softGear

Radio and Streaming Audio Processor User Guide

Thank You for Choosing Ross

You've made a great choice. We expect you will be very happy with your purchase of Ross Technology.

Our mission is to:

1. Provide a Superior Customer Experience
    - offer the best product quality and support
2. Make Cool Practical Technology
    - develop great products that customers love

Ross has become well known for the Ross Video Code of Ethics. It guides our interactions and empowers our employees. I hope you enjoy reading it below.

If anything at all with your Ross experience does not live up to your expectations be sure to reach out to us at *solutions@rossvideo.com*.

David Ross
CEO, Ross Video
dross@rossvideo.com

Ross Video Code of Ethics

Any company is the sum total of the people that make things happen. At Ross, our employees are a special group. Our employees truly care about doing a great job and delivering a high quality customer experience every day. This code of ethics hangs on the wall of all Ross Video locations to guide our behavior:

1. We will always act in our customers' best interest.
2. We will do our best to understand our customers' requirements.
3. We will not ship crap.
4. We will be great to work with.
5. We will do something extra for our customers, as an apology, when something big goes wrong and it's our fault.
6. We will keep our promises.
7. We will treat the competition with respect.
8. We will cooperate with and help other friendly companies.
9. We will go above and beyond in times of crisis. *If there's no one to authorize the required action in times of company or customer crisis - do what you know in your heart is right. (You may rent helicopters if necessary.)*

RSAP · User Guide

- Ross Part Number: **3900DR-304-01**
- Release Date: June 29, 2020.

The information contained in this Guide is subject to change without notice or obligation.

Copyright

Patents

Patent numbers US 7,034,886; US 7,508,455; US 7,602,446; US 7,802,802 B2; US 7,834,886; US 7,914,332; US 8,307,284; US 8,407,374 B2; US 8,499,019 B2; US 8,519,949 B2; US 8,743,292 B2; GB 2,419,119 B; GB 2,447,380 B; and other patents pending.

Notice

The material in this manual is furnished for informational use only. It is subject to change without notice and should not be construed as commitment by Ross Video Limited. Ross Video Limited assumes no responsibility or liability for errors or inaccuracies that may appear in this manual.

Safety Notices

Refer to the "**Important Regulatory and Safety Notices**" document that accompanied your product.

EMC Notices

US FCC Part 15

This equipment has been tested and found to comply with the limits for a class A Digital device, pursuant to part 15 of the FCC Rules.

These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a Commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

Notice — *Changes or modifications to this equipment not expressly approved by Ross Video Ltd. could void the user's authority to operate this equipment.*

Canada

This Class "A" digital apparatus complies with Canadian ICES-003 and part 15 of the FCC Rules.

Cet appareil numerique de la classe "A" est conforme a la norme NMB-003 du Canada.

European Union

This equipment is in compliance with the essential requirements and other relevant provisions established under regulation (EC) No 765/2008 and Decision No 768/2008/EC referred to as the “New Legislative Framework”.

> **Warning** — *This equipment is compliant with Class A of CISPR 32. In a residential environment this equipment may cause radio interference.*

Australia/New Zealand

This equipment is in compliance with the provisions established under the Radiocommunications Act 1992 and Radiocommunications Labeling (Electromagnetic Compatibility) Notice 2008.

International

This equipment has been tested under the requirements of CISPR 22:2008 or CISPR 32:2015 and found to comply with the limits for a Class A Digital device.

> **Notice** — *This is a Class A product. In domestic environments, this product may cause radio interference, in which case the user may have to take adequate measures.*

Maintenance/User Serviceable Parts

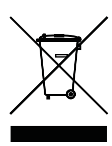
Routine maintenance to this product is not required. This product contains no user serviceable parts. If the module does not appear to be working properly, please contact Technical Support using the numbers listed under the “**Contact Us**” section of this manual. All openGear products are covered by a generous 3-year warranty and will be repaired without charge for materials or labor within this period. See the “**Warranty and Repair Policy**” section in this manual for details.

Environmental Information

The equipment may contain hazardous substances that could impact health and the environment.

To avoid the potential release of those substances into the environment and to diminish the need for the extraction of natural resources, Ross Video encourages you to use the appropriate take-back systems. These systems will reuse or recycle most of the materials from your end-of-life equipment in an environmentally friendly and health conscious manner.

The crossed-out wheeled bin symbol invites you to use these systems.



If you need more information on the collection, reuse, and recycling systems, please contact your local or regional waste administration. You can also contact Ross Video for more information on the environmental performances of our products.

