

Z/IPSTREAM® R/1

Encoder for Streaming Audio



USER'S MANUAL

Z/IPStream R/1 manual v3.0, for Z/IPStream R/1 software v3.0 and later

Applies to: 2001-00254

User Warnings and Cautions

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 autoranging line voltage input. Ensure the power voltage is within the specified range of 100-240VAC. 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: HAZARDOUS VOLTAGES

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 socket 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 on the rear 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.

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.

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Patent information can be found at www.TelosAlliance.com/legal

Updates

Telos Alliance Z/IPStream R/1 features and operations are determined largely by software. The Telos Alliance strives to provide the most stable and feature-rich software available. We encourage you to check for software updates from time to time by visiting our website or by contacting us directly.

Feedback

We welcome feedback on any aspect of our products 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 or suggestions.

We support you...

By Phone/Fax

You may reach our Telos Support Team in emergencies by calling +1 216-622-0247. For billing questions or other non-emergency technical questions, call +1 216-241-7225 between 9:00 AM to 5:00 PM USA Eastern Time, Monday through Friday.

By Email.

Non-emergency technical support is available at Support@TelosAlliance.com.

By Web

The Telos Web site has a variety of information that may be useful for product selection and support. The URL is <https://www.telosalliance.com/Telos>.

SERVICE

You must contact Telos Alliance before returning any equipment for factory service. We will need your unit's serial number, located on the back of the unit. We 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 Alliance Technical Support at +1-216-622-0247. All other customers should contact local representative to make arrangements for service.

Warranty



For the latest Telos Alliance warranty, visit: [telosalliance.com/warranty](https://www.telosalliance.com/warranty)

Register your product

Register your product today to get the full benefits of our warranty, support, and product updates. [telosalliance.com/product-registration/](https://www.telosalliance.com/product-registration/)

The Telos Alliance

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Web: <https://www.telosalliance.com/support-request>

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Creating the Most Exciting and Engaging Audio Experiences Imaginable

Congratulations on your new Telos Alliance product!

The gang here at Telos is committed to shaping the future of audio by delivering innovative, intuitive solutions that inspire our customers to create the most exciting and engaging audio experiences imaginable.

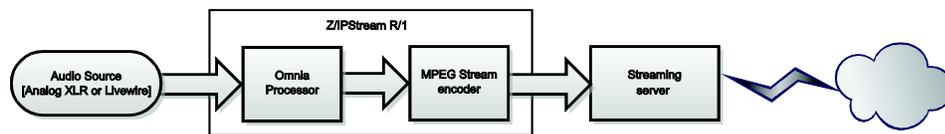
We're grateful that you have chosen audio tools from Telos® Systems, Omnia® Audio, Axia® Audio, Linear Acoustic®, 25-Seven Systems®, and Minnetonka Audio®. We're here to help you make your work truly shine. We hope that you enjoy your Telos Alliance product for many years to come and won't hesitate to let us know if we can help in any way.

The Telos Alliance

1 Quickstart

Streaming with Z/IPStream R/1 involves the following primary tasks:

1. select the audio input,
2. configure the network settings,
3. select an audio processing preset,
4. select the codec used to encode the stream, and
5. select a destination server to distribute the stream to the end-listeners.



This section walks you through a simple setup where the Z/IPStream R/1 is fed analog audio, the network settings are configured via DHCP, and the stream is sent to a ShoutCAST v2 server. If your setup is different, you may safely skip this section. If this is the first time you have used a Z/IPStream R/1, you may to have a look at the [Front Panel] and [Entering Text] sections before you proceed.

Settings

From the front panel of the Z/IPStream R/1, set the following menu options.

Audio



- ◆ Connect an audio source to the XLR inputs and select Audio Input: XLR.



- ◆ Verify Input Gain Setting is 0.0 db.



Audio signal should now be visible on the input metering.



Increase INPUT gain by pressing the encoder knob, turning, and then pressing again to set the value.

Network



The Z/IPStream R/1 offers WAN and LAN/Livewire network ports. In most configurations you only need to connect the WAN port to the network and the LAN port is set as “disabled”. When using Livewire, the LAN port is connected to the Livewire network and the WAN port is connected to the public/outside network.

NOTE:

Do not connect both ports to the same network!

Unless you are using the LAN port with Livewire Audio over IP, configure it as “disabled”.



Connect the WAN Ethernet port to the network. Under Network -> WAN Mode, set the WAN port to Auto (DHCP).

After a reboot, the Network -> WAN IP Address should now show a valid IP address for the connected network.

Omnia Processing



In the Omnia Processing dialog, select a processing preset best suited to the audio source material. If unsure, use **Normalize**. If no audio processing is desired, select the processing preset as **[none]**.

Only processing preset selection is available on the front panel interface. Preset adjustment can be done via Z/IPStream R/1’s internal web configuration by means of a web browser.

Note:

All Z/IPStream R/1 local outputs (XLR, Livewire, and headphone output) are post-process and pre-encode. The post-process audio can be monitored via the output metering of the Status display.



Processing adjustments will be heard real-time on local outputs, but will be heard with a certain amount of delay in the encoded stream. A small portion of this delay is introduced by the encoder while a much larger portion is introduced by buffering in the server and the client used to listen to the stream.

Note: Switch headphone monitoring between pre-processed audio and post-processed audio by pressing the headphone volume knob and selecting “Listen to [input/output] audio.”

Codec 1 & 2



Choices are MP3, AAC, or None. Common output settings are as follows:

MP3

- ◆ Sample Rate: 44100 Hz
- ◆ MP3 Channels: Stereo
- ◆ MP3 Bitrate (16-320 kbps): 128

AAC

Low Complexity

(Good for 3G networks)

- ◆ AAC Format: AAC-LC
- ◆ AAC Channels: Stereo
- ◆ AAC Bitrate (24-320 kbps): 96
- ◆ AAC Transport: ADTS

High Efficiency V2

(Good for Edge/2G networks and lower bit-rates)

Sample Rate: 44100 Hz

- ◆ AAC Format: HE-AAC v2
- ◆ AAC Channels: Stereo
- ◆ AAC Bitrate (14-56 kbps): 48
- ◆ AAC Transport: ADTS

Once the Codec encoders are configured, it should be possible to monitor the encoded audio streams via a connected network audio player. Each codec can be monitored separately via the Z/IPStream R/1's internal Stream Server. In the table below, "ip.address" represents the IP address assigned to the Z/IPStream R/1.

Test Stream Connection URL's

Codec 1	Codec 2
PLS: http://ip.address/play.pls	PLS: http://ip.address/play2.pls
M3U: http://ip.address/play.m3u	M3U: http://ip.address/play2.m3u
ASX: http://ip.address/play.asx	ASX: http://ip.address/play2.asx
RAW: http://ip.address:8000	RAW: http://ip.address:8010

Note:

The Z/IPStream R/1's playlist links are for testing only and should NEVER be deployed as public connection points. Always use a streaming server.

Stream Configuration

Stream configurations are provided by the stream relay provider or set in the stream server configuration files. Z/IPStream R/1 has four (4) configurable stream uplinks which can be set to push audio from either of the two (2) available encoded codecs.

SHOUTcast DNAS v2.0 Example

The free SHOUTcast DNAS software is a good example of a streaming server. SHOUTcast DNAS is available at <http://www.shoutcast.com/broadcast-tools>. A single SHOUTcast instance is capable of relaying both Z/IPStream R/1 codecs, and will require the configuration of 2 of the available 4 stream configurations.

SHOUTcast configuration file

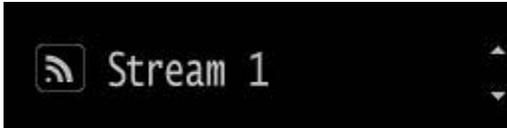
SHOUTcast operates by means of specifying a configuration file at launch, typically, `sc_serv.conf`. The following parameters are needed from the SHOUTcast configuration file.

```
portbase=8010
password=streampwd
adminpassword=adminpwd
```

At least two stream ID's should be specified.

```
streamid_1=1
streamid_2=2
```

Stream 1



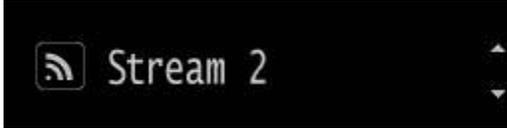
- ◆ Stream Type: SHOUTcast
- ◆ Codec: Codec 1
- ◆ Server Address: *SHOUTcast.ip:{portbase}* [example: 192.168.2.200:8010]
- ◆ Server Password: *{password}* [example: streampwd]
- ◆ Stream Name: *Name of Stream* [sent to clients]
- ◆ Stream Genre: *Genre of Stream* [[SHOUTcast guidelines](#)]
- ◆ Stream URL: URL of Station Website [sent to clients, example: <http://TelosAlliance.com>]
- ◆ Server version: **version 2**
- ◆ Stream ID: 1
- ◆ Stream Availability: **public**

 A screenshot of a web browser window showing the "SHOUTcast DNAS Status" page. The browser's address bar shows "http://192.168.2.200:8010/index.html?sid=1". The page title is "SHOUTcast DNAS Status" in large yellow letters. Below the title is a navigation menu with links for "Status", "Song History", "Listen", "Stream URL", and "Admin Login". The "Listen" link is circled in red. To the right of the navigation menu, it says "SHOUTcast Server v2.0.0.29/posix(linux x86)". Below the navigation menu is a section titled "Current Stream Information" with a dark background. This section contains the following information:

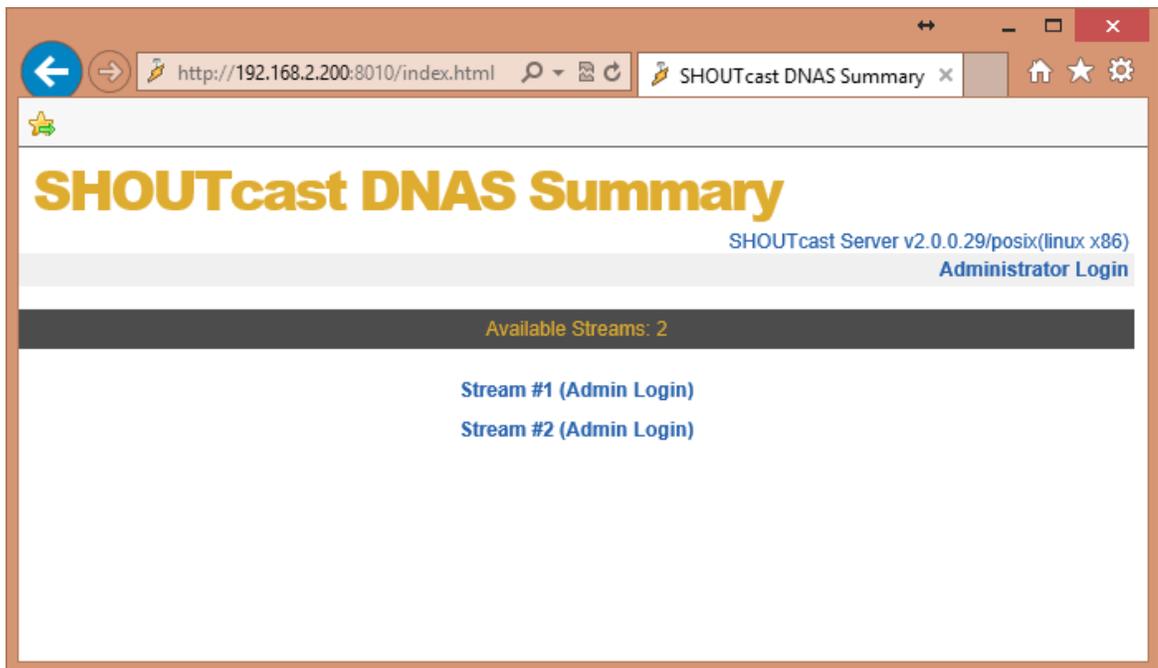
Server Status:	Server is currently up and public (no YP connection)
Stream Status:	Stream is up at 96 kbps with 0 of 128 listeners (0 unique)
Listener Peak:	0
Stream Name:	Name of Stream
Content Type:	audio/aac
Stream Genre:	livewire
Stream URL:	http://TelosAlliance.com

The Listen link now provides a stream listen URL for client deployment.

Stream 2



- ◆ Stream Type: SHOUTcast
- ◆ Codec: Codec 2
- ◆ Server Address: *SHOUTcast.ip*:{portbase} [example: 192.168.2.200:8010]
- ◆ Server Password: { password } [example: streampwd]
- ◆ Stream Name: *Name of Stream* [sent to clients]
- ◆ Stream Genre: *Genre of Stream* [[SHOUTcast guidelines](#)]
- ◆ Stream URL: URL of Station Website [sent to clients, example: <http://TelosAlliance.com>]
- ◆ Server version: *version 2*
- ◆ Stream ID: 2
- ◆ Stream Availability: *public*



Two streams are now available from the SHOUTcast DNAS server.

ICEcast Example

Icecast is a free stream relay software available from <http://www.icecast.org/>.

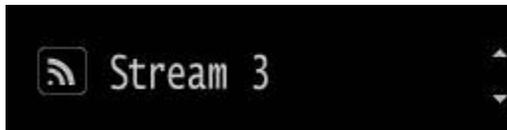
ICEcast Config file

ICEcast must be launched with a configuration file specification, typically, `icecast.xml`. The following parameters are required from the ICEcast configuration file.

```
<authentication>
  <source-password>streampwd</source-password>
  <relay-password>relaypwd</relay-password>
  <admin-user>admin</admin-user>
  <admin-password>adminpwd</admin-password>
</authentication>
<listen-socket>
  <port>8000</port>
</listen-socket>
```

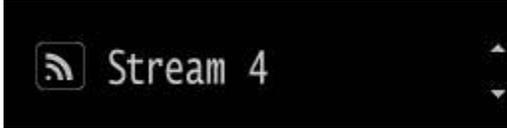
ICEcast per-stream configuration is un-needed. Streams are distinguished by the mount point configuration which is specified in the source device [Z/IPStream R/1].

