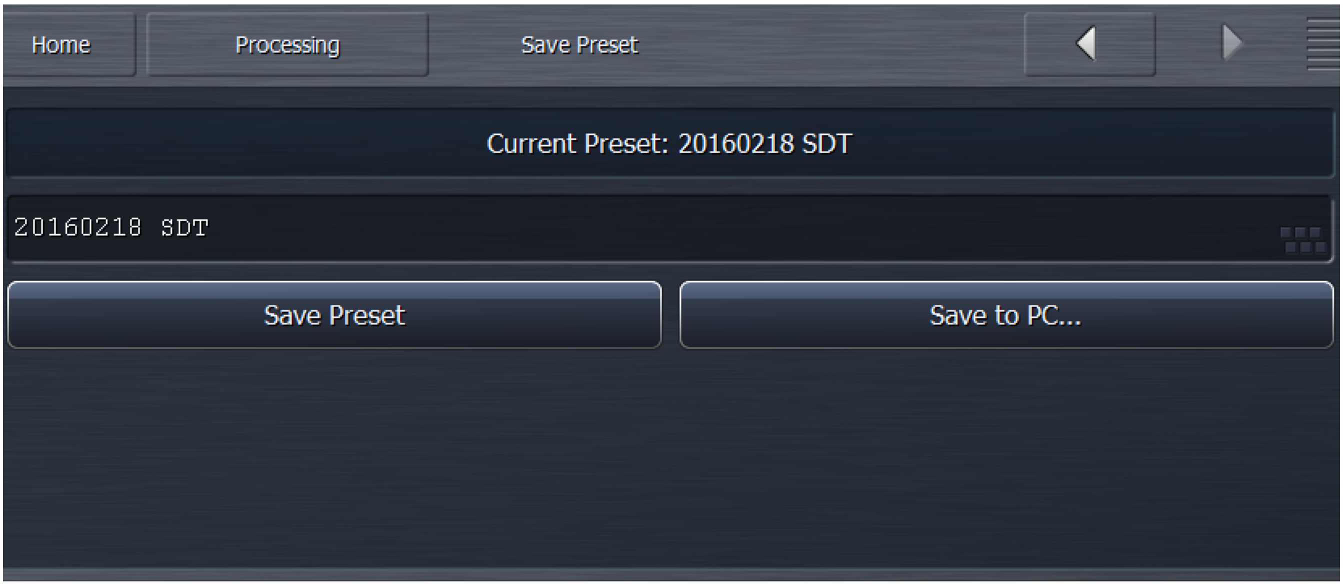
Load Preset

The **Load Preset** controls found here work in the same manner as those in the Undo menu except that they load the processing presets. In addition, presets can be categorized and located by “tags”.





Save Preset Menu



You can rename your current preset, regardless of whether or not is has been modified, by clicking in the field. Type in the new name, and select Save Preset. The new preset will now appear in the list.

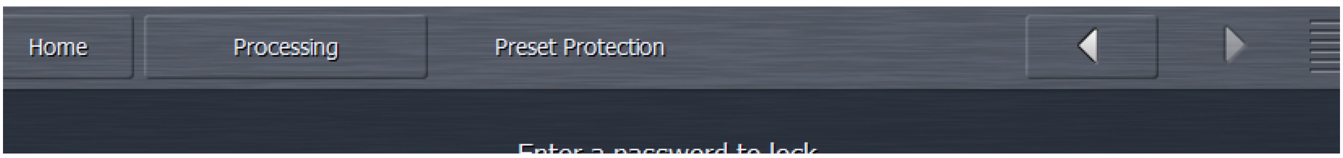
If your Current Preset has been modified, you also have the option to write over the existing preset without renaming it by selecting **Save Preset**.

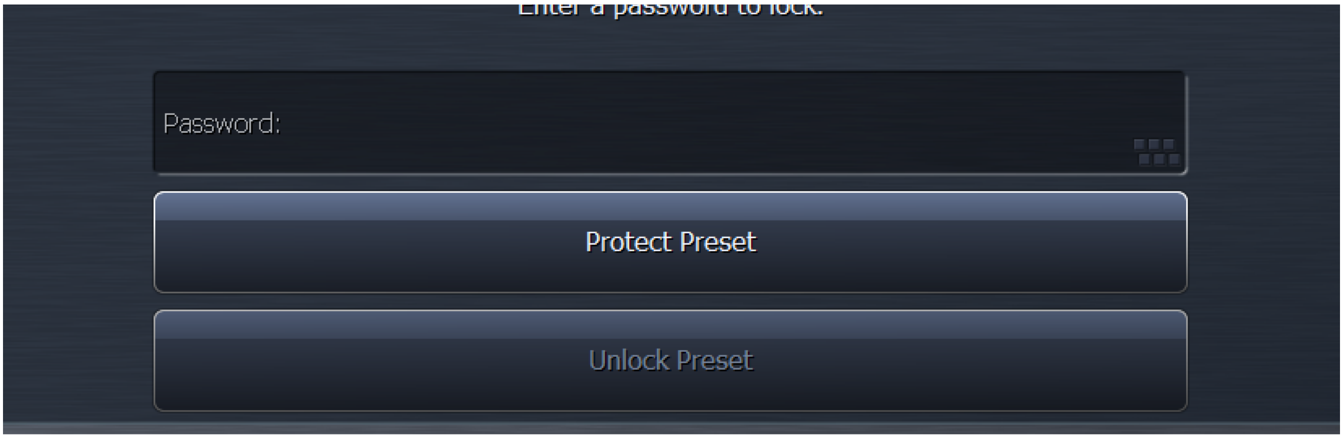
That said, we strongly recommend that after you've made some adjustments and are ready to save those changes to a Custom Preset you give it a unique name and instead of saving directly over your current version. That way, should there come a time when you want to revert to a previous incarnation of a 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.

Note that that you cannot give a Custom Preset the same name as an existing Factory Preset.

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.

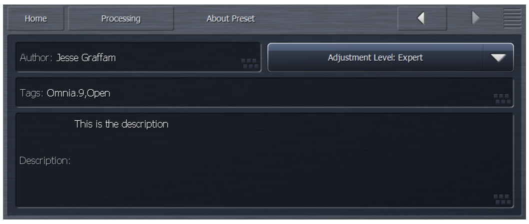




To password protect a preset, enter a password, click on “OK,” and then on “Protect Preset.” A warning dialogue box will appear. If your intention really is to protect the preset, select “Yes.” The final step is saving the new, protected preset with a unique name in the Save Preset menu. Protected presets 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.”

About Preset



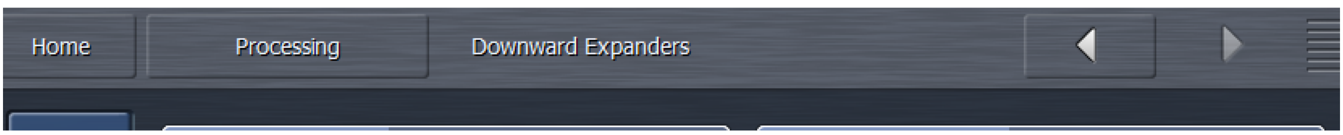
Allows entry of information about the preset, including “tags” which can be used to categorize and quickly locate presets. In addition, several adjustment levels are available, which reveal different controls to the end user. This manual was written in “expert” mode. All changes made in the a given mode can be carried over to the other modes as well.