Company Address

Ross Video Limited
8 John Street
Iroquois, Ontario
Canada, K0E 1K0

Ross Video Incorporated
P.O. Box 880
Ogdensburg, New York
USA 13669-0880

General Business Office:	(+1) 613 • 652 • 4886
Fax:	(+1) 613 • 652 • 4425
Technical Support:	(+1) 613 • 652 • 4886
After Hours Emergency:	(+1) 613 • 349 • 0006
E-mail (Technical Support):	techsupport@rossvideo.com
E-mail (General Information):	solutions@rossvideo.com
Website:	http://www.rossvideo.com

Contents

Introduction

This guide covers the installation, configuration, and use of the Radio and Streaming Audio Processor (RSAP). The following chapters are included:

- "**Introduction**" summarizes the guide and provides important terms, and conventions.
- "**Before You Begin**" summarizes the features of the RSAP.
- "**Audio Processors Overview**" outlines the audio processor features, powered by Orban technology, of the RSAP.
- "**Hardware Overview**" presents information on the RSAP hardware.
- "**Physical Installation**" provides additional information for installing the RSAP.
- "**Cabling**" outlines how to connect to your facility network. Information is also given about connecting peripheral devices you will need for initial setup.
- "**Initial Connection**" provides instructions for configuring the initial IP address for the RSAP.
- "**Using DashBoard**" provides instructions for launching DashBoard, and accessing the RSAP interfaces in DashBoard.
- "**Updating the Network Settings**" provides instructions for configuring the RJ45 ports of the RSAP for communication with your facility network and streaming devices.
- "**Configuring the PTP Settings**" provides instructions for configuring the RSAP to use Precision Time Protocol, and specifying a system clock reference source.
- "**Configuring the AES67 Receiver**" outlines the steps required to configuring the AES67 receiver streams on the RSAP.
- "**Configuring the AES67 Senders**" outlines the steps required for configuring the AES67 sender streams on the Radio and Streaming Audio Processor (RSAP).
- "**Audio Processing Concepts**" outlines the audio processing options of the RSAP.
- "**Using Audio Presets**" outlines how to recall, edit, and save the available presets for audio processing.
- "**Automatic Gain Controls**" outlines how to use the RSAP as a studio Automatic Gain Control AGC to protect a studio-to-transmitter link.
- "**Stereo Controls**" outlines the stereo enhancer and synthesizer features of the RSAP.
- "**Equalizer Controls**" summarizes the available equalization settings of the audio processor.
- "**Multiband Controls**" describes the Multiband settings of the audio processor.
- "**Compressor Controls**" outlines the compression threshold settings for the audio processor.
- "**Bandmix Controls**" outlines output mix controls of the audio processor.
- "**Limiters Controls**" summarizes the bass clip threshold, bass pre-limit mode, multiband final limit drive, Transient Enhance, and MX Limiter settings.
- "**Audio Meters in DashBoard**" provides a brief overview of the audio meter panel in the DashBoard window.
- "**Upgrading the Software**" provides instructions on how to upgrade the RSAP software via DashBoard.
- "**DashBoard Interface Overview**" summarizes the menus and parameters of the RSAP tabs in DashBoard.
- "**Technical Specifications**" provides the specifications for the RSAP.
- "**Service Information**" provides information on the warranty and repair policy for your RSAP.
- "**Glossary**"provides a list of terms used throughout this guide.

Related Publications

It is recommended to consult the following Ross documentation before installing and configuring your RSAP:

- ***DashBoard User Guide***, Ross Part Number: 8351DR-004

- ***RSAP Quick Start Guide***, Ross Part Number: 3900DR-306

Documentation Conventions

Special text formats are used in this guide to identify parts of the user interface, text that a user must enter, or a sequence of menus and sub-menus that must be followed to reach a particular command.

Interface Elements

Bold text is used to identify a user interface element such as a dialog box, menu item, or button. For example:

In the **Network** tab, click **Apply**.

User Entered Text

Courier text is used to identify text that a user must enter. For example:

In the **Language** box, enter `English`.

Referenced Guides

Text set in bold and italic represent the titles of referenced guides, manuals, or documents. For example:

For more information, refer to the ***DashBoard User Manual***.

Menu Sequences

Menu arrows are used in procedures to identify a sequence of menu items that you must follow. For example, if a step reads "**File** > **Save As**," you would click the **File** menu and then click **Save As**.

Important Instructions

Star icons are used to identify important instructions or features. For example:

✶ Contact your IT department before connecting to your facility network to ensure that there are no conflicts. They will provide you with an appropriate value for the IP Address, Subnet Mask, and Gateway for your device.