Stream 3



- ◆ Stream Type: ICEcast
- ◆ Codec: Codec 1
- ◆ Server Address: *ICEcast.ip:{port}* [example: 192.168.2.200:8000]
- ◆ Mount Point: *hifi* [can be any URL compatible string]
- ◆ Server Username: *source* [can be overridden in icecast.xml config]
- ◆ Server Password: `<source-password>` [example: streampwd]
- ◆ Metadata Username: `<admin-user>` [example: admin]
- ◆ Metadata Password: `<admin-password>` [example: adminpwd]
- ◆ Stream Name: *Z/IPStream R/1 HiFi* [sent to clients]
- ◆ Stream Genre: *livewire*
- ◆ Stream URL: URL of Station Website [sent to clients, example: <http://TelosAlliance.com>]
- ◆ Server version: *version 2*
- ◆ Stream Availability: *public*

Stream 4



- ◆ Stream Type: **ICEcast**
- ◆ Codec: **Codec 2**
- ◆ Server Address: *ICEcast.ip:{port}* [example: 192.168.2.200:8000]
- ◆ Mount Point: **lofi** [can be any URL compatible string]
- ◆ Server Username: **source** [can be overridden in icecast.xml config]
- ◆ Server Password: **<source-password>** [example: streampwd]
- ◆ Metadata Username: **<admin-user>** [example: admin]
- ◆ Metadata Password: **<admin-password>** [example: adminpwd]
- ◆ Stream Name: *Z/IPStream R/1 LoFi* [sent to clients]
- ◆ Stream Genre: *livewire*
- ◆ Stream URL: URL of Station Website [sent to clients, example: <http://TelosAlliance.com>]
- ◆ Server version: **version 2**
- ◆ Stream Availability: **public**

Icecast2 Status

Administration Server Status Version

Mount Point /hifi

Stream Title: ProSTREAM HiFi
 Stream Description: ProSTREAM HiFi
 Content Type: audio/aacp
 Mount started: Sat, 04 May 2013 20:08:43 -0500
 Bitrate: 96
 Current Listeners: 0
 Peak Listeners: 0
 Stream Genre: livewire
 Stream URL: <http://TelosAlliance.com>
 Current Song:

Mount Point /lofi

Stream Title: ProSTREAM LoFi
 Stream Description: ProSTREAM LoFi
 Content Type: audio/aacp
 Mount started: Sat, 04 May 2013 20:08:48 -0500
 Bitrate: 48
 Current Listeners: 0
 Peak Listeners: 0
 Stream Genre: livewire
 Stream URL: <http://TelosAlliance.com>
 Current Song:

2

Introducing Z/IPStream R/1

The Telos Alliance Z/IPStream R/1 is a hardware-based audio processor and web stream encoder device in a compact 1U rack-mount chassis. The Z/IPStream R/1 will process and encode audio sourced from either rear-mounted analog audio inputs or via Livewire audio over IP technology¹.

Z/IPStream R/1 uses Omnia Audio's proven audio processing algorithms² to balance the input audio according to the program's format. The processed audio is then encoded by two distinct encoders. Z/IPStream R/1 can send the encoded streams to up to four streaming servers or streaming service providers. The processed audio is also sent to XLR and Livewire outputs for monitoring.

The Z/IPStream R/1 can also accept and parse incoming metadata. The metadata is then sent along with the stream to the streaming server, which will then distribute it to end listeners. It is important to note that listeners always connect to the streaming server, never directly to Z/IPStream R/1.

FYI: MPEG License & Music Copyright

Musical compositions and sound recordings are creative works that are protected by the copyright laws of the United States (title 17, U.S. Code) and other countries. Under U.S. law, the owner of a copyright has the exclusive right (and may authorize others) to reproduce the work, use parts of the work in a new creation, distribute the work in whole or in part, and to publicly display or perform the work (including on web pages and through webcasting). With few exceptions, it is illegal to reproduce, distribute or broadcast a sound recording without the permission of the copyright owner. It is your responsibility to comply with the copyright laws of the United States and other countries in which you broadcast and to pay all applicable royalties to the copyright owners when you become a webcaster.

There have been recent amendments to the copyright law regarding webcasting of sound recordings. These new provisions allow webcasting under the terms of a statutory license, as a way to help webcasters get permission without having to go to each sound recording's owner. The statutory license, however, has strict requirements that you must follow. Some of these requirements include the payment of license fees, limitations on the number of songs from the same album or artist that may be played in a three hour period (called the sound recording performance complement); a prohibition on publishing advance playlists; and a requirement to identify the song, artist and album on the website. There are other requirements as well. The Recording Industry Association of America provides quite a bit of information on copyright law as it applies to webcasting, and both ASCAP and BMI have created license agreements that they are willing to grant to webcasters that they believe conform to the provisions of the new copyright rules for webcasting.

For additional information on the statutory license and other aspects of webcasting, please visit the following sites:

The U.S. Copyright Office <http://www.copyright.gov>

The Recording Industry Association of America <http://www.riaa.com/issues/music/webcasting>

ASCAP <http://www.ascap.com/weblicense/webintro.html>

BMI <http://www.bmi.com/iama/webcaster/index.asp> Streaming Server software

1 For more information on Livewire, please visit www.TelosAlliance.com/livewire.

2 For more information regarding Omnia processors, please visit www.TelosAlliance.com/Omnia.

3 Controls and Connections

Overview

The Z/IPStream R/1 is very simple to set up and use, having a minimal amount of controls and an easy, intuitive interface. Here, we'll take a look at the front and rear panels of your Z/IPStream R/1.

The packing box contains:

- ◆ Z/IPStream R/1 unit
- ◆ AC power cords (Euro 220v, US 110v, unterminated)
- ◆ RJ45 network cable
- ◆ #10 rack screws
- ◆ warranty registration form
- ◆ warranty information sheet

Front Panel



1. **Power LED:** The blue power LED is located to the upper left of the display. When the unit is first powered, the power LED will flash until the unit is fully booted. When booted, the power LED will light solid.
2. **Display Screen:** The OLED display screen displays audio meters and configuration information. Use the display, control knob, and return button to navigate the Z/IPStream R/1's options and settings.
3. **Control Knob:** The main control knob, next to the display, works as a selector. Rotate the knob until the desired option is displayed, then press the control knob to select the option. The control knob is also used in similar fashion for character selection in text entry fields.
4. **Back Button:** The back (or escape) button is adjacent to the control knob. Pressing it steps back, or exits, from a selected option or dialog. Pressing it repeatedly, will return you to the main screen (and will also toggle between the main and status screens).
5. **Headphone Jack:** 1/4" Stereo TRS Jack for monitoring the input and post-process output audio.
6. **Headphone Volume Knob:** The headphone volume knob controls the output level of the audio being sent to the headphone jack. Pressing the headphone volume knob will toggle the headphone output between monitoring the input audio and the post-process output audio.

Analog Audio Hardware Back Panel



1. **AC Mains connection:** 110 - 220v auto-switching power connection. Use supplied power cord appropriate for your area.
2. **LAN Network:** The LAN network jack is a 10/100 RJ45 jack and can be used with a Livewire network for source audio and post-process audio monitoring.
3. **WAN Network:** The WAN network jack is a 10/100 RJ45 jack typically used to send the encoded streams to streaming servers. **NOTE:** Z/IPStream R/1 provides two network interfaces for configuration flexibility. If you are not using Livewire, then you only need to configure one interface. Make sure to leave the other interface as “disabled.” Do not connect both interfaces to the same network!
4. **Analog Audio Inputs:** (female 3-pin XLR, left & right)
5. **Analog Audio Outputs:** (male 3-pin XLR, left & right)
6. **USB:** RS232 metadata input (with optional USB-to-Serial Adapter P/N 2091-00140)
7. **Parallel Interface:** no current functions

Digital Audio Hardware Back Panel



1. **AC Mains connection:** 110 - 220v auto-switching power connection. Use supplied power cord appropriate for your area.
2. **LAN Network:** The LAN network jack is a 10/100 RJ45 jack and can be used with a Livewire network for source audio and post-process audio monitoring.
3. **WAN Network:** The WAN network jack is a 10/100 RJ45 jack typically used to send the encoded streams to streaming servers. **NOTE:** Z/IPStream R/1 provides two network interfaces for configuration flexibility. If you are not using Livewire, then you only need to configure one interface. Make sure to leave the other interface as “disabled.” Do not connect both interfaces to the same network!
4. **Analog Audio Inputs:** (female 3-pin XLR, left & right)
5. **Analog Audio Outputs:** (male 3-pin XLR, left & right)
6. **Digital Audio:** Input (female 3-pin XLR) / Output (male 3-pin XLR)
7. **USB:** RS232 metadata input (with optional USB-to-Serial Adapter P/N 2091-00140)
8. **Parallel Interface:** no current functions

Status Display



The Status screen shows the audio processing meters and indicators. Z/IPStream R/1's audio processor is capable of performing many different processing functions at different times, primarily based on dynamic range differences in the source audio.

The meters analyze the signals and aid in adjusting the specific parameters needed to achieve desired aural results. Although the processing displays are capable of providing a wide range of information, we recommended setting the audio processing based on how it sounds best to you, not relying entirely on meter indications.

IN/OUT

The Status 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 which 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.

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 and that there is nothing more in the level department, except to create distortion.

Automatic Gain Control (AGC)

There are two sections to the AGC bar graphs display: wide-band automatic gain control (WB-AGC) and multi-band automatic gain controls. The multiband AGC bands are low frequency, mid frequency, and high frequency. The bar shows the average value of the gain reduction, while the floating pill indicates the peak value of gain reduction.

The Omnia processor 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 bar graphs will have a dynamic “bounce.” Every now and then, the processor will react quickly and show a larger amount of gain reduction. This action will recover very slowly, and return to rest with the main bar graph. This action is more apparent on dynamically textured audio.

Another feature unique to Omnia processing is processor hold. During brief pauses in audio, the bar graphs will “freeze” indicating the processor is in the hold mode of the algorithm. This can occur when there is dry vocal audio in the signal.

Bar graphs will indicate differently for pre-processed audio than with widely-varying audio level content. Audio signals which lack dynamic range, whether naturally or by prior processing, will possess a lower peak to average ratio. Conversely, audio with wide dynamic range possesses a higher peak to average ratio. Z/IPStream R/1 adapts to each case.

Low dynamic range audio (audio with high RMS and low peak levels) will show more activity in the WB-AGC bar graph and less activity in the multiband bar graphs. The WB-AGC responds to high RMS energy while the multiband section reacts to lower peak energy. Multiband bar graphs may not indicate any action at all with some low dynamic range audio which is normal for Omnia processing. This may occur with heavily processed commercials or music, or with music passages of sustained level.

Wide dynamic range audio (audio with low RMS and high peak levels) causes the opposite to occur. The multiband bar graphs become active, while the WB-AGC section appears to not respond as much. The multiband AGC’s can work aggressively with wide dynamic range audio while the WB-AGC section indicates little activity.

The WB-AGC section is designed to operate much slower than the multiband gain controls, primarily because of the nature of each function. The WB-AGC operates on the audio’s RMS energy. During gain calculations, the incoming audio’s “average” level is established, and gain adjustments, if needed, are made based on those calculations. The WB-AGC bar graph will appear to move slower as it makes changes over relatively long time periods.

The intent of the multiband gain control 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 but not interfere with the sonic integrity of the audio signal. For this reason, the multiband gain controls operate faster, with special background instructions to govern their behavior, and strictly on an as-needed basis.

Limit

The limit bar graphs monitor the level to which the processor is engaging gain reduction in the post-process audio for each the left and right channels.

Entering and Editing Text in the Front Panel UI

Since the Z/IPStream R/1's user interface consists of a selector knob and back button, the method for entering text or numeric information may not be immediately obvious. This section describes how text entry and editing is accomplished. The task seems complex at first, especially when described in words, but it is very simple once you have a chance to practice it a bit; similar to learning to tie one's shoe laces.

When a text field is displayed, the pencil icon  indicates that the field is changeable. Press the main knob to enter one of the edit modes. The edit mode is indicated by the cursor shape: a solid cursor indicates text entry mode, while a rectangle outline cursor indicates erase mode.

text entry mode (solid cursor)



In text entry mode, turning the knob cycles through all available characters. Press the main knob to select the current character; the cursor will move to the next character position. While in text entry mode, pressing the Back button once switches the cursor to erase mode.

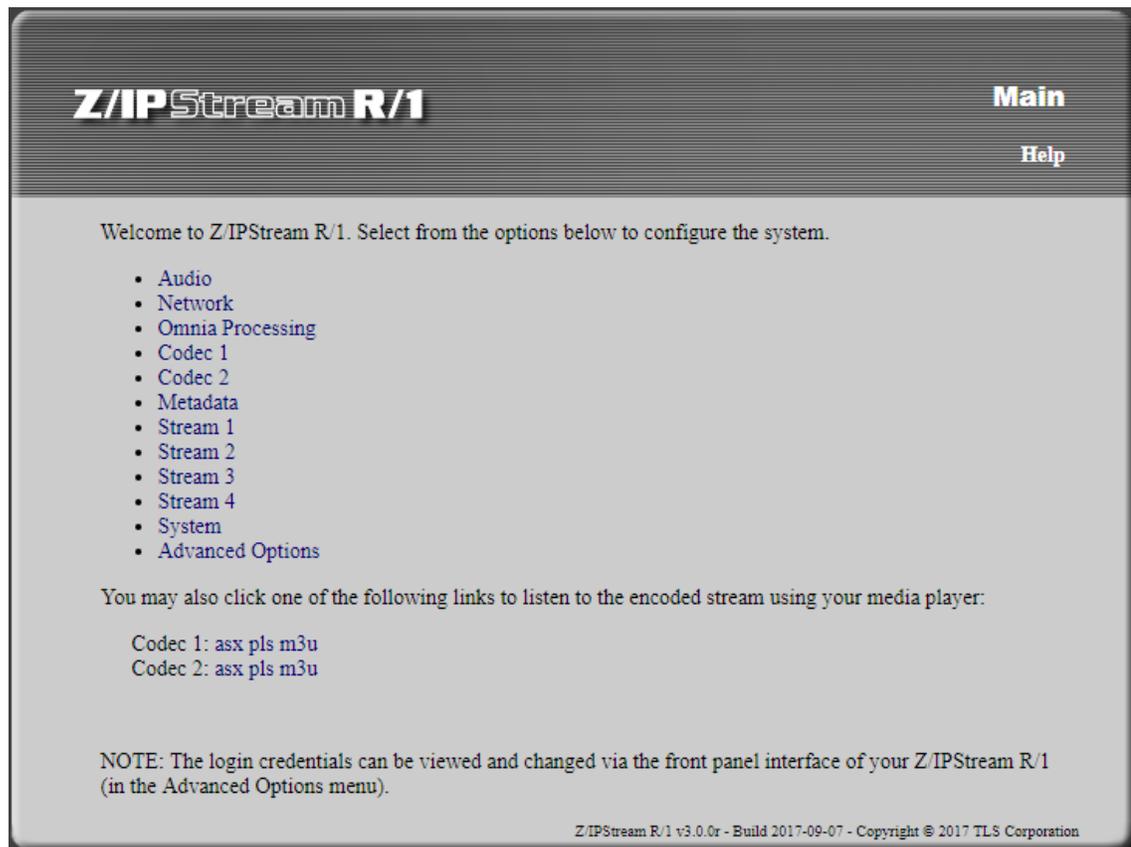
erase mode (hollow rectangle cursor)



In erase mode, turning the knob erases (or restores) characters. While in erase mode, press the main knob to return to text entry mode or press the Back button to exit field editing altogether.

Web Configuration

The Z/IPStream R/1 has an internal web server accessible via a web browser. The web interface can be reached by pointing a web browser to the IP address of either LAN or WAN network, for example <http://192.168.1.15/>.



Default Credentials

The default username/password to access the web interface is user without password and admin/Telos.

NOTE:

For security reasons, it is very important to change the default login credentials. Failure to do so may give others access to your Z/IPStream R/1. The login credentials can be changed on the front panel, under the System dialog.

NOTE:

The Z/IPStream R/1 web configuration requires Javascript and HTML5 capable web browser (for real-time Omnia processor adjustment). Some settings and values may be locked or unavailable if they depend on other configured options.