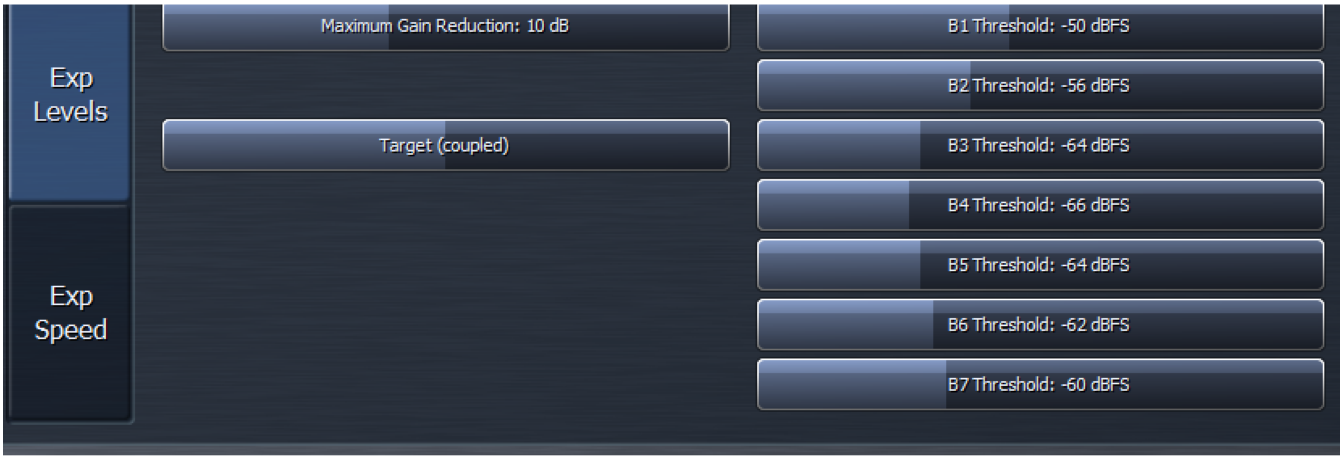
Downward Expanders Menu

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.

Expansion levels





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.

Expansion Speed

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 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 operate above AND below threshold, controlling the dynamics over a much wider range and do not require makeup gain.

Main

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 the Omnia.9 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.

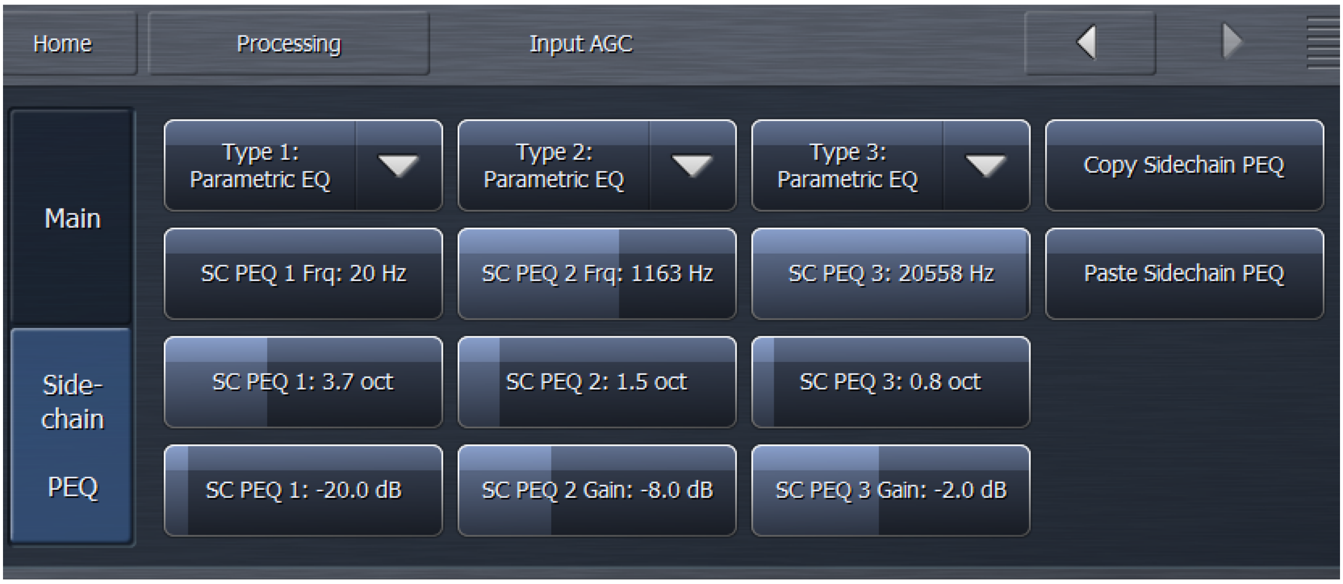
The **Release** control determines the speed with which the Input AGC increases audio below threshold. Lower settings provide slower release speeds, while higher settings result in faster

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 mean the audio must drop to a lower level before the release speed slows or freezes. Higher settings mean 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.

Sidechain PEQ



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

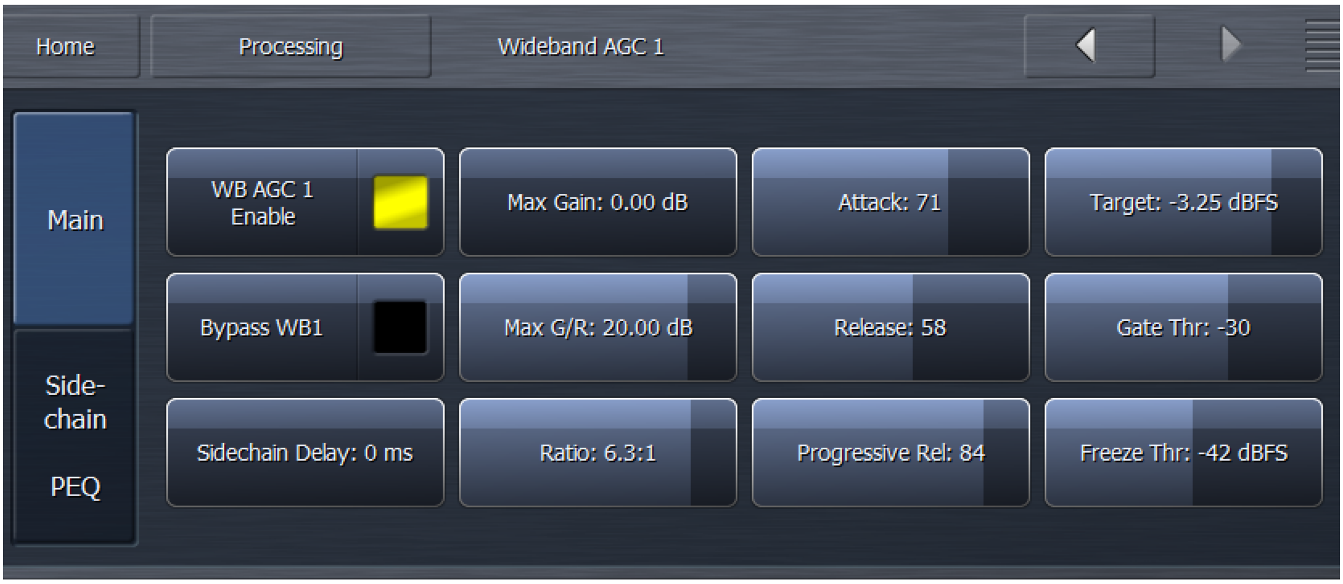
range of 24db per band.
Copy/Paste of Sidechain EQ settings between AGC sections is provided.

Note: You can see a visual representation of the sidechain EQ curve in a display window using Omnia 9'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.