Contacting Ross Video Technical Support

At Ross Video, we take pride in the quality of our products, but if problems occur, help is as close as the nearest telephone.

Our 24-hour Hot Line service ensures you have access to technical expertise around the clock. After-sales service and technical support is provided directly by Ross Video personnel. During business hours (Eastern Time), technical support personnel are available by telephone. After hours and on weekends, a direct emergency technical support phone line is available. If the technical support person who is on call does not answer this line immediately, a voice message can be left and the call will be returned shortly. This team of highly trained staff is available to react to any problem and to do whatever is necessary to ensure customer satisfaction.

- **Technical Support**: (+1) 613-652-4886
- **After Hours Emergency**: (+1) 613-349-0006
- **E-mail**: techsupport@rossvideo.com
- **Website**: http://www.rossvideo.com

Before You Begin

If you have questions pertaining to the operation of the Radio and Streaming Audio Processor (RSAP), contact us at the numbers listed in the section “**Contacting Ross Video Technical Support**”. Our technical staff is always available for consultation, training, or service.

General Features

This section provides a general summary of the RSAP features:

- Ability to pre-process audio in real time for consistency and loudness before it is transmitted or recorded
- Support for up to 8 audio processors which may be any combination of mono/stereo and surround processors
- Incorporates modern “target loudness” concepts (EBU R128 and ATSC A/85) using the ITU-R BS.1770 loudness model
- Allows you to easily set and verify the target loudness of the output
- Compatible with Ravenna® audio-over-IP input/output
- Compatible with all Microsoft® Windows® sound devices with stable drivers supporting the Windows® WASAPI standard
- Precisely controls peak levels to prevent over-modulation or codec overload
- Provides pre-emphasis limiting for the two standard pre-emphasis curves of 50us and 75us
- Controls audio bandwidth to accommodate the transmitted sample frequency
- Includes a sweepable gentle-slope lowpass filter (6-24dB/octave) or a sweepable highpass filter with four selectable slopes
- Separate cutoff frequencies and slopes for music and speech modes
- Provides a DC removal filter with a 0.1Hz - 3dB low frequency cutoff to remove DC offset from source material
- Includes standard presets that accommodate almost any programming format
- Includes a Bypass Test mode facilitates broadcast system test and alignment
- Allows audio processing to be enabled and disabled on-air via a delay-matched pass-through mode
- Includes a built-in line-up tone generator to facilitate quick and accurate level setting
- Support for up to 32 audio channels
  - 1ms (or 125µs) packet time with flow containing up to 8 channels
  - 48KHz L24 audio
- Support for the following PTP profiles:
  - SMPTE ST 2059 profile
  - IEEE 1588 default profile
  - AES67 profile
- RTP timestamp preserve or re-stamp
- Latency < 30ms + link offset
- Independent management and media 1000 BaseT Gigabit Ethernet
- Management under DashBoard control

DashBoard Interfaces

The RSAP requires an Ethernet network connection between it and a computer that will run the DashBoard client. The DashBoard client software enables you to monitor and control DashBoard Connect compatible devices from a computer.

The RSAP includes DashBoard interfaces for configuration and operation. The interfaces are accessed by expanding the RSAP node in the DashBoard Tree View and selecting the appropriate sub-node.

For More Information on...

- displaying the DashBoard interfaces, refer to the chapter "**Using DashBoard**" on page 39.

What are Receivers, Senders, and Streams?

The following terms are used throughout this user guide:

Device

A physical, virtual, or software application that may include multiple sources, destinations, senders, or receivers.

Flow

The continuous raw media content. It can contain more than one essence (e.g. an audio flow can contain multiple channels).

Flows cannot generally be passed around natively, and need to be encapsulated in a stream. Flows from the same source are considered "editorially equivalent", but may be encoded differently.

Receiver

An element within a device that receives exactly one stream, which contains one flow from a network.

Sender

An element within a device which presents exactly one flow, packaged as a stream onto a network.

Stream

One flow, encapsulated within a transport protocol (e.g. SMPTE ST 2110-30 Audio (AES67)).

Network Stream Group

A network stream group allows you to create a grouping of streams on the network into a unit that can be mapped to an audio channel. Network stream groups are required when:

- the output has more channels than a stream on the network and the user wants to drive all channels;
- the user wants to perform complicated audio shuffling

Configuration Overview

Figure 2.1 provides a generalized work-flow of installing and configuring your RSAP.