4 Z/IPStream R/1 Configuration

Audio Menu

Audio I/O interface

XLR (analog)

Z/IPStream R/1 **Audio**
Main

Select the audio I/O parameters below. Click the save button when done.

Audio I/O interface:

Input audio gain: (0 to 22.5 dB)

Output audio gain: (-95.2 to 0 dB)

Output Livewire channel (source):

NOTE: The rear panel audio output plays the Omnia-processed audio. The headphone output plays either the source audio or the processed audio (push the volume knob to toggle).

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Input audio gain

[0.0 dB - 22.5 dB] The Audio Input Level is adjustable from a 0dB level of gain to 22.5 dB level of gain. Gain level is not necessary as +4v balanced audio will show input status at a setting of 0db.

Output audio gain

[-95.2 dB - 0.0 dB] The Audio Output Level affects the signal level for the Z/IPStream R/1's rear outputs and Livewire returns. The default setting of 0.0 dB will direct the full level of the processed audio to the Z/IPStream R/1 outputs.

Output Livewire channel (source):

Enter a unique channel number to appear on the Livewire audio network (connected to the Z/IPStream R/1 LAN/Livewire network jack). Enter 0 to disable Livewire output.

Note:

The post-process audio to the encoder is unaffected by the Audio output Level setting internally.

Livewire

Z/IPStream R/1 **Audio**
Main

Select the audio I/O parameters below. Click the save button when done.

Audio I/O interface:

Warning: Uncompressed audio generates a lot of network traffic. Do not use Livewire unless you have a properly configured network switch!

Input Livewire channel (destination):

Output Livewire channel (source):

NOTE: The rear panel audio output plays the Omnia-processed audio. The headphone output plays either the source audio or the processed audio (push the volume knob to toggle).

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Livewire is an audio over IP technology from Axia Audio. Livewire audio requires that specific conditions be met in the network switching environment to use Livewire functionality. As a multicast IP protocol requiring both IGMP routing and QoS parameters, common consumer network routers and switches are unable to use Livewire IP technology. For more information, please visit www.TelosAlliance.com/livewire.

Livewire connections should utilize the LAN port and typically imply a manual IP configuration.

Input Livewire channel (destination):

Enter the channel number of an active Livewire IP audio source channel generated by another Livewire device.

Output Livewire channel (source):

Enter a unique channel number to appear on the Livewire audio network (connected to the Z/IPStream R/1 LAN/Livewire network jack) . Enter 0 to disable Livewire output.

AES (digital)

Z/IPStream R/1 **Audio**
Main

Select the audio I/O parameters below. Click the save button when done.

Audio I/O interface:

XLR output audio gain: (-95.2 to 0 dB)

Output Livewire channel (source):

NOTE: The rear panel audio output plays the Omnia-processed audio. The headphone output plays either the source audio or the processed audio (push the volume knob to toggle).

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Output audio gain

[-95.2 dB - 0.0 dB] The Audio Output Level affects the signal level for the Z/IPStream R/1's rear outputs and Livewire returns. The default setting of 0.0 dB will direct the full level of the processed audio to the Z/IPStream R/1 outputs.

Output Livewire channel (source):

Enter a unique channel number to appear on the Livewire audio network (connected to the Z/IPStream R/1 LAN/Livewire network jack) . Enter 0 to disable Livewire output.

Network Menu

Z/IPStream R/1 **Network**
Main

Configure the network interfaces below then click the Save button when done.

LAN Interface

Mode:

IP Address:

Netmask:

Gateway:

WAN Interface

Mode:

IP Address:

Netmask:

Gateway:

DNS Servers

DNS 1:

DNS 2:

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The Z/IPStream R/1 is equipped with two RJ45 gigabit network ports marked LAN [Local Area Network] and WAN [Wide Area Network]. The networking settings of the Z/IPStream R/1 are of IPv4 conventions and must be assigned an IP address before the Z/IPStream R/1 will communicate with any other devices.

LAN Interface

Mode

(Auto) DHCP

IP address is provided by a DHCP server on the network.

Note: Livewire network devices are typically configured with a Manual IP address, not DHCP.

MANUAL

A standard IPV4 IP address must be specified.

DISABLED

Disable the LAN port.

IP Address

(Auto) DHCP

The IP Address field will show current IP address, or 0.0.0.0 if no address is assigned.

MANUAL

A valid IPV4 IP address must be specified.

Netmask

(Auto) DHCP

The Netmask field will show the value assigned by the DHCP server.

MANUAL

A valid IPV4 netmask must be specified.

Gateway

(Auto) DHCP

The Gateway field will show the value assigned by the DHCP server.

MANUAL

Specifying a gateway is optional. If the WAN port is in use for connection to the Internet, set the LAN gateway to 0.0.0.0. If the WAN port is disabled, specify the IP address of the network gateway router.

Note:

Never fill both LAN and WAN gateways. That way the network won't be able to operate properly unless static routes are added in Advanced Options / Configure static routes page.

WAN Interface

Mode

(Auto) DHCP

IP address is provided by a DHCP elsewhere on the network. This is the default configuration when connecting to a network routed by a typical consumer-brand gateway router device for Internet access.

MANUAL

A standard IPV4 IP address must be specified.

DISABLED

Disable the WAN port.

IP Address

(Auto) DHCP

The IP Address field will show current IP address, or 0.0.0.0 if no address is assigned.

MANUAL

A valid IPV4 IP address must be specified.

Netmask

(Auto) DHCP

The Netmask field will show the value assigned by the DHCP server.

MANUAL

A valid IPV4 netmask must be specified.

Gateway

(Auto) DHCP

The Gateway field will show the value assigned by the DHCP server.

MANUAL

Specifying a gateway is optional. If the WAN port is in use for connection to the Internet, specify the IP address of the network gateway router.

Note:

Never fill both LAN and WAN gateways. That way the network won't be able to operate properly unless static routes are added in Advanced Options / Configure static routes page.

DNS Servers

DNS server entry is not required, but recommended. DNS is what allows a named address (google.com) to refer to a numerical address (74.125.255.134). If local DNS servers are unknown or unavailable, Google's public DNS server can be used with the IP addresses 8.8.8.8 & 8.8.4.4.

Codec [1 & 2]

Z/IPStream R/1 **Codec 1**

Main

Select the codec parameters below then click Save when done.

Codec:

Sample rate:

Warning: RTMP streams require 44100 Hz sample rate!

Channels:

Bitrate: (24 to 320 kbps)

Transport: (if not sure, select ADTS)

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The codec represents the manner in which the 1.4Mbps stream of un-compressed audio data will be converted into a smaller data stream by means of digital compression. Z/IPStream R/1 applies lossy compression algorithms¹ to the processed audio in order to create a live data stream for end users via the Internet. A lossy compression technique produces a data stream small enough to be delivered to the client via DSL, EDGE wireless, or any common Internet network infrastructure.

MPEG has released different standards as technology has required, and each successive standard has built upon the previous yielding modern codecs which can deliver extremely clear audio at incomprehensibly high ratios of compression.

¹ The Telos Alliance licenses Motion Picture Experts Group [MPEG] compression codecs from Fraunhofer IIS [<http://www.iis.fraunhofer.de/en/bf/amm/produkte/audiocodec/>] and includes the codecs as a source component of Z/IPStream R/1. Lossless encoding, like a ZIP or DMG file on a computer allow an end user to un-compress the data and restore all of the original information. In order to gain much greater compression ratios, audio compression employs lossy compression or encoding by making certain assumptions about human hearing. For instance, a C# note played by a guitar at 85 db of power overpowers a C harmonic of a keyboard at 30 db of power, so the 30db signal is minimized or tossed.

Lossy compression relies upon subjective process, thus there are many techniques to digitally compress audio and there are many more opinions as to which are best. In anticipation of the need for compression algorithms, the Motion Picture Experts Group (MPEG) [<http://mpeg.chiariglione.org/>] was formed by the International Standards Organization (ISO) [<http://www.iso.org/>] and the International Electromechanical Commission (IEC) [<http://www.iec.ch/>] to address the needs of digital compression. Their research has yielded the codecs licensed and used by Z/IPStream R/1.

When selecting a codec, there is a trade-off between more modern compression standards which sound much better than older standards, but may risk incompatibility with older client software and devices. The core of this choice will likely be between .MP3 (MPEG Audio Layer III) and the newer and aptly named Advanced Audio Coding (AAC) (MPEG-4 with MPEG-2 components).

Many factors may be considered when choosing a codec to employ for the output stream.

- ◆ Is content mostly vocal or musical?
- ◆ How is the processor tuned?
- ◆ What types of clients are connecting to the audio stream?

Compression Codec

It is generally accepted that AAC performs better and is the codec of choice. It is rare to find clients which are MP3 only a decade after the AAC codecs became standard, but they do still exist. MP3 was released in 1993 and as the first digital compression technique to find widespread acceptance is in most all digital stream clients that exist.

AAC benefited greatly from being the second go at audio compression as it was able to be designed with all that was learned with MP3. AAC also boasts multiple “profiles” to fit the need of the transmission.

MP3 “The standard for digital audio”



MP3 is an all encompassing audio compression codec. MP3 is the safest choice for decoding compatibility, but creates the least clear audio of all codecs available.

AAC-LC “Low Complexity”



Formerly known as aacPlus or AAC+, Advanced Audio Coding’s Low Complexity codec is a high performance codec for excellent audio quality at standard bitrates. AAC-LC can be found in widespread use, most notably in Apple’s iTunes. When compared side by side with MP3, the superior quality of audio encoded with AAC-LC becomes much more apparent at lower bitrates.

HE-AAC “High Efficiency”



High Efficiency Advanced Audio Coding is a newer AAC codec and incorporates Spectral Band Replication (SBR) bandwidth expansion to improve audio at very low bitrates.

HE-AAC v2 applies a Parametric Stereo feature to HE-AAC codec allowing for even further reduction in bandwidth.

Channels

Choose from mono or stereo encoding.

Bitrate

The full spectrum of Z/IPStream R/1 encoding can be anywhere from 14-320 kbps for the output data stream bitrate. The output bitrate constraints will differ depending on which codec is selected and whether the encode is set to mono or stereo. The range of each codec is shown in the table below:

Codec	Bitrate Range
MP3	16 to 320 kbps
AAC-LC	24 to 320 kbps
HE-AAC	24-96 kbps
HE-AAC v2	14-56 kbps

Transport (AAC only)

AAC audio encoding does not specify a container form for packaging the data stream. The Transport option allows you to select one of four available formats: ADTS, ADTS-CRC, ADIF, or RAW. If you're not sure about this setting, just keep the default value (ADTS) or select a transport format that is understood by the audio player you are targeting (some testing may be required).

ADTS

The default transport container to use for all AAC codecs is Audio Data Transport Stream (ADTS). ADTS encapsulates the packetized elementary streams of encoded AAC audio and metadata for transport over IP networks.

ADTS-CRC

ADTS-CRC includes CRC error correction and stream synchronization features to the ADTS transport and can be used for maintaining transmission integrity when the signal is degraded.

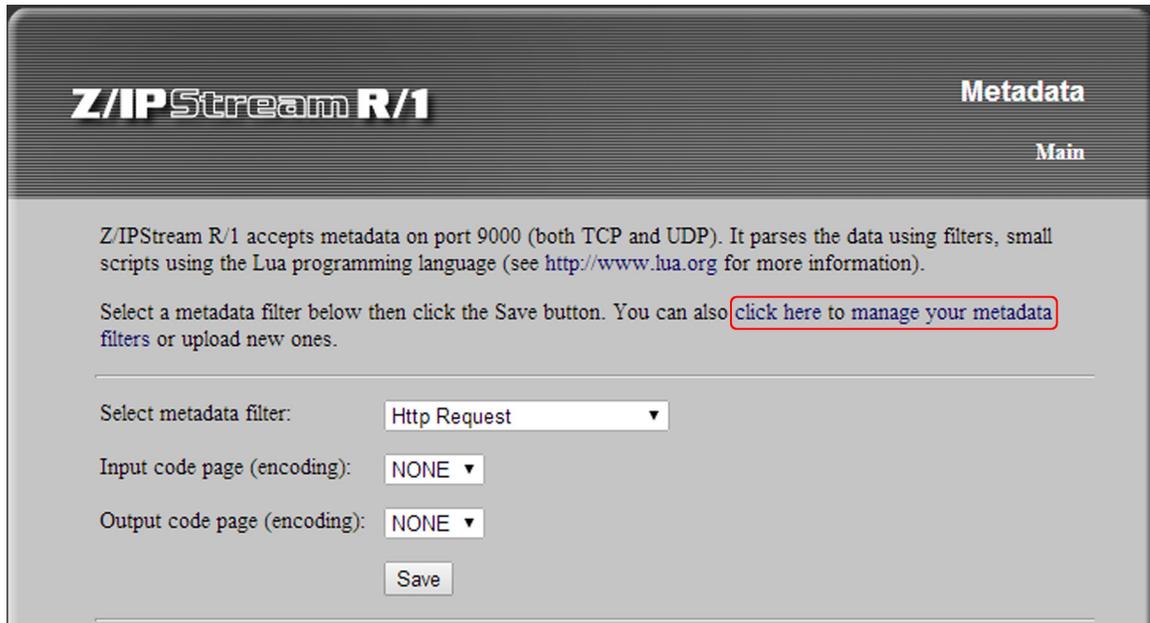
ADIF

Audio Data Interchange Format (ADIF) is a transport container more suited to files than streaming as the data stream is preceded with a header packet and then raw AAC data follows. ADIF may be used in applications where the audio will be muxed with other audio channels for multi-channel transport, but is not likely to be used in traditional streaming.

RAW

Raw mode will release the AAC data stream without any transport container at all and consists of nothing more than the encoded audio data.

Metadata



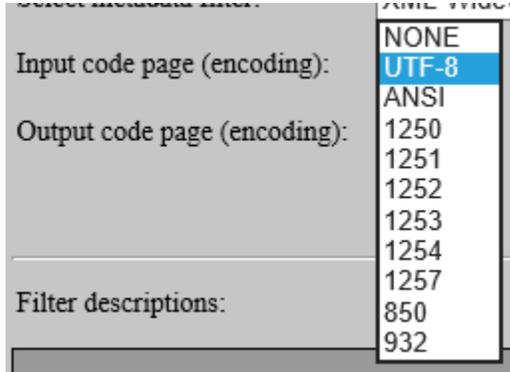
The screenshot shows the 'Metadata' configuration page for Z/IPStream R/1. The page has a dark header with the logo 'Z/IPStream R/1' on the left and the title 'Metadata' on the right. Below the title is a 'Main' link. The main content area is light gray and contains the following text: 'Z/IPStream R/1 accepts metadata on port 9000 (both TCP and UDP). It parses the data using filters, small scripts using the Lua programming language (see <http://www.lua.org> for more information). Select a metadata filter below then click the Save button. You can also [click here to manage your metadata filters](#) or upload new ones.' Below this text are three dropdown menus: 'Select metadata filter:' with 'Http Request' selected, 'Input code page (encoding):' with 'NONE' selected, and 'Output code page (encoding):' with 'NONE' selected. At the bottom of these fields is a 'Save' button.