Main



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. However, the Maximum Gain control in the Wideband AGC1 section has a range of 24dB.

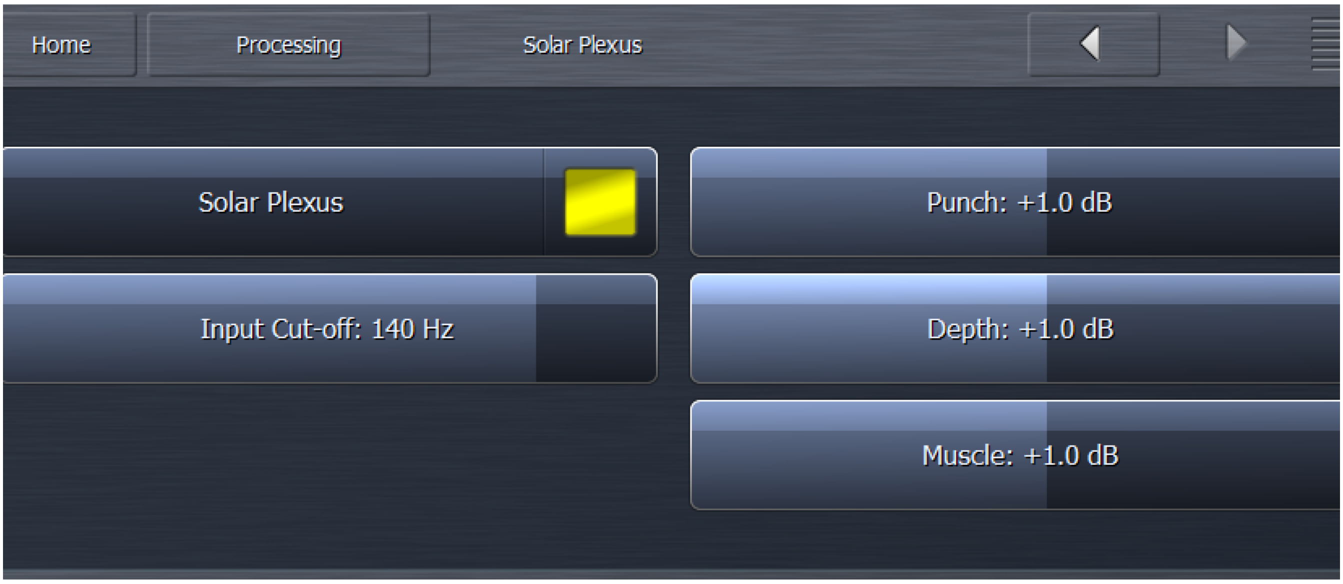
The **Maximum Gain Reduction** control sets the maximum amount of gain reduction possible in the Wideband AGC1 compressor, and is adjustable from 0 to 24dB in one-quarter dB increments.

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.

Sidechain PEQ

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.

Solar Plexus

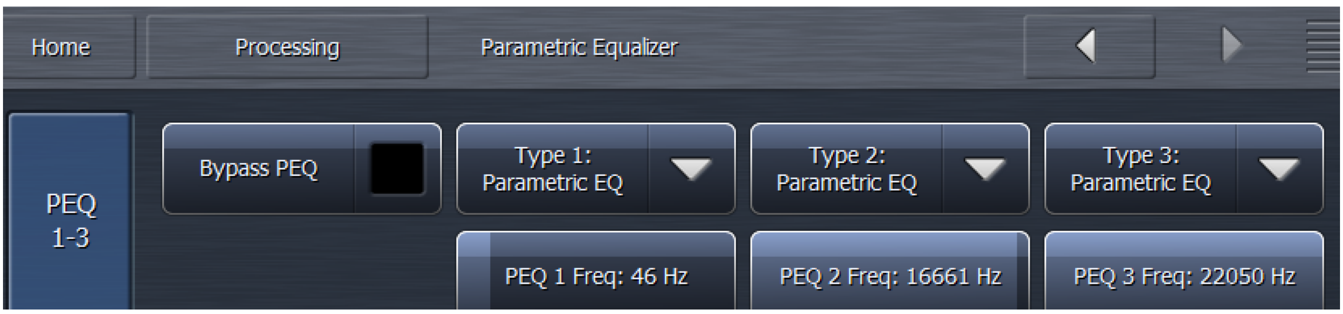


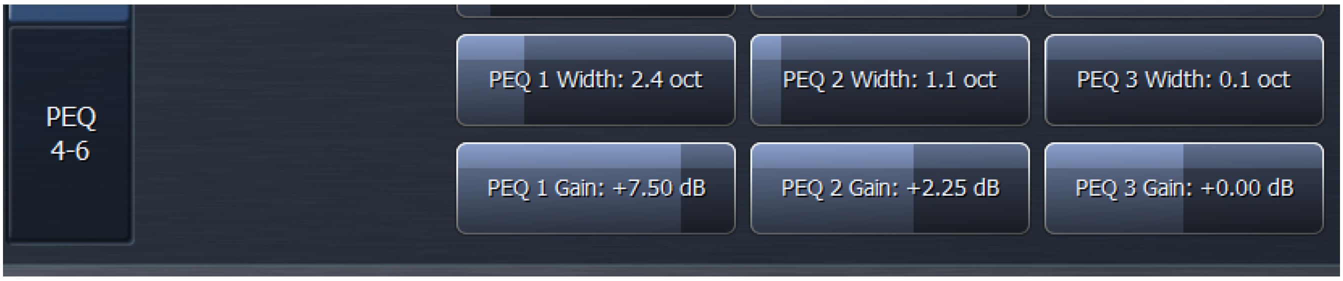
_Screen%20Shots/16-35-NfRemote-Solar%20Plexus.png

Solar Plexus is Omnia exclusive bass enhancement technology that adds lower octave energy to program material. A little bit goes a long way. Experiment with the controls to find a combination that works best for your needs.

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. All bands are identical and can be set in any configuration.





Bypass

Removes the equalizer from the audio path.

Type

Determines what type of EQ or filter is employed.

Frequency

Sets the center frequency for each band. The range of this control is 20 to 22,050Hz.

Width

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.

Gain

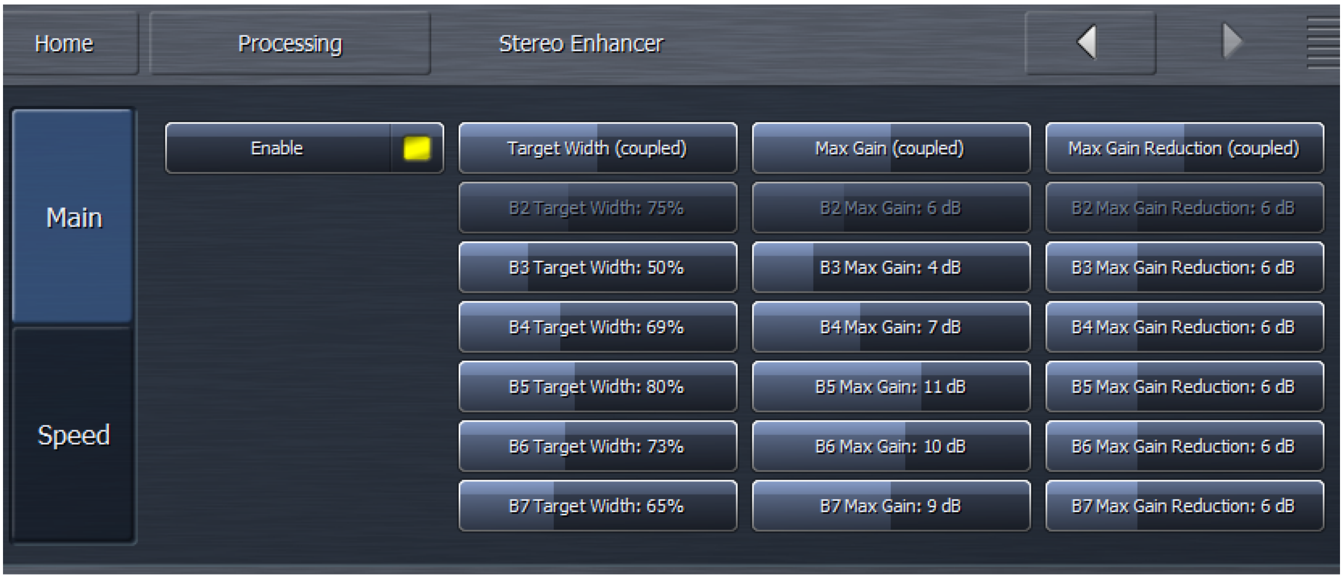
Determines how much the audio selected with a combination of the Frequency and Width sliders is boosted or cut. Each band can be boosted or cut by 12dB in one-quarter dB increments for a total range of 24db per band.