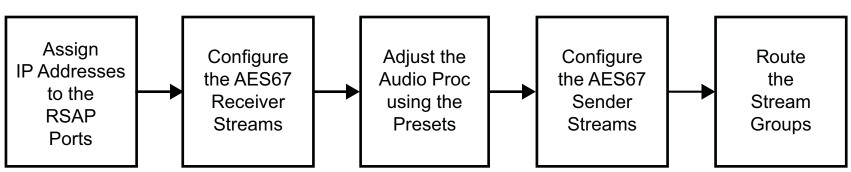


Figure 2.1 Configuration Work-flow

Audio Processors Overview

This chapter summarizes the audio processing features of the RSAP.

For More Information on...

- the specific processing options in DashBoard, refer to "**Audio Processing Concepts**" on page 55.

Functional Block Diagram

Figure 3.1 provides a general functional block diagram for the audio processing features of the RSAP.

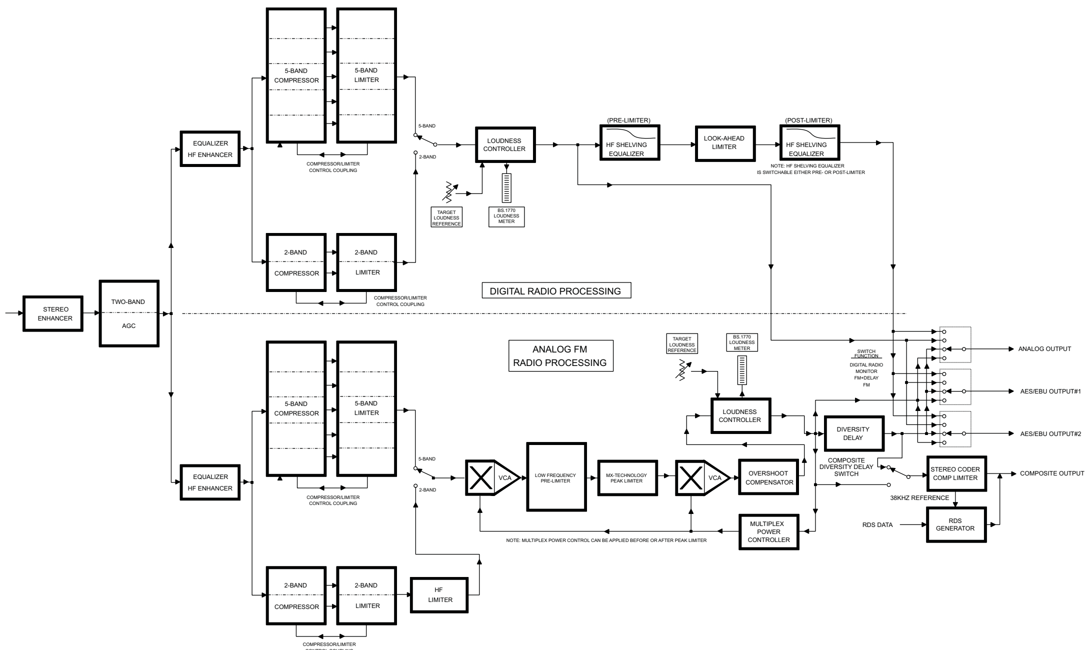


Figure 3.1 Functional Block Diagram — Audio Processing

Audio Processing Features

This section provides a brief overview of the main audio processing features.

Increase the Density and Loudness of Program Material

The RSAP can increase the density and loudness of the program material by multi-band compression and sophisticated peak limiting, improving the consistency of the station's sound and increasing loudness and definition remarkably, without producing unpleasant side effects.

Automatic Left/Right Phase-skew Corrector

This feature can eliminate comb-filtering artifacts in a mono downmix and reduce multi-path distortion by minimizing energy in the stereo sub-channel, which carries the L–R signal.

Bass can be made Monophonic

Provides crossover frequencies of 80Hz or 100Hz.

Automatic Gain Control (AGC)

A two-band gain setting provides phase-linear crossover, adjustable band coupling, and window gating compensates for widely varying input levels. The AGC rides gain over an adjustable range of up to 25dB, compressing dynamic range and compensating for operator gain-riding errors and for gain inconsistencies in automated systems.

Five-Band Compression

This features includes selectable phase-linear and all-pass crossover topologies to provide a consistent, “processed” sound, free from undesirable side effects.