Metadata, or “data about the data,” can be combined with the audio stream for delivery to end clients. The metadata includes textual information about the current content of the audio stream, such as track title and artist, current program titles, or branding and can also include a URL to associate with the stream.

Most end clients which are used to play the streaming audio contain displays which can be used to present the text information in real-time. In the event that no metadata is provided to Z/IPStream R/1, the stream’s “Stream Name” configuration will be included as the stream’s text content.

Data sent to the Z/IPStream R/1 is parsed internally by “filters”, which are short scripts using the Lua programming language (see <http://www.lua.org/>). Lua scripts can be modified and uploaded to Z/IPStream R/1, to ingest incoming data and pass along the desired pieces (IE, “Track - Artist”) for real-time presentation within the playback client. Z/IPStream R/1 ships with a set of metadata filters and you may be able to feed it with data that matches one of the formats it understands. If your metadata source is not able to send data in one of the supported formats, let us know and we’ll be able to help.

Z/IPStream R/1 is currently able to support incoming data formatted with the following code pages:



(http://en.wikipedia.org/wiki/Code_page):

- ◆ UTF-8 character encoding [default]
- ◆ ANSI - Windows
- ◆ 1250 – Central and East European Latin
- ◆ 1251 – Cyrillic
- ◆ 1252 – West European Latin
- ◆ 1253 – Greek
- ◆ 1254 – Turkish
- ◆ 1257 – Baltic
- ◆ 850 – “Multilingual (Latin-1)” (Western European languages)
- ◆ 932 - Japanese (DBCS)

Input

Automated playback systems typically provide for submitting metadata to a streaming server. Metadata connections are made to Z/IPStream R/1 via TCP, UDP connections to port 9000 or RS232 serial connections with an RS232 to USB adapter².



If no automated data stream is available, a manual entry form is included in Z/IPStream R/1 to allow manual entry of playlist information.

² Connecting RS-232 serial data requires the use of a USB to RS-232 adapter manufactured by SYBA Multimedia. Fully tested and supported for use with the Z/IPStream R/1 (Order Telos part number 2091-00140).

Process

The most generic method of metadata parse is provided via the Line Parser Sample metadata filter.

Select metadata filter: Line Parser Sample

Input code page (encoding): UTF-8

Output code page (encoding): UTF-8

Save

As stated in the metadata table in the metadata webpage:

Line Parser Sample

This file implements a metadata filter that will parse strings that look like this: “t=title...lu=url...\n” where \n stands for a newline character.

If the playback system has mutable output to TCP, the output string may be able to be constructed for the Z/IPStream R/1. If so, the key parameters would resemble the following:

Playlist Data Output

- ◆ IP address: 192.168.2.109 [IP of Z/IPStream R/1]
- ◆ metadata port: 9000
- ◆ string output: t={title} - {artist} | u=http://mystation.org \n

The automation system would then submit that string of characters via TCP to 192.168.2.109:9000 where the Line Parser Sample.Lua filter extrapolates the two variables from the input string and passes them to the encoder. The Line Parser Sample filter will also log both successes and failures in the Z/IPStream R/1 system log.

Note:

The Lua Line Parser requires that TCP strings be terminated with a line feed [\n] character. This is the indication the line filter uses to start processing the data. Character string parsers (often used with UDP data sources) do not share this requirement.

SHOUTcast Stream Information

Current Stream Information	
Server Status:	Server is currently up and public (no YP connection)
Stream Status:	Stream is up at 96 kbps with 0 of 128 listeners (0 unique)
Listener Peak:	0
Stream Name:	ProSTREAM HiFi
Content Type:	audio/aac
Stream Genre:	livewire
Stream URL:	http://TelosAlliance.com
Current Song:	The Rolling Stones - Brown Sugar

Icecast Stream Information

Mount Point /hifi  

Stream Title: ProSTREAM HiFi
Stream Description: ProSTREAM HiFi
Content Type: audio/aacp
Mount started: Sat, 04 May 2013 20:27:35 -0500
Bitrate: 96
Current Listeners: 0
Peak Listeners: 0
Stream Genre: livewire
Stream URL: <http://192.168.2.200:8000/hifi>
Current Song: The Rolling Stones - Brown Sugar

Connected VLC Media Player

Media Information ✕

General Metadata Codec Statistics

Title: ProSTREAM HiFi

Artist:

Album: Date:

Genre: livewire Track number: /

Now Playing: **Chuck Berry - No Particular Place to Go** Language:

Publisher:

Copyright:

Encoded by:

Comments:

Location: <http://192.168.2.200:8000/hifi> Close



Custom Metadata Filter Scripts

The scripting engine inside the Z/IPStream R/1 uses the Lua Programming Language. All data arriving via port 9000 (UDP & TCP), and RS232 will pass to the Lua script. Scripts are managed via the Metadata Filters page.

Lua Script Example

The following Lua script will engage the Lua Line Parser when written to a file and saved as a Z/IPStream R/1 metadata filter.

```
UseLineParser()

function OnLineReceived(text,endOfLine)
    local title=""
    local url="http://TelosAlliance.com"
    title=text
    LogInfo("LineParser Script: title='",title,"', url='",url,"'\n")
    SendMetaDataSong(title,url)  -- Send metadata to application
end
```

Save this script with a name, like LineParserRaw.Lua and use the filters page of the Z/IPStream R/1 to load the file to the Z/IPStream R/1.

This script is an example of a basic script that takes any line of text submitted, and pass it directly though the metadata processor and reports the submitted characters to both the stream and the Z/IPStream R/1 system log when selected as the Z/IPStream R/1's metadata filter.

Code Breakdown

UseLineParser()

This line configures the filter to process the incoming data one line at a time. Another available option is UseXmlParser(), which will configure the filter to process the incoming data as XML. If neither of these functions are called, the filter will default to processing the incoming data one character at a time.

function OnLineReceived(text,endOfLine)

This function is called to deal with the incoming data when the line parser is used. The line parser accumulates the incoming data until a new line, or line feed character is detected. If the incoming line is very long, the function may be called before an end of line character is detected. The text variable will contain the line data. The endOfLine variable will be true if the the function was called in response to an end of line character, or false otherwise.

```
local title=""
```

```
local url="http://TelosAlliance.com"
```

The **local** keyword introduces variables used by the script. The variable **title** is set to empty initially. The URL is set to the home URL for the stream. You may configure a different URL string here, if required by your player.

```
LogInfo("LineParser Script: title='",title,"', url='",url,"'\n")
```

The LogInfo function is used to write data to the Z/IPStream R/1 logs. This function, when used in combination with a syslog server application will allow real-time analysis of the script during the customization, or "debug" phase of the script creation.

```
[10:06:56.208] Core.Info: Url (URL encoded)=''
[10:06:56.208] Core.Info: Reply from ICEcast server at chico:8000 (lofi): HTTP/1.0 200 OK,
Metadata update successful
[10:08:38.937] Metadata.Info: Lua: SendMetaDataSong(text='Thievery Corporation - Sol
tapado', url='')
[10:08:38.937] Metadata.Info: MetaData SendMetaDataSong: No code page conversion needed
('UTF-8', 'UTF-8')
[10:08:38.951] Core.Info: meta_song(title='Thievery Corporation - Sol tapado',url='')
[10:08:38.952] Core.Info: Send metadata to SHOUTcast v2 server at 192.168.2.200...
[10:08:38.952] Core.Info: Title (XML encoded)=Thievery Corporation - Sol tapado'
```

All data inside the double quotes of the **LogInfo** container, (**“LineParser Script: title=”,title,””, url=”,url,””\n”**) appears in the system log output. The variables from earlier in the script are specified by name to deliver their contents to the output. The **\n** specifies a newline character to represent the end of our submission to the log.

SendMetaDataSong(title,url)

The **SendMetaDataSong** function sends the title and url information to the stream server which will then pass it along to the clients. Although the word “Song” appears in the function name for historical reasons, you may send any information here; it does not have to be song related.

-- Send metadata to application

The Lua language uses two hyphens as a **comment** marker. The Lua interpreter will ignore any lines that begin with two hyphens. This provides a means to “mark up”, or document, the source file with notes for future reference, but has no effect on the function of the script. In this case, the comment is a reminder of what the role is served by this line of code.

end

The end statement signifies the end to the commands which comprise the **OnLineReceived** function.

Metadata Tools

Metadata filter creation and tuning can be as nuanced as tuning an audio processor. In order to provide real-time control of the filter process, some helpful tools are available which may assist with the process.

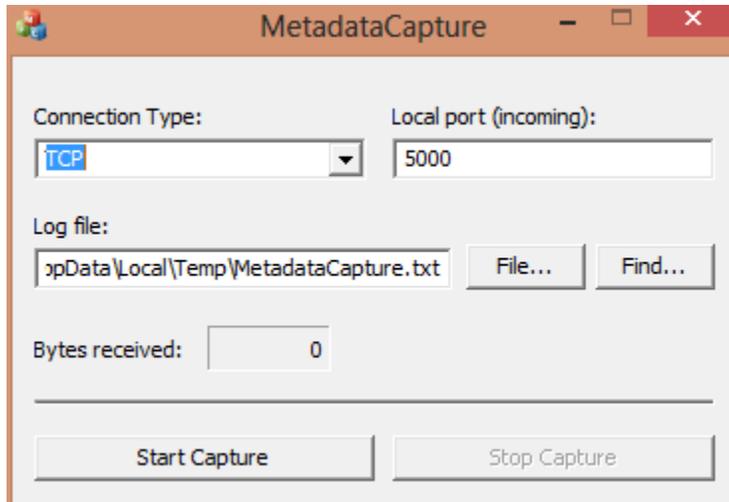
Metadata Pack

Download from: ftp://ftp.zephyr.com/pub/Tools/MetaData/Metadata_Pack.zip

Metadata Pack contains two Windows EXE programs, **MetadataCapture.exe** and **MetadataSend.exe**.

The download also contains a PDF file explaining application use.

MetadataCapture.exe



Metadata Capture application allows the TCP or UDP data from a remote automation system to be routed to this application to capture a precise example of what data is being sent by the automation system to the Z/IPStream R/1. When active, all data will be dumped to a file with timestamped delimiters.

Sample Output from Wide Orbit Automation for Radio

```

:::2012/10/27 23:25:08.641: Waiting for TCP connections on socket 9000:::

:::2012/10/27 23:25:14.581: Accept TCP connection on socket 9000 from
192.168.2.148:49296:::

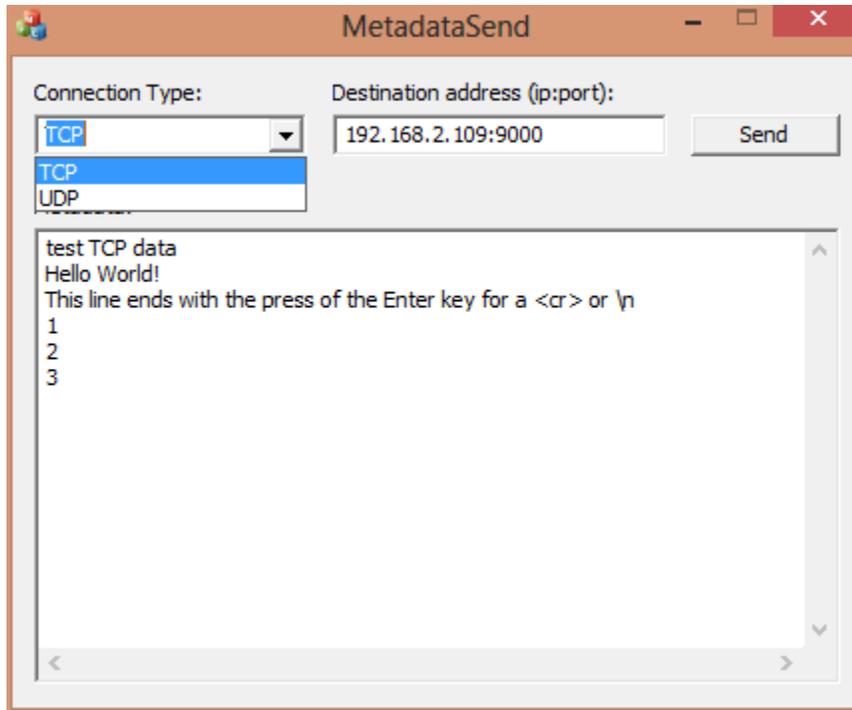
<nowplaying><sched_time>7749700</sched_time><air_time>84336000</air_time><stack_
pos></stack_pos><title>Wah Wah Man</title><artist>Young-Holt Unlimited</
artist><trivia>What It Is! Funky Soul and Rare Gr</trivia><category>MUS</cate-
gory><cart>001U</cart><intro>0</intro><end></end><station>RockLAB</station><du-
ration>227100</duration><media_type>SONG</media_type><milliseconds_left></
milliseconds_left></nowplaying>

:::2012/10/27 23:25:23.292: Closed TCP connection on socket 9000 from
192.168.2.148:49296:::

:::2012/10/27 23:25:23.292: Exiting TCP thread on port 9000:::

```

MetadataSend.exe

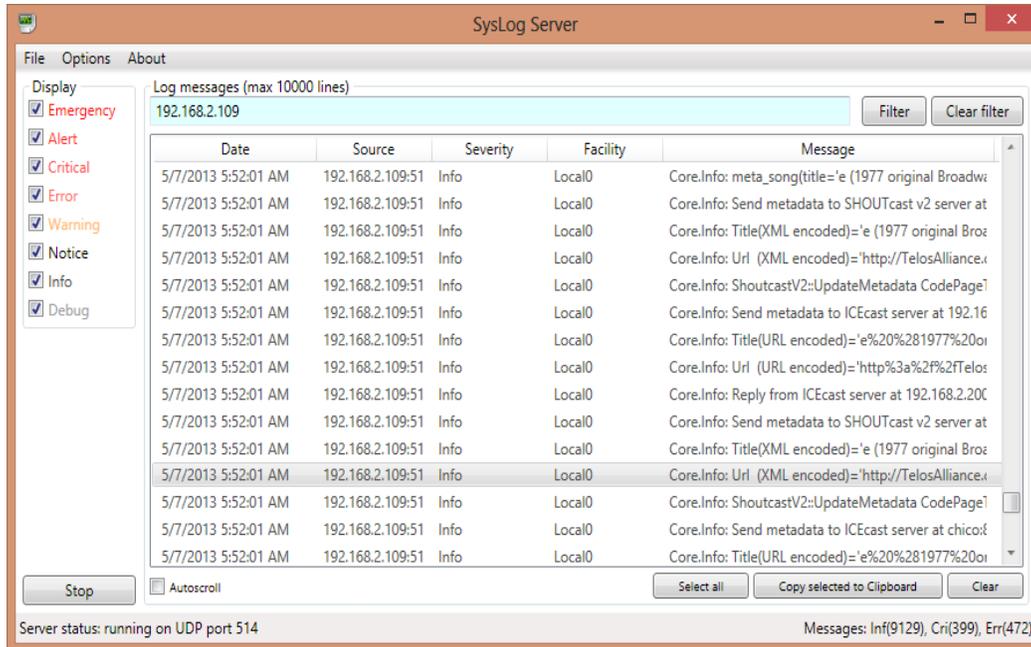


The Metadata Send program allows for direct push of data to the Z/IPStream R/1 on command. Copying data from the Metadata Capture program and pasting into Metadata Send is a common way of testing the script.