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 is a very versatile and powerful tool for creating your on-air sound. A visual representation of the effects of the PEQ can be displayed using the built-in real time analyzer.

Stereo Enhancer Menu

Omnia 9 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. 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.

Main



Enable

Turns the Stereo Enhancer on or off.

Target Width

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.

Maximum Gain

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.

Maximum Gain Reduction

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.

Speed

Attack

Determines the speed at which the stereo image is narrowed.

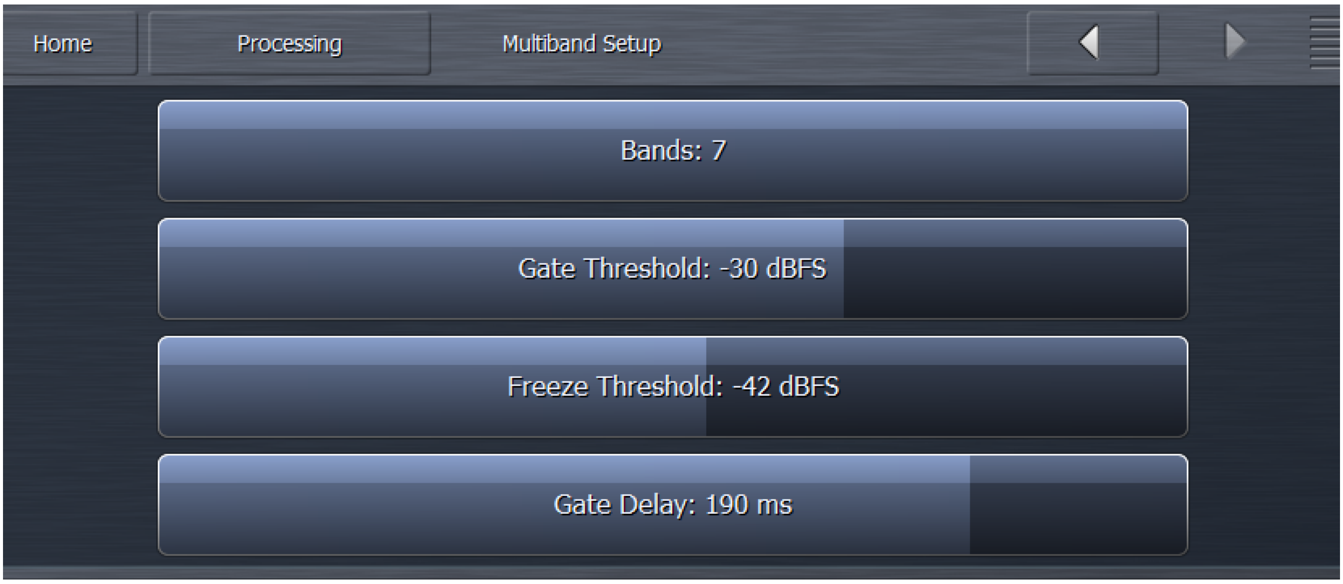
Release

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 gate threshold, freeze threshold, and gate delay controls.



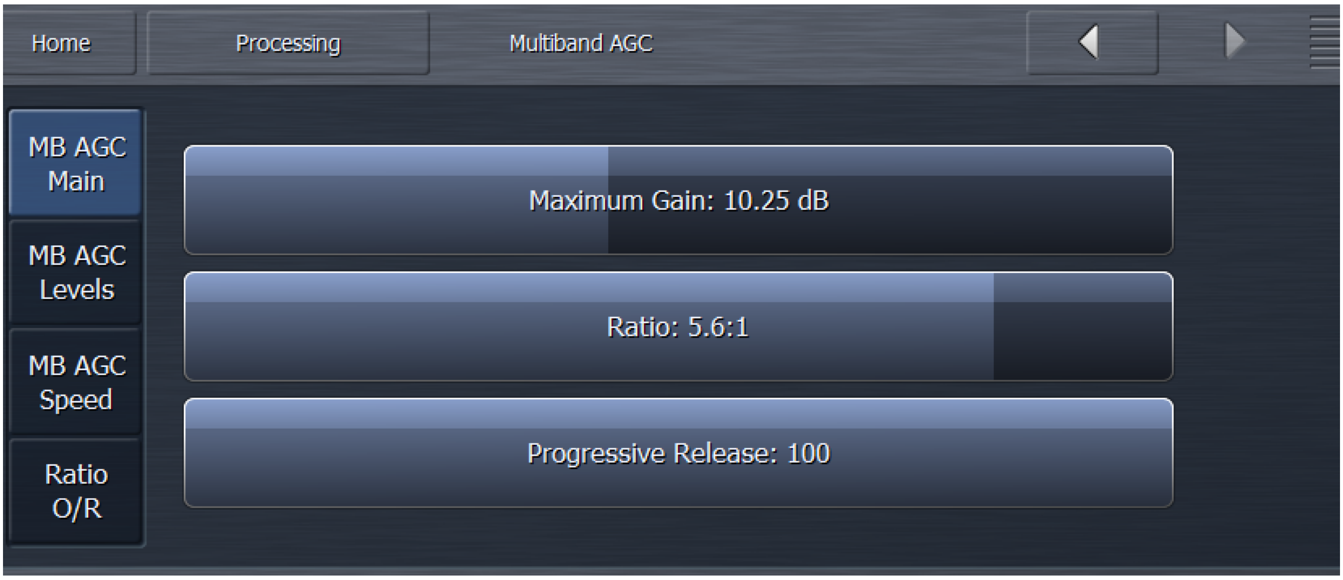
The **Band** slider determines the number of bands in the multiband processing section and ranges from 2 to 7.

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

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.

Multiband AGC - Main

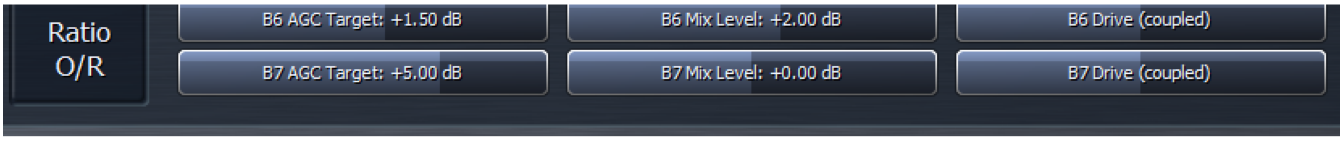


The **Maximum Gain** control works in conjunction with the Ratio control to determine how much gain is available below target. If the Input AGC Ratio is set at Infinity:1 and the Input AGC Maximum Gain is set to 36dB, the Input AGC has 36dB of range below target. At a ratio of 2.0:1 and the same Maximum 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 target.