Band-coupling Settings

This allows the gain differences between adjacent bands of the five-band compressor to be constrained to any desired value, while preserving as much of the frequency balance of the original program material as desired unless doing so would otherwise cause objectionable spectral gain inter-modulation artifacts.

Parametric Soft-knee Compression Curves with Adjustable Ratio

This allows you to fine-tune the audio to your exact requirements and make the RSAP an excellent mastering processor.

The five-band compressor can be operated in both in-line and parallel modes. Parallel mode drives downstream processing with the sum of the input and output of the multi-band compressor. This increases the loudness of quiet material while preserving the dynamics and punch of loud material. Typical applications include preparing classical music for in-flight entertainment or other noisy environments.

Stereo Enhancer

This feature increases the energy in the stereo difference signal (L-R) whenever a transient is detected in the stereo sum signal (L+R). Gating circuitry prevents over-enhancement and undesired enhancement on slightly unbalanced mono material.

Equalizers

The shelving bass equalizer and four-band parametric equalizer let you color the audio to your exact requirements. To facilitate A/B comparisons, equalizers can be bypassed individually and globally.

Dynamic High-Frequency Enhancer

The enhancer constantly monitors the ratio of HF to broadband energy in the incoming audio and can automatically re-equalize it to achieve a target balance between broadband and HF energy.

Subharmonic Synthesizer

Generates punchy bass from bass-shy material. It is particularly useful for older music recordings.

MX Peak Limiter Technology

For applications where target loudness is below approximately -12LkFS/LUFS, you can reduce CPU load by approximately 50% without compromising audio quality (compared to full MX limiter operation) by disabling the MX limiter and instead using our smooth, low-IM look-ahead limiter for the very light peak limiting required at these low Loudness Levels. This is the normal mode of operation for sound-for-picture applications, where target loudness is typically -23LUFS or -24LUFS and peak limiting rarely occurs at all.

In all modes of operation, the peak limiter offers “true peak” setting by oversampling the peak limiter’s side-chain at 192kHz or higher. This prevents clipping in a playback device’s analog signal path by predicting and settingling the analog peak level following the playback device’s reconstruction filter.

Without true peak setting, analog clipping can occur even if all peak values of the digital samples are below 0dBFS. This phenomenon has also been termed “0dBFS+.” Thanks to true peak setting, sample rate conversion, unless it removes high frequency program energy or introduces group delay distortion, cannot cause sample peaks to increase more than 0.5dB. For example, sample rate conversion from 48 kHz to 44.1 kHz is highly unlikely to cause sample peak clipping in the 44.1kHz audio data.

Accuracy is typically 0.15dB even with heavy peak limiting, so the output level can be set to -0.2dBFS without true peak levels exceeding 0dBFS.

Stereo Synthesizer

With excellent downmix compatibility. “Wide” and “Narrow” modes can create an attractively spacious stereo or surround output from mono program material. Synthesis can be invoked manually, or activated automatically by sensing silence in the right channel input.

Third-generation CBS Loudness Controllers™ for DTV Applications

Separate loudness settings are available in both the surround and downmix processing chains. Reduce annoyance more than simple loudness setting alone, and without audible gain pumping. Attack time is fast enough to prevent audible loudness overshoots, so the setting is smooth and unobtrusive. Material processed by the CBS Loudness Controller has been shown to be well setting when measured with a longterm loudness meter using the BS.1770 standard.

BS.1770-4 and CBS Loudness Meters™

Measure the subjective loudness of the RSAP output and are displayed in DashBoard. When downmix processing is active, there are two independent loudness settings and two loudness meters available.

BS.1770 Safety Limiter

When enabled, it can further improve the measured performance using the BS.1770 meter and which was added for the benefit of organizations with strict objective limits on the indication of a BS.1770-2 meter regardless of the actual subjective loudness as determined by human listeners.

Pure Peak Limiting Preset

Performs high quality peak limiting in mastering applications.

AGC Applications

The RSAP can be used as a studio AGC (including peak limiting) to protect a studio-to-transmitter link (STL), optimally using the native dynamic range of the STL.

Uses “Multi-rate” Digital Signal Processing

Internal processing always occurs at sample rates from 48kHz to 256kHz as needed, and provides 20kHz audio bandwidth (unless specifically constrained by the user-adjustable low-pass filter). Built-in high-quality synchronous sample rate converters facilitate interfacing with 44.1kHz, 96kHz and 192kHz systems.

Uses 32bit or 64bit Floating-point Arithmetic

Can interface to 16bit and 24bit Microsoft® Windows® audio devices.