Note: To emulate TCP data, a linefeed is required. Data should be entered with at least one RETURN key press at the end of the line.

The download will also contain a PDF file explaining application use.

Syslog Server

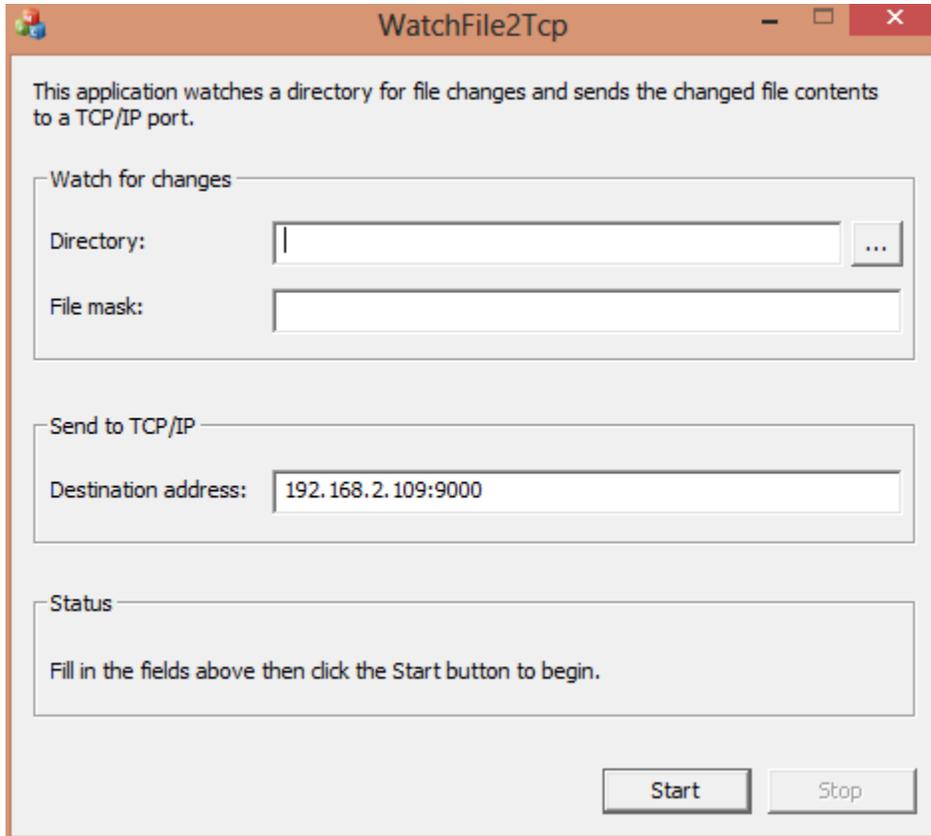


Download from: <ftp://ftp.zephyr.com/pub/Tools/MetaData/SyslogServer.zip>

Syslog Server is a free syslog server application to display incoming system logs in real-time. If desired, logs can also be saved to file. Seeing the log at the exact moment of submitting a test batch of data can be extremely helpful and is recommended process for script creation.

The download will also contain a brief PDF file explaining application use.

File to TCP



Download from: ftp://ftp.zephyr.com/pub/Tools/MetaData/Metadatas_Pack.zip

Some automation playback systems are only capable of writing metadata output to a file. The File to TCP application, when running, will watch for a file or folder to change, and when the change occurs, it copies the contents of the file to the TCP port of the Z/IPStream R/1.

The download will also contain a PDF file explaining application use.

Command Line usage

Syntax

WatchFile2TCP.exe /run [configuration file]

Key

/run Start the instance when loaded (same as pressing the Start button manually)

Example

```
WatchFile2TCP.exe /run configuration_1.cfg
```

Streams

Z/IPStream R/1 can be configured to push connect to up to four separate streaming servers.

Set the target server to match the required streaming server protocol.

HTTP Streaming

Z/IPStream R/1 **Stream 1**
Main

Configure the parameters for stream 1 below. Click the save button when done.

Stream type:

Audio codec:

Server address: (e.g. 192.168.1.23:8000)

Server password:

Stream name: (e.g. the station name)

Stream genre:

Stream URL:

Server version:

Stream availability:

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Z/IPStream R/1 contains a built-in mini server on TCP port 8000. This server can accept only a few connections and should not be used as the main streaming server or be deployed to the public.

The HTTP server is intended to be used for stream monitoring with a player or to have another media server pull the stream from Z/IPStream R/1 allowing monitoring of the encoded stream. This is helpful in troubleshooting server connections or just as a quick way to test everything is working as expected. The HTTP server is also useful when adjusting audio processing parameters as the audio can be monitored with compression. The audio delay should also be shorter than listening to the stream delivered through an external server.

Test Stream Connection URL's

Codec 1	Codec 2
PLS: http://ip.address/play.pls	PLS: http://ip.address/play2.pls
M3U: http://ip.address/play.m3u	M3U: http://ip.address/play2.m3u
ASX: http://ip.address/play.asx	ASX: http://ip.address/play2.asx
RAW: http://ip.address:8000	RAW: http://ip.address:8010

Streaming Services

A streaming service can provide all that is needed for distribution of Z/IPStream R/1's encoded audio stream. Typically pricing via a monthly base rate and bandwidth costs. The streaming service provider will provide the stream configurations for Z/IPStream R/1 and return to the client any number of services including stream link URL's to provide to listeners. A streaming service can also provide failover and optional configurations should a connection be lost.

Stream Server Service Providers

- ◆ TritonDigital: <http://www.tritondigital.com/>
- ◆ Akamai: <http://www.akamai.com/>
- ◆ LimeLight: <http://www.limelight.com/>
- ◆ Stream Guys: <http://www.streamguys.com/>

Additional streaming services can be found at <http://www.radiotoolbox.com/hosts/>

SHOUTcast

Target Server

Set the target server to SHOUTcast.

SHOUTcast Server Address

IP.address:port of the SHOUTcast server, I.E. 192.168.2.200:8010

SHOUTcast Server Password

Value of **password=** from the SHOUTcast configuration file.

Stream Name

When an Internet user or device connects to the stream, the Stream Name will be sent in the initial meta data burst. This title is typically displayed on the end-user's player or device. Capitalization and punctuation of Stream Name is fine and will be displayed as such in the end client.

Stream Genre

SHOUTcast DNAS server requires a genre field for all incoming connections. While the genre may not be displayed on the end user's device or client application, some streaming servers and services may publish the existence of the encoded audio stream in a directory. SHOUTcast recommends the genres listed in their forums: <http://forums.winamp.com/showthread.php?t=303241>

Stream URL

When an Internet user or device connects to the stream, the Stream URL will be sent in the initial meta data burst. This value can be set to an URL that is common to all the songs streamed to the SHOUTcast server (e.g. the website of the radio station).

Server version

Z/IPStream R/1 can connect to SHOUTcast servers using SHOUTcast protocol version 1 or a newer enhanced SHOUTcast protocol version 2. If the server version is not known it will probably be the older protocol version 1.

Stream ID

If the SHOUTcast protocol version 2 is selected, a stream ID must be entered. Typically this will be 1 if only one stream is sent to the server.

Stream availability

Allows defining if the stream is meant to be public and can be included in a server specific directory listings or private and should not be listed to the end-users. A good example of this is SHOUTcast DNAS which will publish any streams configured as public to <http://www.SHOUTcast.com/>.

ICEcast

Target Server

Set the target server to ICEcast.

Audio Codec

Select which of the encoded streams will be delivered by this configuration.

ICEcast Server Address

IP.address:port of the ICEcast server, I.E. 192.168.2.200:8010

Mount Point

Specify a unique URL compatible string for the connection URL for the stream, I.E. hifi to have the connect string for this stream be <http://192.168.2.200:8000/hifi>. It is acceptable to specify an extension if desired, such as .mp3 or .aac.

Stream User Name

The default stream username for ICEcast is source unless it is overridden in the icecast.xml configuration file.

Stream Password

The stream password from the icecast.xml configuration file.

Metadata User Name

The ICEcast administrator username. Note that the configuration will work without a metadata username, but metadata will not pass to ICEcast or clients.

Metadata Password

The administrator password from the icecast.xml configuration file. Note that the configuration will work without a metadata username, but metadata will not pass to ICEcast.

Stream Name

The name of the stream as displayed to clients.

Stream Genre

Genre of the audio stream.

Stream URL

Web address of the station's website.

Stream availability

Allows defining if the stream is meant to be public and can be included in a server specific directory listings or private and should not be listed to the end-users.

RTP

Target Server

Set the target server to RTP

Audio Codec

Specify which of the two codec encodes to use for the RTP stream.

Target server:	<input type="text" value="RTP"/>
Audio codec:	<input type="text" value="Codec 1 (AAC-LC, 44100 Hz, 96 kbps, Stereo)"/> <input type="text" value="Codec 2 (HE-AAC v2, 44100 Hz, 48 kbps, Stereo)"/> <input type="text" value="Codec 3 (PCM 44.1 KHz)"/> <input type="text" value="Codec 4 (PCM 48 KHz)"/>
RTP destination address:	<input type="text" value="192.168.2.202"/> (e.g. 239.1.1.5)
RTP destination port:	<input type="text" value="4080"/>
Time to live (TTL):	<input type="text" value="15"/> (e.g. 15)
Stream name:	<input type="text" value="ProSTREAM RTP"/>
RTP interface:	<input type="text" value="WAN"/>
Force payload type to MPA:	<input type="text" value="No"/> (for Luci Live compatibility)
Samples per packet:	<input type="text" value=""/> (e.g. 512)
<input type="button" value="Save"/>	

Note:

RTP allows uncompressed delivery as well. Select a PCM codec to be able to specify the Samples per RTP packet.

RTP destination address

Enter the destination UDP address to which RTP packets will be sent. Address can be unicast or multicast.

RTP destination port

Enter the UDP port to which RTP packets will be sent.

Time to live (TTL)

Defines how “far” from a sending host a given UDP packet should be forwarded. For example, use value 1 to limit UDP streams to local network subnet only.

Stream name

The name entered here will be sent in the SDP discovery broadcast messages. This title is typically displayed on the end-user’s player or device when stream is discovered and played.

RTP interface

If more than one network interface for Z/IPStream R/1 is configured, RTP interface selection allows to define to which network interface RTP stream will be sent. Usually interface needs to be set to LAN to broadcast stream to Local Area Network.

RTMP

Target server

Set the target server to RTMP for RTMP streaming services. Service providers like TritonDigital, Akamai, and LimeLight often use RTMP services for live audio streaming.

Server address

Enter the server address and port number where the address and port are separated by a colon character. For example, if the address is 192.168.1.23 and the port is 1935, you would enter 192.168.1.23:1935 in the address field.

Use Authentication

If RTMP server needs authentication, set this field to “yes.” The fields for stream username and stream password will appear automatically.

The screenshot shows a configuration form for RTMP streaming. The 'Use Authentication' dropdown is set to 'yes'. Below it, the 'Stream user name' field contains 'source' and the 'Stream password' field contains 'streampwd'. The 'Publish path' field contains 'live' and the 'Stream name' field contains 'ProSTREAM LoFi'. A note at the bottom states: 'When AAC codec is used with RTMP stream servers the transport format must be set to RAW or ADIF!'. A 'Save' button is located at the bottom of the form.

Stream user name

If RTMP authentication is selected, enter the server username.

Stream password

If RTMP authentication is selected, enter the server password.

Publish path

Enter the stream publish path. Usually this will be the RTMP application name used to receive the stream (e.g. “live” is the default RTMP application for Adobe Flash Media Server).

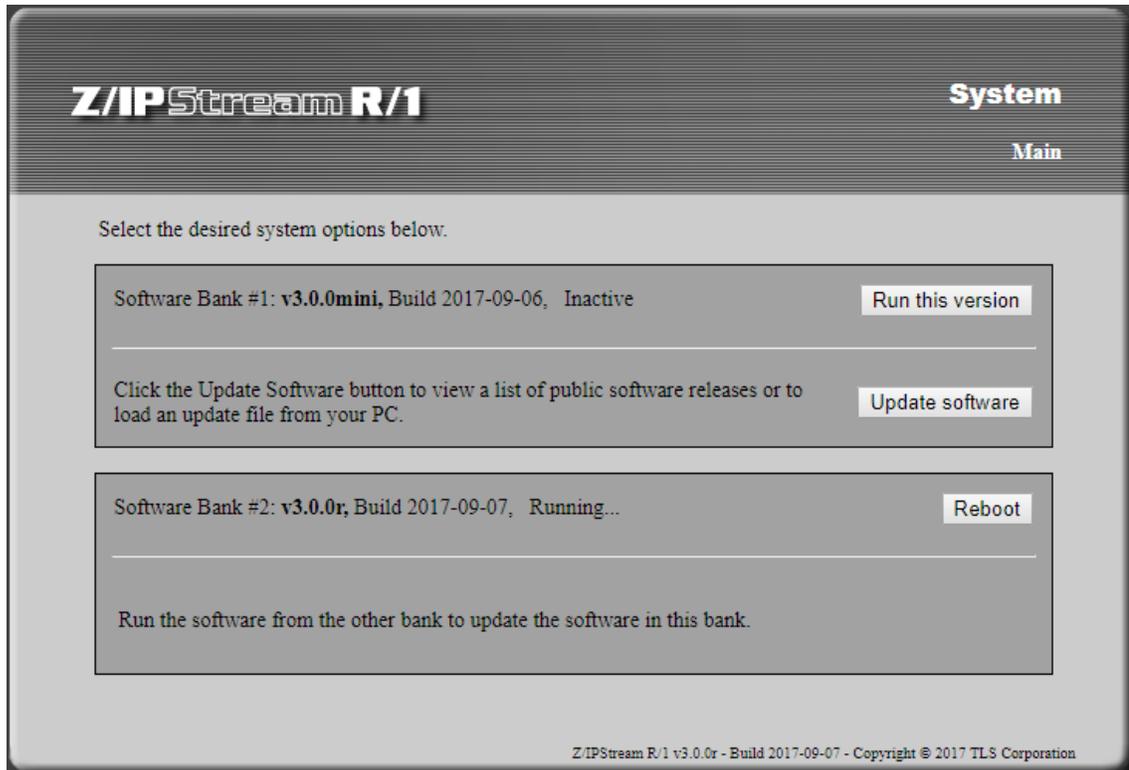
Stream name

Enter the RTMP stream name. RTMP stream can also be written with multiple paths (e.g. path1/path2/livestream).

Note:

RTMP streams require 44100 Hz sample rate to be used in the Z/IPStream R/1 MP3/AAC codec settings and RAW or ADIF transport format when AAC codec is selected.

System



The system page can be accessed to either switch the firmware bank of the Z/IPStream R/1, or to reboot the unit. The System Web interface will also allow the upload of new software images.

Note:

It is good practice to schedule periodic maintenance of the Z/IPStream R/1 encoder and check for updated software. New software can be found at <http://www.TelosAlliance.com/Telos/ZIPStream-R1>.