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 that 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 **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.

Multiband AGC Levels



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.

The **Band Mix** controls allow you to adjust the final output of each band after all Wideband and Multiband processing has been applied and are duplicates of those found in the separate Band Mix menu. Please see the section on the Band Mix menu below which contains additional important information about these controls.

When raising the target of a particular band (to increase gain and consequently raise its output) it was normally necessary to reduce the output of the same band in the band mix control to compensate. Likewise, lowering the target of a band (to decrease gain and therefore lower its output) it was necessary to increase the output of that band in the band mix menu.

The **Drive** controls provide simultaneous and opposite adjustment to both the AGC Target and Band Mix controls. This allows the overall spectral balance to remain consistent.

It is worth mentioning again here the importance of a concept unique to Omnia.9. 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 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 AGC Speed

The **Multiband AGC Speed** 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. 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 AGC Ratio Override

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.

Multiband Compression

Enables compression within the multiband section, in addition to multiband AGC. These controls set the maximum gain and ratio for each band of compression.

Multiband Compression – Speed

Controls the speed at which the multiband compressors operate. These functions operate identically to the functions described earlier in the multiband AGC section.

Multiband Compression – Delay

The Sidechain Delay feature is useful for both adding punch and managing the amount power within a particular band (especially low frequency bands). This particular menu controls the sidechain delay specifically within each band of the multiband compressors.

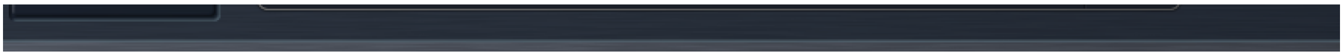
Multiband Thresholds Menu

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.

Wet Voice Detection Menu



Speech is one of the most difficult waveforms to process cleanly, as the human voice is complex in nature and is typically asymmetrical in form. Stations that choose to process aggressively in an effort to maximize loudness may find that bare vocals come through with unacceptably high levels of audible distortion.

The Wet Voice Detection menu (so named because it not only works on bare, “dry” speech but also with speech contained within music) houses the controls that allows Omnia.9 to overcome this by automatically detecting (in the Auto mode) when the input audio is speech and using a separate set of multiband targets, attack rates, and release rates. This allows the dynamics section to do more of the “heavy lifting” and reduces the amount of clipping necessary to maintain the same level of loudness.

Setting the Speech control to “Off” completely disables the speech detection circuit. “Auto” allows the processor to automatically detect the presence of speech. “Force” overrides the main multiband settings and uses the Speech Detection settings (as determined in the current preset) at all times.

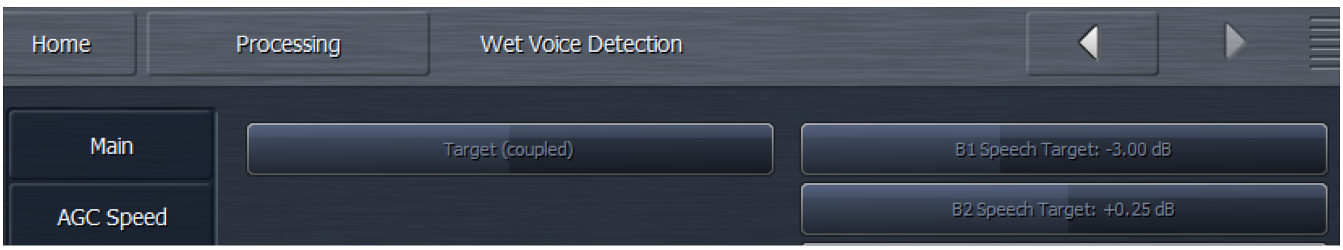
The default setting depends upon preset chosen; most presets have this featured set to “Off” by default, while the default setting for more aggressive, loudness-driven presets is “Auto.”

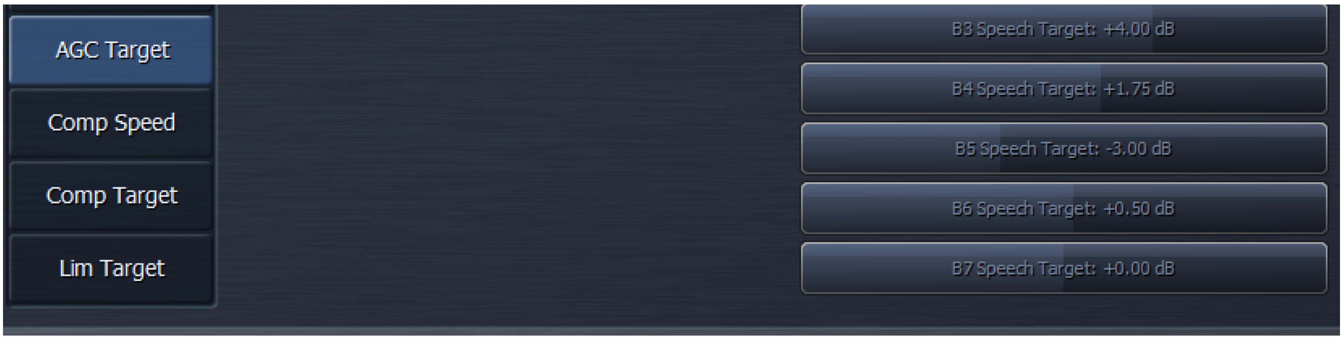
Separate speed and target values for compression and AGC in each band of the multiband section can be selected here. Separate target values can be set for the multiband limiter section as well. These values will be applied whenever speech is detected.

Wet Voice – AGC Speed

Controls the attack and release thresholds for the multiband AGC when speech is detected.

Wet Voice – AGC Target





Controls multiband AGC target values when speech is detected.

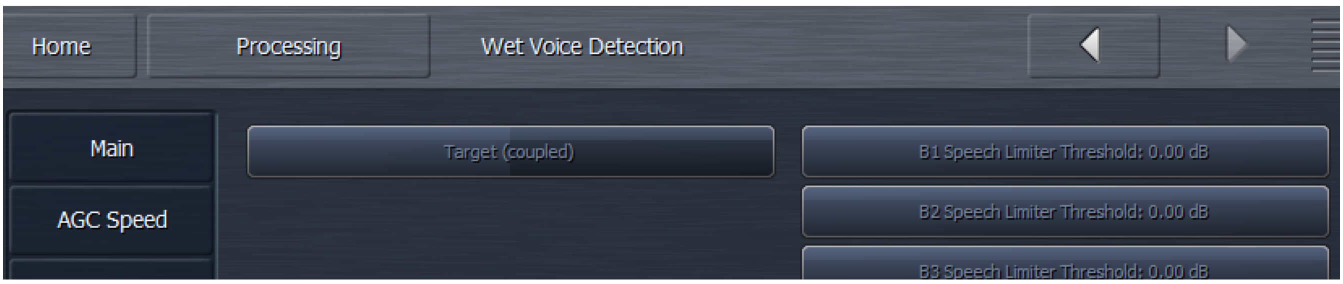
Wet Voice – Comp Speed

Controls the attack and release thresholds for the multiband compression when speech is detected.

Wet Voice – Comp Target

Controls multiband compression target threshold values when speech is detected.

Wet Voice – Limiter Target



Controls multiband limiter target threshold values when speech is detected.