Audio Processing Signal Flow

The signal flows through RSAP in order through the following blocks:

- DC Removal
- Input Conditioning, includes configurable high-pass filtering, low-pass filtering and phase rotation

- Stereo Synthesizer, auto-detecting and configurable
- Mono Bass processing blends the left and right channels below 80Hz or 100Hz (stereo processing only)
- Subharmonic Synthesizer creates energy one octave below program energy in the range of 50-90Hz or 60-120Hz when such energy is not present at the input and when music is detected
- Left/Right Phase Skew Correction corrects phase shifts between the left and right stereo channels that could otherwise cause comb filtering in the mono sum (stereo processing only)
- Stereo Enhancement uses upward expansion of the stereo difference signal as triggered by transients in the stereo sum signal
- Two-Band Gated AGC, with target-zone window gating and silence gating
- Equalization, including high-frequency enhancement
- Downward Expander in five bands
- Multiband Compression in five bands
- Automatic Loudness Control plus a BS.1770 Safety Limiter
- Dual-mode automatic stereo upmixer
- Peak Limiting of several varieties

Sample Rate Conversion

The base sample rate of the RSAP internal processing is 48kHz even when the input and output are configured for 44.1, 96 or 192kHz; in these cases, synchronous sample rate conversion (SRC) occurs at the input and output. This 48kHz rate accommodates a 20kHz audio bandwidth with a comfortably wide 4kHz transition band for the anti-aliasing filter.

Moreover, the noise and distortion produced by a given digital filter at 48kHz is about 6dB lower than the noise and distortion produced by a filter having the same frequency response but operating at 96kHz. RSAP uses many digital filters, both in its equalizer section and for the crossovers in the multi-band compressor.

Hence, we believe that 48kHz is the ideal rate for RSAP audio processing. Higher sample rates would not only increase CPU usage but would decrease the signal-to-noise ratio of the processing.

Input Conditioning

- A high-pass filter with a cutoff frequency of 0.15Hz removes DC offset from source material without causing significant tilt of low-frequency squarewaves.
- A high-pass filter, tunable between 20 and 200Hz and with selectable 6, 12, 18, or 24dB/octave slopes, is useful for production applications where it is necessary to remove low frequency rumble from a recording, and in news/talk broadcasting formats.
- A gentle-slope (6, 12, 18, or 24dB/octave) low-pass filter, tunable between 4 kHz and 15 kHz and with selectable 6, 12, 18, or 24 dB/octave slopes, is useful for production and mastering.
- A configurable phase rotator makes speech more symmetrical, reducing its peak-to-average ratio by as much as 6dB without adding nonlinear distortion. Hence, phase rotation can be very useful for loudness processing of speech.
- A very steep low-pass filter can be tuned between 10kHz and 20kHz (where 20kHz is no filter). Set it to complement the bandwidth of the transmission channel that RSAP is driving.

Stereo Synthesizer

The stereo synthesizer creates an artificial stereo difference signal (L-R) by passing the mono input through a multistage all-pass filter. After matrixing with the original mono input (which is the L+R signal) to produce the synthesized left and right channels, the result is a “complementary comb filter” whose notches are spaced in frequency in an approximately logarithmic manner. Because only the L-R signal is created artificially, it cancels out of a mono mix-down, making the synthesizer’s output completely mono-compatible. This processing is only available in stereo mode.

✶ The synthesizer can be invoked manually or by automatic detection of silence on the right input channel.

Mono Bass Processing

This applies a steep-slope 80Hz or 100Hz high-pass filter to the stereo difference signal (L-R). A compensating delay is applied to the L+R signal, making the bandwidth of the transition between mono and well-separated stereo as narrow as possible. This processing is only available in stereo mode.

✶ We strongly recommend activating mono bass processing when the stereo synthesizer is active.

Subharmonic Synthesizer

The subharmonic synthesizer generates subharmonics of fundamental frequencies in the 50-90Hz or 60-120Hz range, user-selectable via the Sub Harmonic Cutoff Frequency setting.

The subharmonics are one octave below the frequencies from which they are generated (i.e., 25 to 45Hz or 30 to 60Hz) and track the levels of their generating frequencies.

If input program material below 45 or 60Hz is present, the subharmonic synthesizer automatically reduces the level of the synthesized subharmonics to prevent excess build-up of energy below 45 or 60Hz.

To prevent introducing unnatural coloration in speech, the subharmonic synthesizer is disabled when the automatic speech/music detector detects speech. This is particularly critical when the Sub Harmonic Cutoff Frequency setting is set to 120Hz because the synthesizer can cause “thumping” sounds. It is therefore important to keep raw speech in the center of the stereo image, because this is one of the criteria that the speech/music detector uses to discriminate between speech and music.