Firmware is delivered via a single compressed .tgz file (it may be delivered within a .zip file from the website; if so, unzip to access the .tgz file). From the system page of the web interface, the “Update Software” button will bring up a file browser where the new firmware image file can be updated to the software bank. Once uploaded, Z/IPStream R/1 can be rebooted to the new software bank. All existing configuration options will remain intact.

Advanced Options

Change web interface username/password

Z/IPStream R/1 **Change Password**

[Main](#) [Advanced Options](#)

Username:

Old password:

New password:

Repeat new password:

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The default username/password is user without password and admin/Telos.

NOTE:

For security reasons, it is very important to change the default login credentials. Failure to do so may give others access to your Z/IPStream R/1.

Logs/Date Time

Z/IPStream R/1 Logs
Main

Wed, 13 May 2015 17:21:35 GMT

Date: (YYYY MM DD)

Time: (HH MM SS, enter hour as 24-hour value)

Remote logging support:

Server address (UDP port 514): (e.g. 192.168.1.23)

Last 500 log message lines ([refresh page to update](#)):

```
[16:13:23.007] Syslog.Info: TRACE messages will not be logged!
[16:13:23.007] Syslog.Info: Start child...
[16:13:23.008] Syslog.Info: Child process...
[16:13:23.009] Syslog.Info: Continuing...
[16:13:24.062] Syslog.Info: srv wrote 16 to 0x7f000001:46574...
[16:13:24.067] Core.Info: Start child...
[16:13:24.067] Core.Info: Wait for child 1901...
```

Date/Time

If required, it is possible to reset the Z/IPStream R/1 system time from the Logs menu.

Logs

To help troubleshooting problems, Z/IPStream R/1 allows to view last 500 system log messages when the web interface control panel is used. If a problem occurs (cannot connect to the streaming server, etc.), log messages should be reviewed before any Z/IPStream R/1 reboots because the log messages are deleted when the unit reboots.

If more than 500 log message lines must be reviewed it is possible to set up a SysLog compatible server on another computer and forward log messages to this server by entering its IP address in the Z/IPStream R/1 logs section. This allows all new log messages to be sent to the remote SysLog server to assist with system setup and diagnosis.

Syslog Server

Assign the IP address of a computer running a syslog server. All logs written to the Z/IPStream R/1 syslogs will also be written at the same time to the syslog server assigned.

A syslog server will present logs as they are generated which can assist with setup and diagnosis. Metadata setup especially benefits from use of the real-time syslog server.

Ping Service

The ping service provides a means to test connectivity to certain IP addresses from the Z/IPStream R/1. This can be useful in diagnosing connection issues from the Z/IPStream R/1 to remote servers. Ping results will display both in the Web interface and in syslog as *pinger* events.

```
pinger.Warning: Host 192.168.1.200 failed to respond to pings after 15 sec!
```

Host IP

Specify up to four (4) separate IP addresses to test for ping response.

Ping Interval

Specify the interval in seconds to wait between ping attempts.

Alarm after

Trigger an alarm after a specified number of seconds of failed attempts. Must be equal to or larger than the Ping interval.

The ping service is not configurable via the front panel.

Configure Static Routes

Z/IPStream R/1 **Static Routes**

[Main](#) [Advanced Options](#)

This page allows you to configure static network routes. The list below displays the currently configured routes (if any). You may also add new static routes below.

Please do not use your browser's back or refresh buttons on this page. Doing so may repeat the last operation with unintended consequences.

Currently configured static routes:

172.16.0.0/16 via 192.168.0.254 on LAN	delete
--	------------------------

Add static route:

Destination address (e.g. 192.168.1.0/24):

Gateway:

Interface: WAN ▼

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Static Routing allows path selection for specific connections. A static route allows explicit specification of the gateway and the network port (WAN or LAN).

Destination Address

Destination address can be expressed as an IP address or a network range in [CIDR notation](#) [[cidr/flsm supernet calculator](#)]. IE 192.168.0.0/24 expresses the same information as 192.168.2.0 : 255.255.255.0 in network speak.

Note:

If a single IP address is entered, Z/IPStream R/1 will append the IP address with /32 to indicate a single IP address in the range.

Gateway Address

A gateway is an IP address of a router, or a device which can speak to multiple networks.

Interface

Interface specifies which physical network connection the Z/IPStream R/1 should use for this communication. LAN and WAN refer to the two back-panel network ports.

Note:

The Gateway Address must be within the range specified for the selected port (as per the Z/IPStream R/1's Network configuration) or the route will not be accepted.

View network diagnostics

Network diagnostics will present the standard Linux output of network configurations. No user functions are included.

Interface configuration

```
-- Interface configuration
eth0    Link encap:Ethernet  HWaddr 00:13:95:06:eb:e2
        inet addr:192.168.2.109  Bcast:192.168.2.255  Mask:255.255.255.0
        inet6 addr: fe80::213:95ff:fe06:ebe2/64 Scope:Link
        UP BROADCAST RUNNING MULTICAST  MTU:1500  Metric:1
        RX packets:5225450 errors:0 dropped:0 overruns:0 frame:0
        TX packets:418301 errors:0 dropped:0 overruns:0 carrier:0
        collisions:0 txqueuelen:1000
        RX bytes:646595854 (616.6 MiB)  TX bytes:430907987 (410.9 MiB)
        Interrupt:40 Base address:0xe000

lo      Link encap:Local Loopback
        inet addr:127.0.0.1  Mask:255.0.0.0
        inet6 addr: ::1/128 Scope:Host
        UP LOOPBACK RUNNING  MTU:16436  Metric:1
        RX packets:2825 errors:0 dropped:0 overruns:0 frame:0
        TX packets:2825 errors:0 dropped:0 overruns:0 carrier:0
        collisions:0 txqueuelen:0
        RX bytes:348397 (340.2 KiB)  TX bytes:348397 (340.2 KiB)
```

Link information

```
-- Link information
1: lo:  mtu 16436 qdisc noqueue state UNKNOWN
    link/loopback 00:00:00:00:00:00 brd 00:00:00:00:00:00
2: dummy0:  mtu 1500 qdisc noop state DOWN
    link/ether 62:ea:d4:7e:98:68 brd ff:ff:ff:ff:ff:ff
3: eth0:  mtu 1500 qdisc pfifo_fast state UNKNOWN qlen 1000
    link/ether 00:13:95:06:eb:e2 brd ff:ff:ff:ff:ff:ff
4: sit0:  mtu 1480 qdisc noop state DOWN
    link/sit 0.0.0.0 brd 0.0.0.0
5: usb0:  mtu 1492 qdisc noop state DOWN qlen 1000
    link/ether ba:4e:73:4f:5c:c1 brd ff:ff:ff:ff:ff:ff
```

IP rules

```
-- IP rule
0:      from all lookup local
32764:  from all to 192.168.2.109 lookup eth_tbl
32765:  from 192.168.2.109 lookup eth_tbl
32766:  from all lookup main
32767:  from all lookup default
```

IP routing

```
-- IP routing
192.168.2.0/24 dev eth0 proto kernel scope link src 192.168.2.109
default via 192.168.2.3 dev eth0
```

Kernel IP routing table

Destination	Gateway	Genmask	Flags	Metric	Ref	Use	Iface
192.168.2.0	*	255.255.255.0	U	0	0	0	eth0
default	192.168.2.3	0.0.0.0	UG	0	0	0	eth0

Kernel IP routing cache

Source	Destination	Gateway	Flags	Metric	Ref	Use	Iface
192.168.2.50	239.192.255.3	239.192.255.3	m1	0	0	87	lo
192.168.2.109	239.192.255.3	239.192.255.3	m1	0	0	237	eth0
192.168.2.202	192.168.2.109	192.168.2.109	il	0	0	1865	lo
192.168.2.201	192.168.2.109	192.168.2.109	il	0	0	501	lo
192.168.2.90	239.192.255.3	239.192.255.3	m1	0	0	245	lo
192.168.2.209	192.168.2.255	192.168.2.255	ib1	0	0	25	lo
192.168.2.20	239.192.255.3	239.192.255.3	m1	0	0	69	lo
192.168.2.204	192.168.2.255	192.168.2.255	ib1	0	0	33	lo
google-public-d	192.168.2.109	192.168.2.109	l	0	0	134	lo
192.168.2.109	192.168.2.200	192.168.2.200		0	3	6	eth0
192.168.2.204	192.168.2.109	192.168.2.109	il	0	0	237	lo
192.168.2.109	239.192.255.3	239.192.255.3	m1	0	0	41	eth0
192.168.2.202	255.255.255.255	255.255.255.255	ib1	0	0	60	lo
localhost	localhost	localhost	l	0	4	5	lo

View memory usage diagnostics

```

Mem: 120196K used, 901412K free, 0K shrd, 1216K buff, 22096K cached
CPU: 16.6% usr  8.3% sys  8.3% nic 66.6% idle  0.0% io  0.0% irq  0.0% sirq
Load average: 0.24 0.24 0.26 1/93 2026
  PID  PPID  USER      STAT   VSZ  %MEM  CPU  %CPU  COMMAND
 1912  1910  root       S      82592  8.0   0  24.9  ./core
 2026  1923  root       R N    2604  0.2   0   8.3  top -b -n1
 1923   1  root       S N   58668  5.7   0   0.0  ./webcfg
 1925   1  root       S N   53824  5.2   0   0.0  ./metadata
 1790   1  root       S      28536  2.7   0   0.0  /usr/sbin/rsyslogd -c3
 1924   1  root       S N   27972  2.7   0   0.0  ./plcysrv
 1922   1  root       S N   23044  2.2   0   0.0  ./ui
 1926   1  root       S N   19648  1.9   0   0.0  ./pinger
 1908  1906  root       S N   11424  1.1   0   0.0  ./syslog
 1802   1  root       S      5372  0.5   0   0.0  /usr/sbin/sshd
 1910   1  root       S      5332  0.5   0   0.0  ./core
 1815   1  root       S      5240  0.5   0   0.0  /usr/sbin/snmpserv /etc/snmpserv/s
 1878   1  root       S      3300  0.3   0   0.0  /usr/sbin/axialwrdr
 1906   1  root       S N    3228  0.3   0   0.0  ./syslog
 1882   1  root       S      3020  0.2   0   0.0  /usr/sbin/axiaadvd -fspy /tmp/lwst
 1886   1  root       S      2976  0.2   0   0.0  /usr/sbin/axiagpr
  101   1  root       S <    2884  0.2   0   0.0  udevd --daemon
 1830   1  root       S      2604  0.2   0   0.0  /usr/sbin/crond
   1    0  root       S      2148  0.2   0   0.0  init [2]
 1811   1  root       S      2000  0.2   0   0.0  /usr/sbin/inetd
 2025  1923  root       S N    1996  0.1   0   0.0  route -C
 1966   1  root       S      1812  0.1   0   0.0  /sbin/getty 38400 tty1
 1967   1  root       S      1812  0.1   0   0.0  /sbin/getty 38400 tty2
 1968   1  root       S      1812  0.1   0   0.0  /sbin/getty 38400 tty3
 1969   1  root       S      1812  0.1   0   0.0  /sbin/getty 38400 tty4
 1970   1  root       S      1812  0.1   0   0.0  /sbin/getty 38400 tty5
 1971   1  root       S      1812  0.1   0   0.0  /sbin/getty 38400 tty6
 1891   1  root       S      1668  0.1   0   0.0  ./onnetchg /flash/ipcodec/current/

```

The memory usage page is based on the standard Linux tool `top`. This presents a list of all processes in the CPU and can be very helpful in debugging rogue processes. No user functions are included.

View system diagnostics

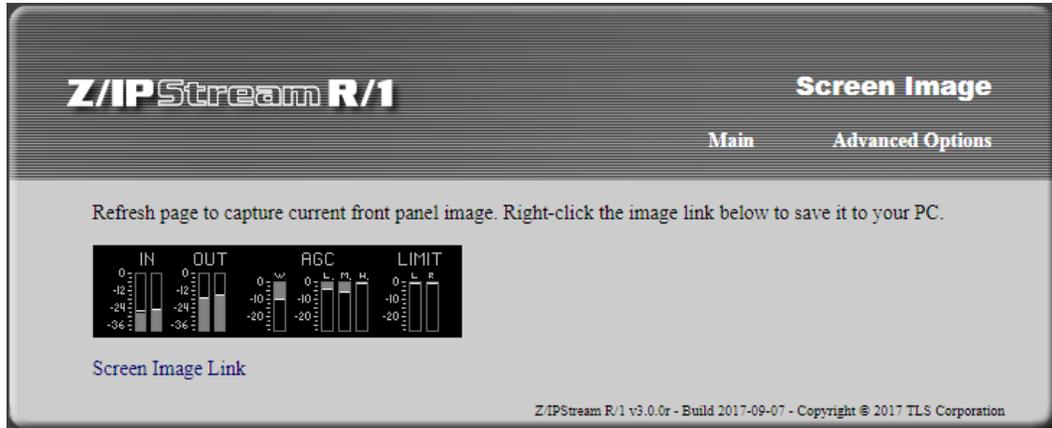
```

: BAR 7: assigned [io 0x1000-0x1fff]
[ 0.245098] pci 0000:00:1c.0: PCI bridge to [bus 01-01]
[ 0.245108] pci 0000:00:1c.0: bridge window [io 0x1000-0x1fff]
[ 0.245123] pci 0000:00:1c.0: bridge window [mem 0x80000000-0x801fffff]
[ 0.245135] pci 0000:00:1c.0: bridge window [mem 0x80200000-0x803fffff pref]
[ 0.245150] pci 0000:00:1c.1: PCI bridge to [bus 02-02]
[ 0.245159] pci 0000:00:1c.1: bridge window [io 0xe000-0xefff]
[ 0.245172] pci 0000:00:1c.1: bridge window [mem 0xfeb00000-0xfebfffff]
[ 0.245184] pci 0000:00:1c.1: bridge window [mem 0xcff00000-0xcfffffff pref]
[ 0.245214] pci 0000:00:1c.0: enabling device (0104 -> 0107)
[ 0.245238] pci 0000:00:1c.0: PCI INT A -> GSI 16 (level, low) -> IRQ 16
[ 0.245252] pci 0000:00:1c.0: setting latency timer to 64
[ 0.245275] pci 0000:00:1c.1: PCI INT B -> GSI 17 (level, low) -> IRQ 17
[ 0.245287] pci 0000:00:1c.1: setting latency timer to 64
[ 0.245299] pci_bus 0000:00: resource 4 [io 0x0000-0x0cf7]
[ 0.245307] pci_bus 0000:00: resource 5 [io 0x0d00-0xffff]
[ 0.245316] pci_bus 0000:00: resource 6 [mem 0x000a0000-0x000bffff]
[ 0.245325] pci_bus 0000:00: resource 7 [mem 0x000d0000-0x000dffff]
[ 0.245335] pci_bus 0000:00: resource 8 [mem 0x3fc00000-0xdfffffff]
[ 0.245344] pci_bus 0000:00: resource 9 [mem 0xf0000000-0xffffffff]
[ 0.245353] pci_bus 0000:01: resource 0 [io 0x1000-0x1fff]
[ 0.245362] pci_bus 0000:01: resource 1 [mem 0x80000000-0x801fffff]
[ 0.245371] pci_bus 0000:01: resource 2 [mem 0x80200000-0x803fffff pref]
[ 0.245380] pci_bus 0000:02: resource 0 [io 0xe000-0xefff]
[ 0.245389] pci_bus 0000:02: resource 1 [mem 0xfeb00000-0xfebfffff]
[ 0.245398] pci_bus 0000:02: resource 2 [mem 0xcff00000-0xcfffffff pref]
[ 0.245531] NET: Registered protocol family 2
[ 0.245701] IP route cache hash table entries: 32768 (order: 5, 131072 bytes)
[ 0.248051] size: 524280
[ 0.248135] TCP established hash table entries: 131072 (order: 8, 1048576 bytes)
[ 0.249463] TCP bind hash table entries: 65536 (order: 6, 262144 bytes)
[ 0.249868] TCP: Hash tables configured (established 131072 bind 65536)
[ 0.249876] TCP reno registered
[ 0.249886] UDP hash table entries: 512 (order: 1, 8192 bytes)
[ 0.249908] UDP-Lite hash table entries: 512 (order: 1, 8192 bytes)

```

System diagnostics presents the output of the Linux `dmesg`. This can be helpful in diagnosing startup or other system-level issues. No user functions are included.