Wideband AGC2 Menu

The 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 AGC2 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 **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.

Wideband AGC 2 – Sidechain PEQ

Controls the sidechain PEQ for Wideband AGC 2. These EQ controls function identically to the ones contained in Input AGC and Wideband AGC 1.

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.

Wideband AGC 3 – Sidechain PEQ

Controls the sidechain PEQ for Wideband AGC 3. These EQ controls function identically to the ones contained in Input AGC, Wideband AGC 1, and Wideband AGC 2.

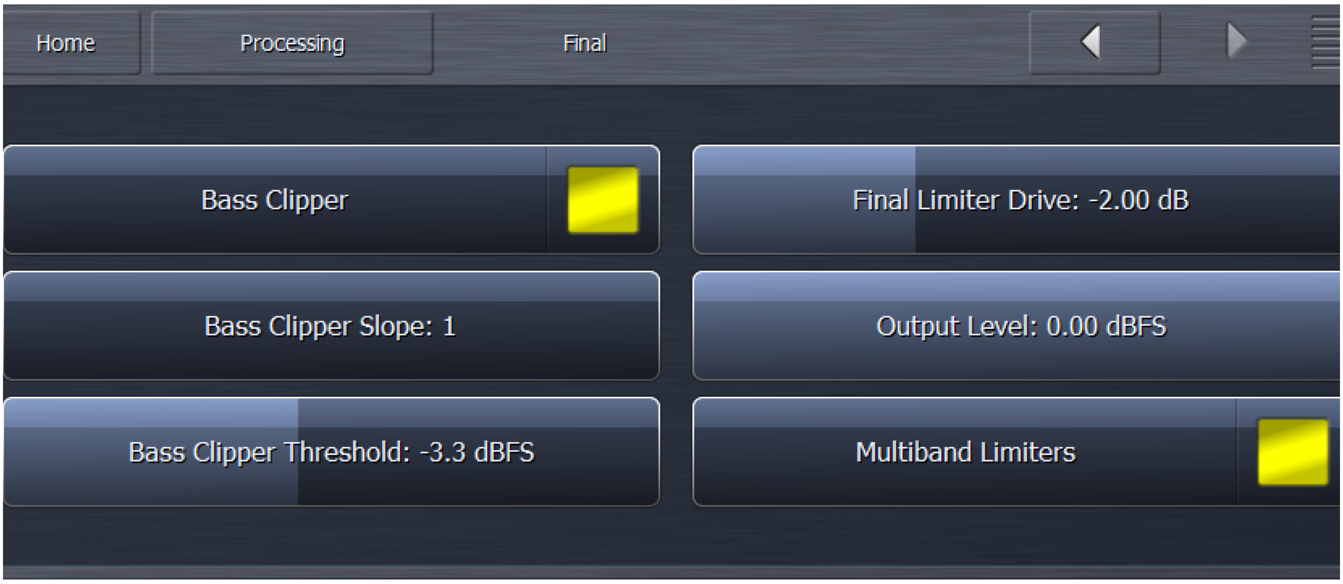
Band Mix Menu

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.

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.

Final



Final output controls for bass clipper, final limiter, multiband limiter enable/disable, and output level.

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.

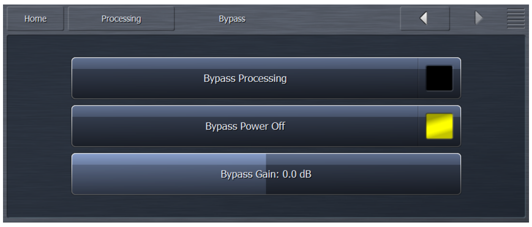
The Bass Clipper Threshold control sets the threshold of the Bass Clipper. Raising the threshold (moving the slider to the left) reduces the amount of clipping performed by the Bass Clipper, but place a greater burden on the Final Limiter. Conversely, lowering the threshold (moving the slider to the 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.

Factory Presets

The Omnia.9 audio processor included with Z/IPStream R/2 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 handpicked “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’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 strongbass 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’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 multiband 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’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’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’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 multiband 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 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 multiband 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 and 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.

Frequently Asked Questions (FAQ)

Q: Do I really need to use the included rails for mounting R/2 in the rack?

A: Not necessarily, but it does keep strain off the front panel and makes it easier to mount in the rack. If you don’t want to use the rails, you might still want to consider placing an inverted rack shelf underneath R/2 for support.

Q: I’m not getting any response from R/2 at all…No LEDs, no fans, no front panel display. What’s wrong?

A: Check that you’ve connected AC power to both power supplies. A green LED should illuminate on each supply to indicate presence of AC power. Try different power cords (stranger things have happened) and check that the AC power outlet(s) are actually live. Press the front panel “On/Off” button. If R/2 still doesn’t power up, contact technical support.

Q: I’ve connected power, but the unit won’t stop beeping. How do I stop the beeping?

A: You probably did not connect AC power to both supplies. There are a number of good reasons to do this, particularly if your facility is designed with redundant AC power feeds to each rack, though you should attach power to both supplies even if this is not the case. If the unit still beeps after you have connected power to both supplies and double-checked that the the power source(s) feeding both supplies are actually good (and that the power cords are OK) you may have a power supply fault. Remove the faulty or disconnected supply and the beeping should stop.

Q: I powered up the R/2 and noticed a lot of fan noise. Is this normal? Can I mount R/2 in the studio?

A: The fans on R/2 are temperature controlled and individually monitored for speed. They are designed to spin up to full speed at power up, but should slow down to a more moderate speed during boot. A slight increase in fan speed during operation could indicate elevated internal temperatures (possibly due to lack of airflow) but if the fans remain at full speed after boot or during otherwise normal operation, it may indicate a fan or other hardware failure. While the R/2 can operate relatively quietly, it would likely be audible in a studio environment with live mics. We recommend mounting R/2 in the rack room or other control room away from live mics if possible. It isn’t necessary to access the front panel after initial setup. All monitoring and configuration can be performed remotely via the web GUI once the unit has been powered up and assigned an IP address.

Q: Why can’t I access R/2 via the network?

A: Check that you have set the IP address, subnet mask, and gateway correctly via the front panel, or that a valid IP configuration has been acquired via DHCP (Chapter 5). Verify that you are connected to the correct network interface port (Chapter 3: NIC 1 on the left, NIC 2 on the right). Ensure that the link and activity LEDs on the corresponding port illuminate, and that the corresponding (Net 1/Net 2) LED on the front panel flashes with network activity (Chapter 4). Note that R/2 will NOT respond to a “ping” request, though you should be able to access the internal web GUI from a browser (Chapter 6). Verify that you are able to use the same port on the network switch with another system.

Q: I’ve got R/2 up and running, now what?

A: Once you’re able to access the web GUI, the default login is “user” with no password. First, add an audio source (Chapter 7). Second, add a processing or encoding instance (Chapter 8). Finally, adjust the processing as desired (Chapter 9 for Omnia 3, Chapter 16 for Omnia.9).

Q: Why don’t I see anything in the list of “Discovered Channels” under the “Livewire Driver” settings?

A: Verify that at least one of the R/2 network interfaces is configured and connected to your Livewire network, and that the proper interface is selected in the “Livewire Driver” settings (Chapter 7). If you still aren’t seeing any channels in the list, try restarting the Z/IPStream R/2 Encoder serivce (Chapter 14).

Q: I’ve configured several audio sources from Livewire or RTP, but why don’t I see any audio levels on the front panel?