Stereo Enhancement

The RSAP provides a stereo enhancement which increases the energy in the stereo difference signal (L-R) whenever a transient is detected in the stereo sum signal (L+R). By operating only on transients, this increases width, brightness, and punch without unnaturally increasing reverberation (which is usually predominantly in the L-R channel).

✶ Use stereo enhancement with care if you are driving a low bit-rate codec. At low bit-rates, these codecs use various parametric techniques for encoding the spatial attributes of the sound field. Stereo enhancement can unnecessarily stress this encoding process.

Left/Right Phase Skew Correction

The phase skew corrector maximizes the quality of a mono mix-down that might occur in a receiver or player device. At higher frequencies (where audible combined filtering of the mono sum is most likely to occur), the corrector removes phase differences between the left and right channels, converting the HF signal into “intensity stereo” while preserving phase differences at lower frequencies where these differences are important for psychoacoustic “envelopment.” The Phase Correct Crossover setting of the active preset sets the crossover frequency above which phase correction occurs, and IN/OUT enables or disables the phase corrector via a delay-matched cross-fade.

This process can not only correct problems due to phase skew between the left and right channels of an analog recording due to head gap misalignment, it can also correct comb filtering caused by spaced microphones feeding the left and right channels, which can occur on drum kits and other sources that have been multi-mixed.

When RSAP is used as a pre-processor for an FM audio processor, the phase corrector minimizes the amount of energy in the stereo sub-channel, which consequently minimizes multi-path distortion without compromising stereo separation. It can allow more stereo enhancement to occur for a given amount of multi-path distortion. The process also minimizes the amount of peak overshoot during SSB/VSB operation of the stereo encoder (if available), thus minimizing the amount of composite limiting needed to constrain peak modulation to 100%.

✶ Because the phase skew corrector can subtly alter the stereo spatial effect, we recommend using it only as necessary (for example, with formats that play older recordings from the analog era). It can be smoothly activated and disabled via a delay-matched cross-fade, so it is practical to do live switching between a preset with the process active and one where it is inactive.

Because it adds considerable delay and uses a significant amount of CPU power, the phase skew corrector can be bypassed completely. If you are not using it and do not need to activate it smoothly “on-air,” bypass it.

Two-Band Gated AGC

The AGC is a two-band device, using a patented “master/bass” band coupling and a linear-phase crossover.

In Stereo mode, there are two gain-setting side-chains, one for each stereo channel. To preserve RMS operation, the Multiband MAX Delta Gain Rate setting operates a constant-power, symmetrical panoramic potentiometer (panpot). When this setting is set to 0, it applies equal amounts of left and right energy to both side-chain, which causes the left and right side-chain to track each other, maintaining 100% stereo coupling. As this setting is set to higher values, it progressively applies less right-channel energy to the left-channel side-chain and vice-versa. Setting the setting to a higher value will partially correct stereo program material with left/right channel imbalances. Higher settings should be used with caution, as they can cause instability in the stereo image.

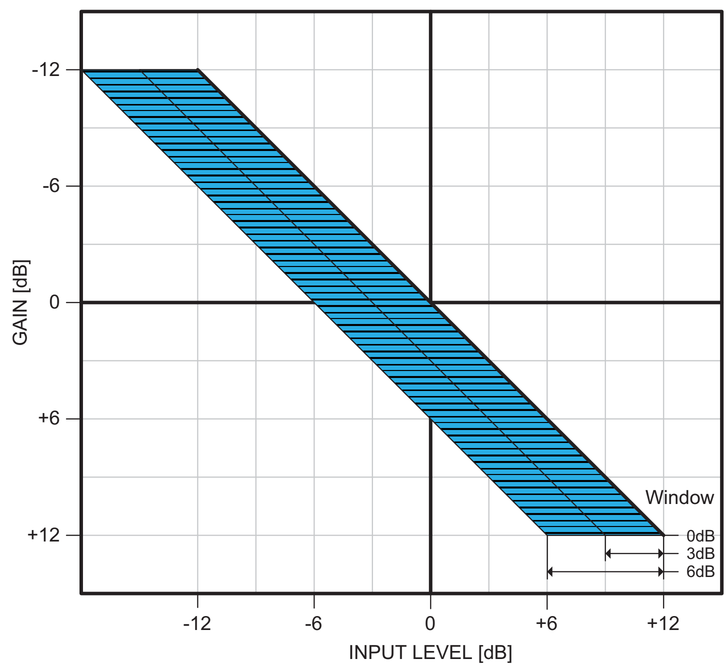


Figure 3.2 AGC Gain vs. Input Level Showing Window

The AGC contains a compression ratio setting that allows you to vary to ratio between 2:1 and essentially 1:1. Lower ratios can make gain riding subtler on critical formats like classical and jazz. See **Figure 3.3**.