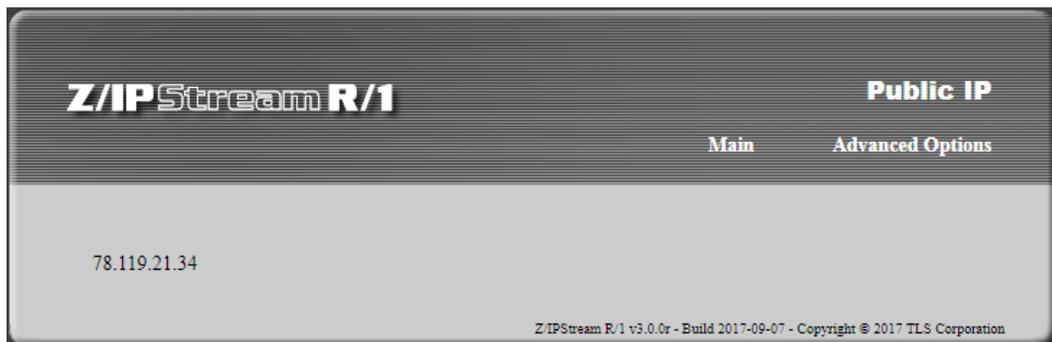
View front panel image



View front panel image will present the same image as what is currently displayed on the unit front panel. A direct link is included on this page. No user functions are included.

Display public IP (works only if DNS is configured)

Display public IP will return a simple HTML page with nothing more than the public IP address of the routed IP network. Since this tool references against <http://whatismyip.akamai.com/>, failure of this command can be an indicator of an improperly or un-configured DNS entry under the Z/IPStream R/1's Network Settings. If in doubt, Google's public DNS servers can be used @ 8.8.8.8 and 8.8.4.4.



5 Omnia Processing

The dynamics processing algorithms in use in the Z/IPStream R/1 are based upon Omnia Audio’s highly successful Omnia.3net, a hardware-based DSP audio processor.

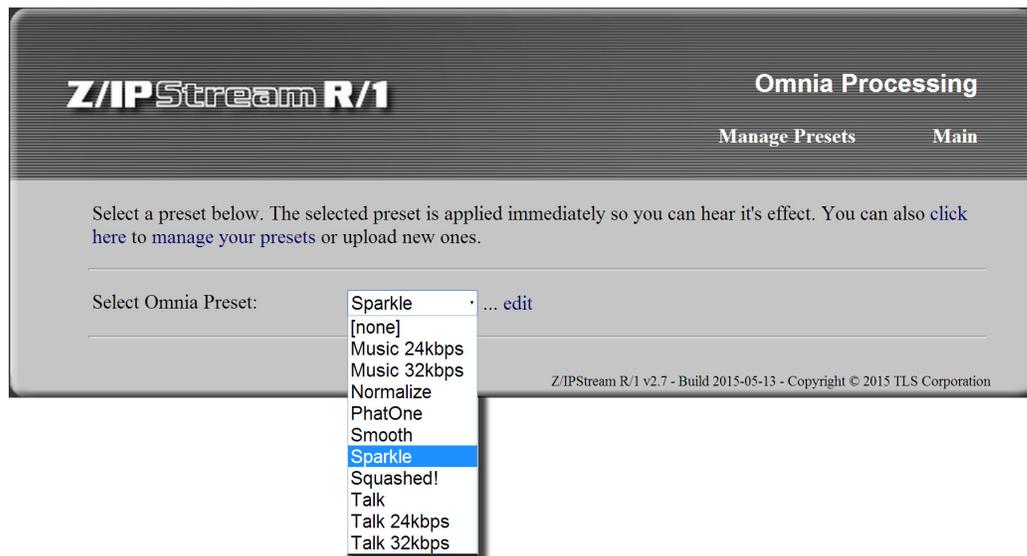
Z/IPStream R/1 employs the following Omnia processing architecture:

- ◆ Input
- ◆ Wideband AGC
- ◆ 3 Band Combined Compressor/Limiter
- ◆ Adjustable Bandwidth Low-pass Filter
- ◆ Final Look-Ahead Limiter
- ◆ Output
- ◆ Processing Preset Editor/Preset Management

All audio processing adjustments are performed in the Preset Editor, which is only available via the web interface. A “preset” represents all of the control values for each control in the Omnia processor. When a preset is chosen for the processor, control values for the processor’s settings are loaded into the processing structure and applied to the input audio.

Presets allow different processing setups to be loaded into the processor. Factory presets have been provided as starting points for customizing your sound.

It is possible to gain dramatic sonic improvement from Z/IPStream R/1’s Omnia processing right away using one of the factory presets. Start by selecting one from the “Omnia Processing” menu and then click “...edit” to bring up the Preset Editor.



Preset Editor

Using the Preset Editor, it is possible to customize the chosen preset to suit your own aural requirements.

Note: Be sure to click “Save” in the upper-right hand corner of the editor window when you are finished to save your changes. If you do not, they will be lost when the unit is rebooted. To revert to the preset’s original values, press cancel or change preset in Omnia Processing page.

Processor Adjustment Overview

1. Start with a factory preset and adjust processor values in small increments.
2. Listen to the post-processed audio, ideally through the codec to be used, then adjust, and listen again. Aural evaluation should occur over time, not in short moments.
3. While making changes to the system, adjust in small increments. Do not make too many different adjustments all at once. Multiple adjustments at once can make for more difficult determination of what affected an aural change. Make changes in increments of no more than +0.5 dB or -1 dB, or in 1 or 2 step increments. Turn functions on and off to better determine their effect on the audio.
4. Sleep on it. Spend time adjusting and then listening, and when the system gets to a point where it sounds good, stop. When making changes to processing, ears become less sensitive to the adjustments. Spreading the adjustment period over a number of days is recommended.

The following section will detail each control if you’d like to customize one of the factory presets to get your own custom sound.

If you are using a factory preset as-is, please read the “Input” and “Output” adjustment sections as these are very important for proper operation and sound quality.

Processor Adjustment

Bypass

The Bypass control (which appears in each menu), when turned ON, will completely bypass all processing, effectively connecting the input directly to the output. Bypass control state will not be saved. To bypass processing, please select “[none]” processing preset.

Input

Editing preset "Sparkle"

Input

Gain Left 0.0 dB

Gain Right 0.0 dB

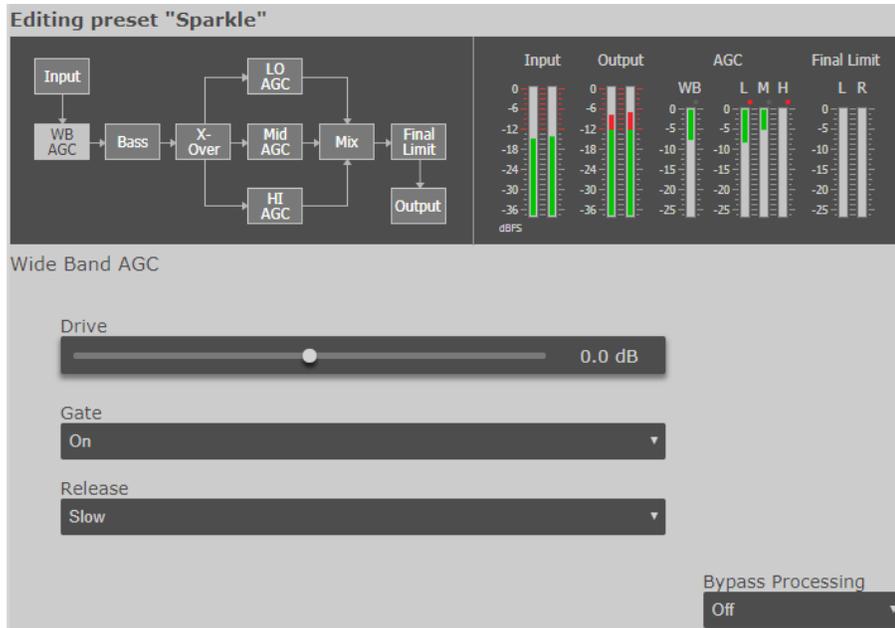
Bypass Processing
Off

Proper input gain is essential for optimum processing.

Gain

Set the “Gain Left” and “Gain Right” controls so that the peak-reading Input meters are peaking “just into the red” at -12 dBFS with normal “OVU” program audio playing.

WB AGC



The Wide-Band AGC performs overall gain-riding to the input signal over a 30dB range. It acts like an automated “hand on the pot”, keeping the overall level to the 3-Band compressor/limiter nice and consistent.

Drive

Once the Input Gain is set, you can adjust the Wide-Band AGC Drive if desired. This is measured using the “WB” gain-reduction meter. This meter reads from the top down where “0” is no gain-reduction at all. Normally, this should read down to between about 10-15dB with normal “OVU” program audio playing. This will allow room both above and below for automatic gain adjustment.

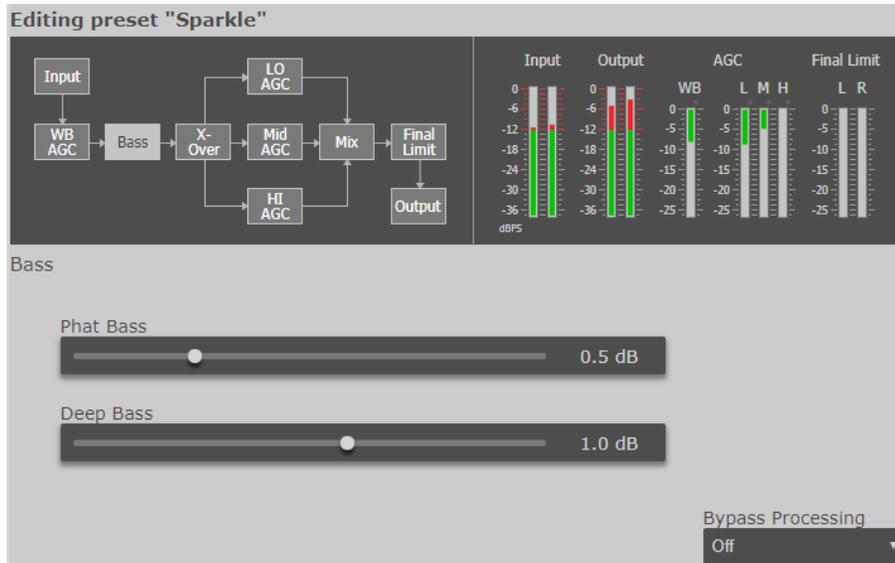
Gate

This is a silence gate that freezes the gain when silence is detected to help keep noise from being brought up during program pauses. The normal setting is “On” but the silence gate can be defeated by setting this control to “Off” if desired.

Release

This controls the overall speed of operation of the WB AGC. “Slow” will preserve more of the natural dynamic range in the program material. “Fast” will bring up lower level passages faster and react faster to program level changes. “Medium” is a nice compromise between the two.

Bass



Some bass boost is normally used when processing to balance out the spectrum and add some “punch.” These two controls adjust the amount of the Z/IPStream R/1’s bass enhancement section. They are more than just simple EQ’s. If your source material has it, Z/IPStream R/1 will put it in the mix, and with muscle!

Making changes to the bass is done using the following parameters, and for maximum effectiveness, in the order listed: (NOTE: Settings of “0.0” apply no bass boost.)

1. In the Bass menu, increase the amount of Deep Bass boost to taste. This is the lowest “kick drum” bass, so be sure to monitor with speakers or headphones that can reproduce bass below 80Hz.
2. In the Bass menu, increase the amount of Phat Bass boost to taste. This special bass effect can help smaller speakers sound like they have more bass than they really do.
3. Increase the drive to the LF AGC in the LO AGC screen.
4. Speed up the release times of the LO AGC section.
5. Slightly increase the low control in the mixer section.

When processing, moderation is the key. If all of the above steps are utilized in the pursuit of more bass, severe low frequency over-enhancement can occur. Excessive bass boost can produce the illusion that the presence and high frequencies have been lost. They are still there, but are being acoustically masked by the lower frequencies. Generally, an increase in the Bass menu’s bass boost functions alone will provide more than sufficient enhancement to the low end.

Phat Bass

Phat bass widens the higher low-end frequencies in an effort to expand bass sound on smaller playback speakers.

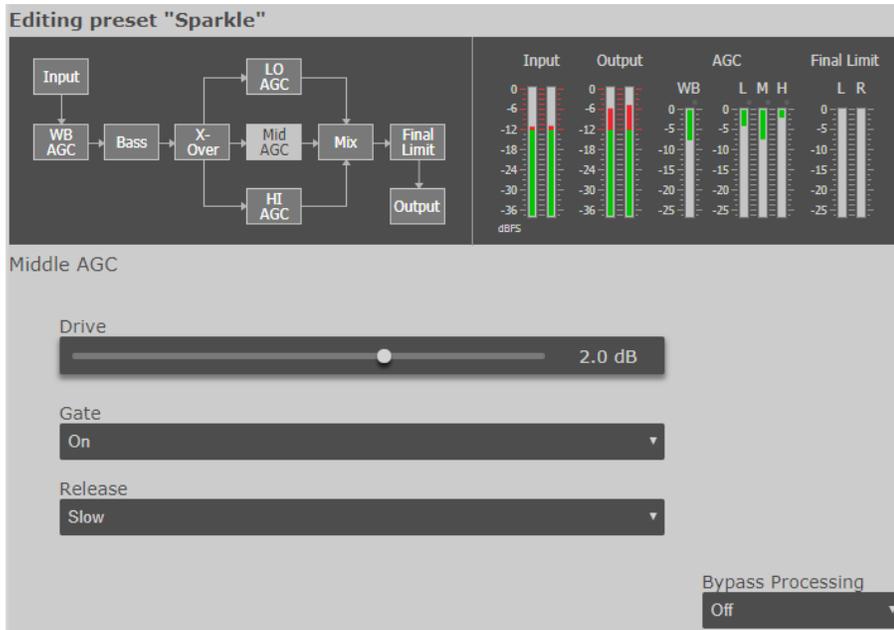
Deep Bass

Deep bass focuses on frequencies of 80hz and below.