A: By default, the front panel will display the AES audio input levels. Livewire audio sources can be displayed by using the up/down arrow keys from the metering screen. All other processing and audio level metering can be performed via the web GUI (Chapters 6 through 9) or via NfRemote for Omnia.9 processing (Chapter 16).

Q: Why aren’t the local AES/EBU audio card inputs appearing in the list of available WAVE devices?

A: This may indicate an issue with the audio card. Shut down R/2 and remove power from both AC power inlets for 30-60 seconds. Power R/2 back up. If the AES inputs are still not available to add as WAVE (Local Audio Card) sources, contact technical support.

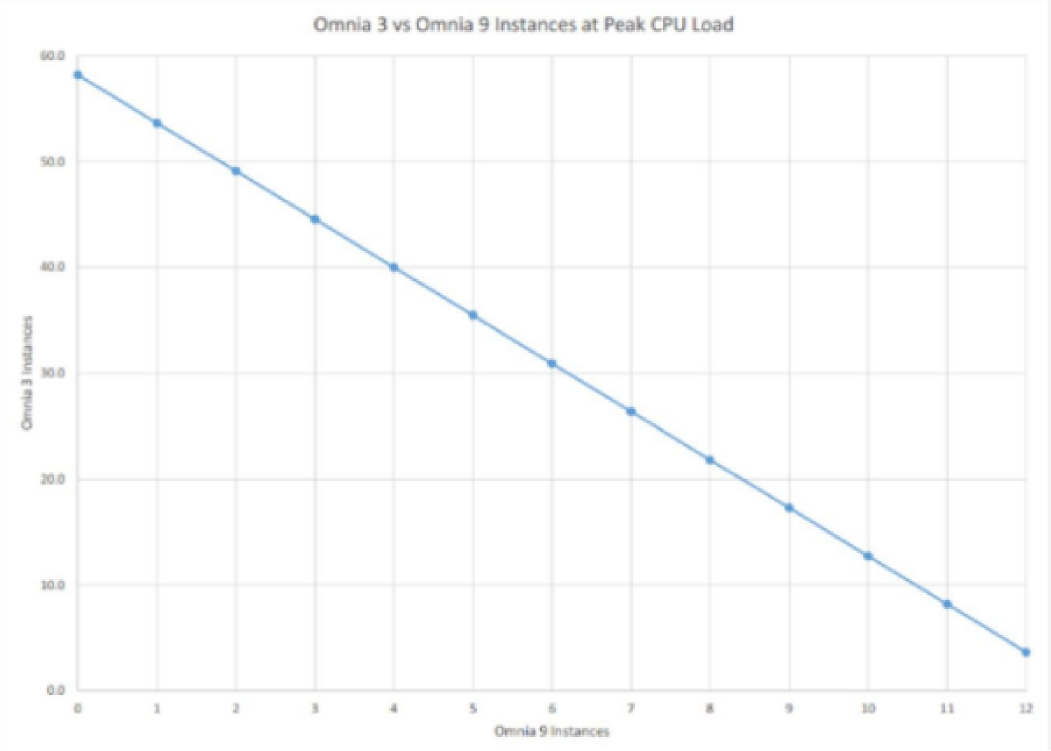
Q: How can I add Omnia.9 processing or additional audio sources to my R/2?

A: Contact sales or your favorite Telos/Omnia dealer to license additional sources (see Chapter 12 for license management). Up to 8 audio sources are supported on R/2, each with included Omnia 3 processing or optional Omnia.9 processing. These sources can be each used in multiple processing and encoding instances simultaneously (see Chapers 7, 8, 9 and 16).

Specifications

Processing

Includes up to 8 instances of standard 3-band Omnia processing or full Omnia.9 processing with up to 7 bands of processing. The number of audio processing instances that may be used simultaneously depends on overall system configuration and resource usage. As expected, Omnia.9 is more resource-intensive than the 3-band Omnia processor. The chart below illustrates the number of instances that can be run under typical usage scenarios. It is provided as a guide, the actual number may be different for your specific application.



Stream encoding

Each channel supports AAC-LC, HE-AAC, HE-AAC v2, and MP3 encoding at bitrates from 16 kbps up to 320 kbps (depending on codec). A program may be encoded using multiple codec formats and bitrates simultaneously. A special multi-rate encoder supports encoding for adaptive streaming applications. The multi-rate encoder properly generates the required Stream Access Points for adaptive streaming.

Ethernet Remote Control

- Gigabit Ethernet supports HTML web interface for administration
- REST API for remote control,
- SNMP monitoring
- Dedicated control application for Omnia.9 processing
- Various metadata update methods via Ethernet supported as well.

Front Panel Controls and Indicators

- No front panel controls
- Diagnostic LED Indicators for power, network and drive activity

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Audio I/O

- Livewire+/AES67 and AES/EBU audio I/O
- Supports AES/EBU input at up to 24 bits, 92 kHz
- Supports direct input from RTP streams

Power Requirements

Dual power supplies, each rated at 100-264 VAC, 50/60Hz, auto-sensing, 100W max. total.

Dimensions and Weight

- One rack unit- 1.75”H x 19”W x 15.5”D (44 x 483 x 394 mm)
- Net weight: 9 lbs (4 kg); shipping: 12 lbs (5.4 kg) approximate.

Environmental

- Fan cooled
- Operating environment: 0 to 50 degrees C
- Non-operating –20 to 70 degrees C

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Warranty

Standard 2-year limited parts and labor. For the latest Telos Alliance warranty, visit: telosalliance.com/warranty

Release Notes

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Z/IPStream R/2 v1.08.02 - December 15, 2020
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Versions certified by Kantar Media
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- v1.07.01+ beta as module v1.22(5.1)
- v1.05


----------------------------------------
RESOLVED ISSUES in v1.08.02
----------------------------------------

- Adds network settings and system shutdown to the
  on-screen console.

----------------------------------------
RESOLVED ISSUES in v1.07.04
----------------------------------------

- Promote v1.07.04 out of beta status.

----------------------------------------
RESOLVED ISSUES in v1.07.04 beta
----------------------------------------

- Adds RestAPI audio source silence information.

----------------------------------------
RESOLVED ISSUES in v1.07.03 beta
----------------------------------------

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- Adds "title" metadata fields.
  and "|" character for separating RTMP artist

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RESOLVED ISSUES in v1.07.02 beta
----------------------------------------

- Increases HE-AAC v2 encoder bitrate upper limit to 64kbps.

----------------------------------------
RESOLVED ISSUES in v1.07.01 beta
----------------------------------------

- Moves Kantar Media Watermarking to a separately
  certified module.

----------------------------------------
RESOLVED ISSUES in v1.06.19 beta
----------------------------------------

- Updates Support Link component.
- Improves Stream Monitor feature.

----------------------------------------
RESOLVED ISSUES in v1.06.18 beta
----------------------------------------

- Modified the Lua time functions to return the results
  as strings, to avoid truncation of UINT64 values.

----------------------------------------
RESOLVED ISSUES in v1.06.17 beta
----------------------------------------

- Fix metadata2 UDP source issue.
- Add time functions to Lua filters and templates.

----------------------------------------
RESOLVED ISSUES in v1.06.16 beta
----------------------------------------

- Fixes silence issues when a program uses both AES input
  and output.

----------------------------------------
RESOLVED ISSUES in v1.06.15 beta
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- Adds support link component (disabled by default).

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RESOLVED ISSUES in v1.06.05 beta
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- Improves resource logging.
- Adds scan detection.

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RESOLVED ISSUES in v1.06.03 beta
----------------------------------------

- Updates Triton Digital Services to SM v4.2.2 MP v4.4.12 and
  ME v4.4.9.
- Adds additional system resource logging.

----------------------------------------
RESOLVED ISSUES in v1.06.02 beta
----------------------------------------

- Adds ability to send RTP streams.

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RESOLVED ISSUES in v1.06.01 beta
----------------------------------------

- Improves RestAPI to prevent Web interface and API lockup.
- Adds automatic memory usage logging if a memory
  leak condition occurs.
- Adds automatic encoder application and system
  performance counter logging.

----------------------------------------
RESOLVED ISSUES in v1.05.07
----------------------------------------

- Enables the TCP/IP KEEP_ALIVE option by default.

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RESOLVED ISSUES in v1.05.06
----------------------------------------

- Fixes missing sensor readings when using the latest
  hardware rev 2.0.

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RESOLVED ISSUES in v1.05.05
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- Updates Triton Digital Services to SM v4.2.1, MP v4.4.6 and
  ME v4.4.7.
- Improves StwCue filter.

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RESOLVED ISSUES in v1.05.03
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- Improves StwCue filter.

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RESOLVED ISSUES in v1.05
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- Adds Kantar Media Watermarking. Only versions certified by Kantar
  Media should be used to run watermarking engine. See a list of
  certified version numbers on the top of this page.

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RESOLVED ISSUES in v1.04.02
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- Added StwCue support to metadata2.

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RESOLVED ISSUES in v1.04.01
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- Fixes license activation not working properly.
- Fixes Metadata2ui template editor not being able to save
  template under certain conditions.
- Metadata2ui can now launch a web browser connection to a
  configured instance.
- Metadata2ui translator instance now shows a vertical scroll
  bar.

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RESOLVED ISSUES in v1.03.23
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- Fixes HLS HTTP POST stream storage on Akamai servers.

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RESOLVED ISSUES in v1.03.22
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- Improves xHE-AAC encoder stream cross-compatibility.

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RESOLVED ISSUES in v1.03.21
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- Fixes incorrect xHE-AAC mime type for HTTP stream server.
- Updates Stream Monitor feature.
- Each audio source license now unlocks two Stream Monitor features.

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RESOLVED ISSUES in v1.03.20
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- Updates Triton Digital Services to SM v4.2.0, MP v4.4.4 and
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  ME v4.4.4.

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RESOLVED ISSUES in v1.03.18
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- Adds HTTP player warning messages when silence or no data
  event occurs.
- Updates frontpanel graphics.

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RESOLVED ISSUES in v1.03.16
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- Adds audio source, streaming and HTTP player error and
  silence detection textual information fields to SNMP.
- Adds Triton Digital Services ME/MP component logs.
- Adds mount point links to Triton Digital Services page.
- Adds harware resource logging every 5 minutes.

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RESOLVED ISSUES in v1.03.12
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- Reduces potential delays coming from the Service Manager
  logging.
- Main metadata logging has been reenabled. Debug diagnostics
  are still turned off.
- Metadata packets are now uniquely numbered, for easier
  tracking.
- By using the "Send to PDM" template and "Receive from PDM"
  filter, we can now measure end-to-end processing delays.

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RESOLVED ISSUES in v1.03.11
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- Turns off metadata info logging to ensure that it does
  not introduce additional delays. All warnings and errors
  are still logged.

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RESOLVED ISSUES in v1.03.10
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- Fixes HTTP codes 201 - 205 not being accepted for
  HLS HTTP POST upload.

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RESOLVED ISSUES in v1.03.09
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- Updates the metadata ingest processor to speed up
  packet parsing.
- Limits the log messages sent to the metadata2ui to
  show only the relevant information.
- Adds "pause on error" option to the logs view in
  metadata2ui.
- Adds addtional packet processing statistics to
  give a better view of where delays may be introduced.

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RESOLVED ISSUES in v1.03.08
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- Adds buffering to metadata2 output so that slow network
  connections do not delay internal processing of other
  packets.
- All metadata2 output sent to a destination is now
  serialized through a single send queue, instead of
  having multiple senders be active at the same time.
- Add metadata2 packet processing timing statistics. A
  histogram of packet processing times is written to the
  logs every hour.
- The metadata2 log output now includes internal message
  timestamps in addition to the timestamps added by the
  service manager.
- The metadata2ui now displays error messages in red and
  warnings in blue. The info messages are still displayed
  as black.
- The autoscroll checkbox in the metadata2ui log view did
  not always work as expected. This has now been resolved.

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RESOLVED ISSUES in v1.03.07
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- Improves metadata2 application stability.
- Adds error messages for events that took longer than
  600 msec to process.
- Fixes issue with metadata2ui where the socket connection
  was kept active until the application was closed.

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RESOLVED ISSUES in v1.03.06
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- Increases HTTP stream player silence detect time to
  10 minutes.
- Allows 0 seconds to be used as Audio Processing program
  sync buffer size.

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RESOLVED ISSUES in v1.03.05
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- Adds HTTP stream player for uploaded stream monitoring.
  Silence detection and connection status information is
  available with Error/Warning messages (forwarded to e-mail,
  if notifications are set) and added SNMP variables.
- Updates Triton Digital services connect component to
  solve resource leakage issues. Fixes station display and
  control issues in web interface.
- Adds Triton Digital services connect component logfiles to
  web interface.
- Fixes scheduler not working properly.
- Reduces xHE-AAC Independend Flag interval to 16 frames.

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RESOLVED ISSUES in v1.02.29 beta
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- Fixes a metadata issue where previous events are not
  cancelled when using a delay block.

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RESOLVED ISSUES in v1.02.28 beta
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- Updates Triton Digital services to v4.1.7.

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RESOLVED ISSUES in v1.02.27 beta
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- Fixes issue where a missing Lua function in metadata2
  filters or templates could cause memory leak over
  longer time period.

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RESOLVED ISSUES in v1.02.26 beta
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- Prevents metadata2ui making changes to factory filters and
  templates.
- Adds Triton mount point information to web interface.
- Imports updated metadata2 filters and templates if the
  secondary bank copy matches the current settings file.
- Adds IPMI IP address to the web interface Status page.

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RESOLVED ISSUES in v1.02.22 beta
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- Updates AudioVault 10.x and AudioVault 9.2 metadata2 filters
  to version 3.
- Adds Center Stage Live (ANDOXML) metadata2 filter.
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RESOLVED ISSUES in v1.02.20 beta
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- Adds temperature, fan speed and power supply status sensor
  readings to web interface Status page.
- Reports sensor readings to SNMP variables, sends SNMP traps
  when readings are in warning state.
- Adds SNMP "value updated" trap to audio sources, streams and
  silence detection failed counters reporting trap.

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RESOLVED ISSUES in v1.02.19 beta
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- Changes name in NfRemote title bar from serial number
  to friendly name set in options.
- Adds Center Stage Live (ANDOXML).lua and andoxml2-picurl.tx2
  filter files.

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RESOLVED ISSUES in v1.02.17 beta
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- Adds processed internal audio sources.

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RESOLVED ISSUES in v1.02.16
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- Adds NS field when setting static IP address using LCD
  front panel.

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RESOLVED ISSUES in v1.02.15
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- Adds friendly device name option (S/N by default).
- Adds Omnia 3 preset download option.
- Adds Triton Digital Services logs and gzip compression
  option.

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RESOLVED ISSUES in v1.02.14
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- Improves web interface international character support.

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RESOLVED ISSUES in v1.02.13
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- Adds SNMP local or any subnet remote client limit option.
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RESOLVED ISSUES in v1.02.12
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- Fixes SNMP not being set up properly by web interface.
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