X-Over

The X-Over (crossover) splits the audio from the WB AGC into 3 frequency bands so each band can be optimally processed. The three bands are: Lo (bass) Mid (midrange - voice and solo instruments) and Hi (treble). There are no adjustable controls in the crossover section.

Low, Mid, Hi AGC



Each band has a Drive, Gate and Release control, similar to the WB AGC.

Drive

The Drive control adjusts the volume of the audio from the crossover into the compressor/limiter for the selected band. Higher settings will drive the selected band into more gain-reduction. Some EQ can also be done with the Drive controls if one band is adjusted differently than the others. Start by setting them equally though.

More drive will provide a denser, more compressed (but louder) sound. Less drive will provide a more natural and "open" sound.

Generally, you will see these meters move faster than the WB AGC. Generally, between 5 and 10 dB of gain-reduction is a good nominal amount. Note that the Hi band may show less gain-reduction than the others. This is normal.

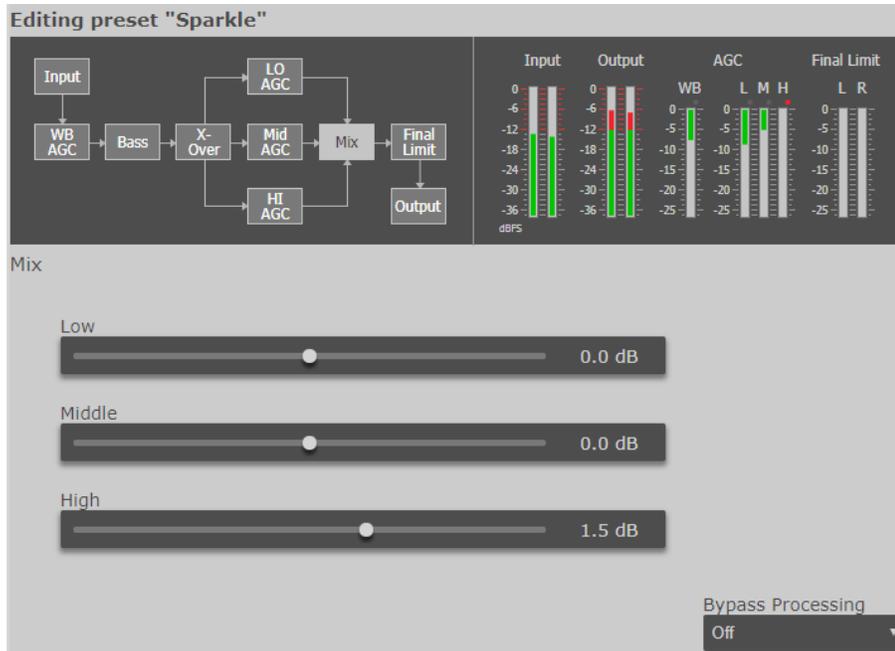
Gate

This is a silence gate that freezes the gain when silence is detected to help keep noise from being brought up during program pauses. The normal setting is "On" but the silence gate can be defeated by setting this control to "Off" if desired.

Release

This controls the overall speed of operation of the compressor/limiters. "Slow" will preserve more of the natural dynamic range in the program material. "Fast" will react faster to program level changes. "Medium" is a nice compromise between the two.

Mix



The final mixer level controls set the output level of each of the 3 compressor/limiter bands into the final look-ahead limiter. Note that it is possible to reduce the level below “0” as well as increase it.

Level [Low, Middle, High]

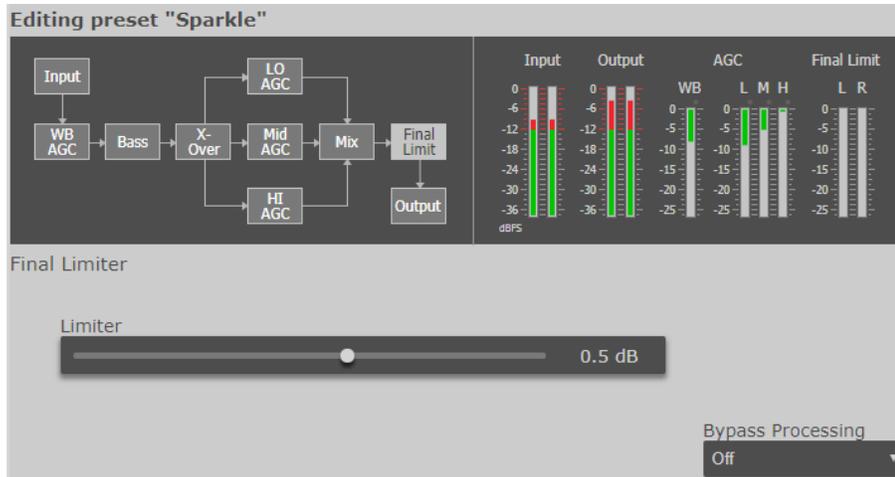
While some EQ adjustment can be done at this stage, be careful not to turn one band’s mix level up too much higher than the others. This will cause too much of that band’s signal to hit the final limiter, causing possible undesired effects such as overload (excessive limiting of one band’s audio) or pumping.

It is generally better to leave these set at their default settings and use the Bass and Drive controls to EQ the sound to your liking.

Then, if further correction is required, you can reduce the level of a particular band here if needed, such as reducing the High mixer if the audio sounds too bright.

Try to use the mixer only to reduce rather than boost as a rule-of-thumb.

Final Limiter



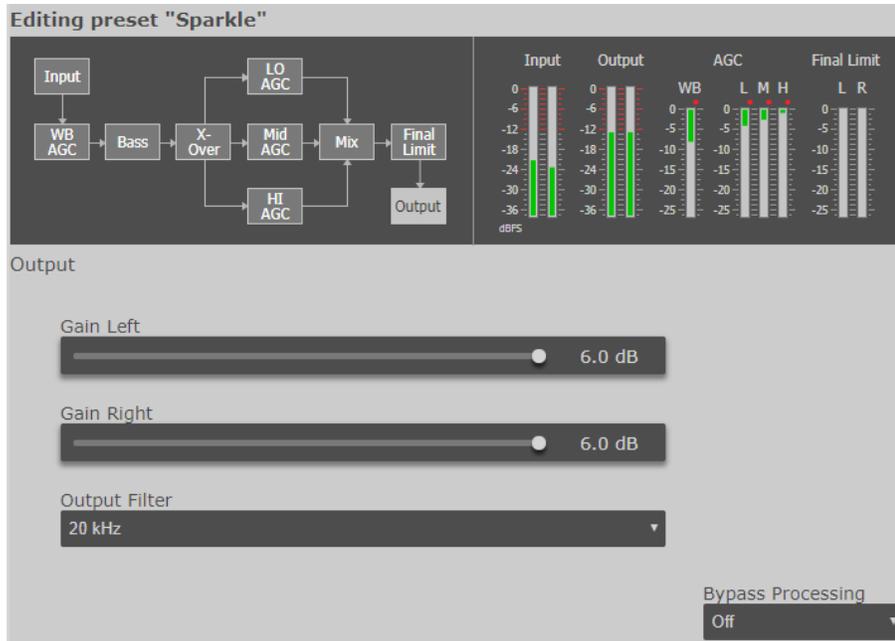
The final limiter is a fast, look-ahead peak limiter that provides final peak control to the processed audio.

Limiter

Generally, set the Limiter level control until you see about 2 to 4dB of gain-reduction on the Final Limit meters with normal program audio playing.

Too low a setting will decrease loudness while too high a setting can sound too dense and over-processed.

Output



Gain [Left & Right]

The Gain Left and Gain Right controls set the final peak output level that feeds the codec. Normally, this can be set to just under the maximum setting of 6.0dB. This setting corresponds to about 0dBFS. Normally, a setting of 5.8dB should be fine, but be sure to listen for clipping distortion through the codec and adjust as needed.

Output Filter

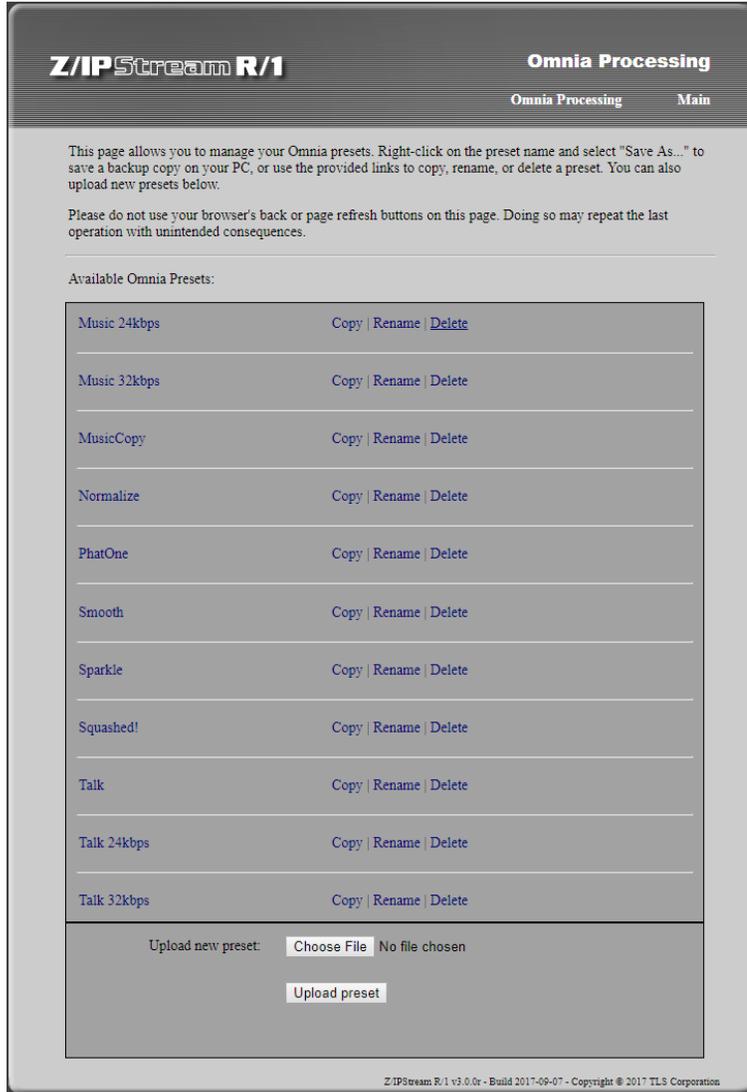
Dynamics processing is an extremely useful tool for reducing audible artifacts from lossy-compression codecs. High frequencies tend to be the first perceptible annoyances, followed by that swishy-swirly water like sound common at lower bitrates. These artifacts can be reduced, and in some cases eliminated, when audio processing is applied as a tool.

Generally, as codec bitrates are reduced, so is the audio bandwidth. Z/IPStream R/1 contains a selectable low pass filter in the output section. Reducing the audio bandwidth by using a lower frequency setting can help reduce the audible side-effects of low bitrate coding. Additionally, reducing the High band MIX setting can help.

Also, operating the HF Band in the slow release time will reduce HF density which improves intelligibility as the encoder masking algorithm is not loaded as heavily with HF content.

Managing Presets

From the Omnia Processing menu, click Manage Presets to open the preset management window.



In the example shown here, we made some changes to the PhatOne preset and then clicked “Save”.

Now, here in the Manage Presets window you will see that the PhatOne preset line has an additional choice: “Restore”. If we were to click on “Restore” now, before saving a copy, the original factory PhatOne preset will be restored and the changes we made would be lost.

So if you do make some changes to a preset and save them, it would be a good idea to click “Copy” to save the preset with your changes to a new name. This will create a user preset with that name.

Once that is done, you can Restore the factory preset and can compare it to the new user preset that contains the changes you made from the original.

Note that the Restore option will only appear when you save custom changes to a factory preset or delete a factory preset. This means that factory presets can always be restored, even if you delete them to remove them from the preset list. This IS NOT the case for user presets. Deleting a user preset is final.

A1 Telos Alliance Warranty

Telos Alliance Limited Warranty



For the latest Telos Alliance warranty, visit: telosalliance.com/warranty

A2 Specifications

Audio Coding

Codecs:

- ◆ MP3: 16 to 320 kbps
- ◆ AAC-LC: 24 to 320 kbps
- ◆ HE-AAC: 24-96 kbps
- ◆ HE-AAC v2 (aacPlus) : 24-96 kbps

AAC Transport Modes:

- ◆ ADTS
- ◆ ADTS-CRC
- ◆ ADIF
- ◆ RAW METADATA FORMATS:
 - ◆ Character Parser Sample
 - ◆ Line Parser Sample
 - ◆ Nexgen Audio Sense
 - ◆ Simian Template 1
 - ◆ XML Parser Sample
 - ◆ XML-Jazler
 - ◆ XML-Jazler2
 - ◆ XML-MediaTouch
 - ◆ XML-MediaTouch2
 - ◆ XML-Sample2
 - ◆ XML-Zetta
 - ◆ User-definable

Analog Audio Input

- ◆ Analog: Balanced XLR, +4 dBu
- ◆ Input Impedance: 6K Ohm differential
- ◆ Analog to Digital Converter: 24bits

Analog Audio Output

- ◆ Analog: Balanced XLR
- ◆ Output Clipping: +22dBu
- ◆ Output Impedance: 50 Ohm differential
- ◆ Digital to Analog Converter: 24bits

Digital Audio Inputs and Outputs

- ◆ Reference Level: +4 dBu (-20 dB FSD)
- ◆ Impedance: 110 Ohm, balanced
- ◆ Signal Format: AES3 (AES/EBU)
- ◆ AES3 Input Compliance: 24-bit with sample rate conversion
- ◆ AES3 Output Compliance: 24-bit
- ◆ Internal Sampling Rate: 48 kHz
- ◆ Input Sample Rate: 32 kHz to 192 kHz
- ◆ Output Sample Rate: 48 kHz
- ◆ A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- ◆ D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- ◆ Digital: Livewire AoIP, via LAN port

Audio Performance

- ◆ THD+N: < 0.03% @ +12dBu, 1 kHz Sine
- ◆ Freq Response: +/- 1dB 25– 20 kHz
- ◆ Head Room: 18dB
- ◆ Dynamic Range: > 87dB Unweighted > 90 dB "A" Weighted
- ◆ Crosstalk: > 80 db

Remote Control

- ◆ LAN via built-in Websserver

Power

- ◆ Internal supply, 85–250 VAC auto-switching, 50–60 Hz.
- ◆ Power consumption: 14.2 Watts

Dimensions

- ◆ 19" (48.3 cm) standard rack mounting front panel
- ◆ 1.75" (4.5 cm) height, 6.5" (16.51 cm) depth
- ◆ Shipping Weight: 8 lbs. (3.62 kg)

Z/IPStream R/1 Self Test

1. Press and hold the Headphone Volume Knob in during power up until the screen activates.
2. Release Headphone Volume Knob and then press the Control Knob.
3. Activate all front panel controls to test function.
4. Once complete, the "Frontpanel" status will confirm OK to indicate controls are good.

THE TELOS ALLIANCE®

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