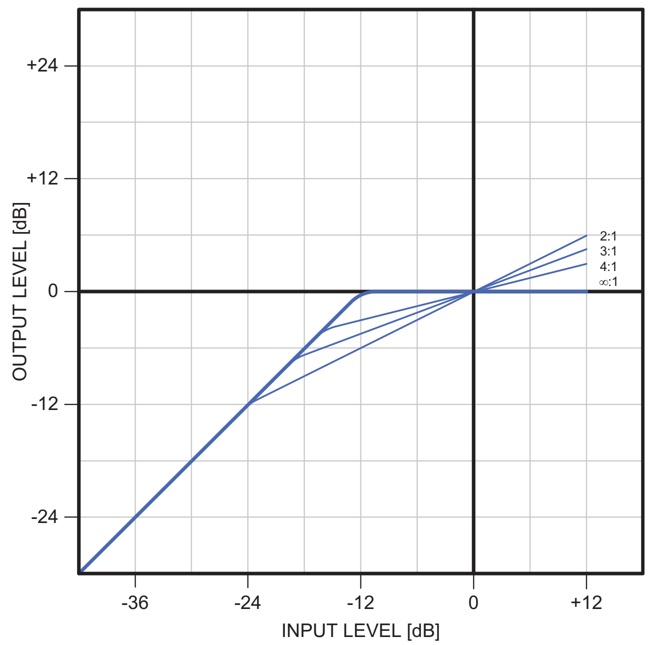


Figure 3.3 AGC Output vs. Input as a Function of the AGC Ratio Control

The AGC incorporates target-zone gating. If the input program material's level falls within a user-settable window (typically 3dB), the release time slows to a user-determined level. It can be slow enough (0.5 dB/second) to effectively freeze the operation of the AGC. This prevents the AGC from applying additional, audible gain riding to material that is already well settingled. It also lets you run the AGC with fast release times without adding excessive density to material that is already dense. **Figure 3.4** shows the behavior of the window for when the AGC Drive is set to 12dB.

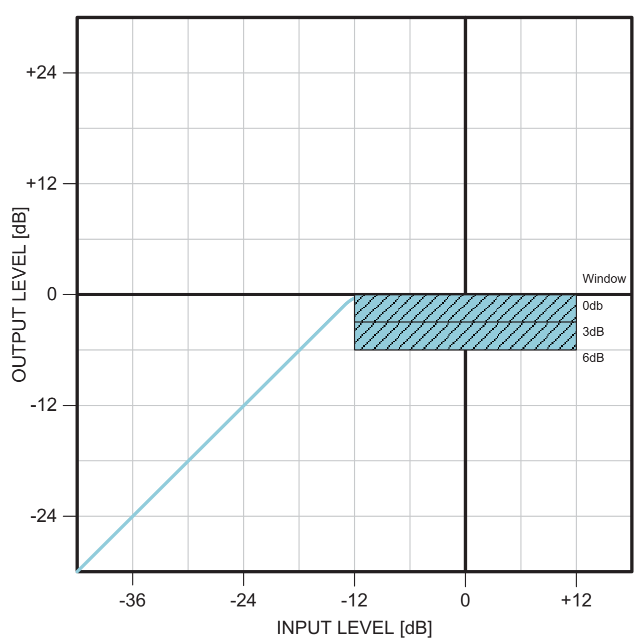


Figure 3.4 AGC Output vs. Input Showing Window

The AGC has a dedicated silence-gating detector whose threshold can be set independently of the silence gating applied to the multi-band compressor. In stereo mode, the gating detector is operated from the output of the left-channel panpot implemented by the Multiband MAX Delta Gain Rate setting, so if this setting is set for fully uncoupled operation (not recommended) and there is energy only in the right channel, then the gate will activate.

Equalization

The RSAP has a bass shelving equalizer, four bands of fully parametric bell-shaped EQ, and a dynamic high-frequency enhancer.

You can set the slope of the bass shelving EQ to 6, 12, or 18dB/octave, adjust the shelving frequency and set the amount of equalization.

The bass, midrange, and high frequency parametric equalizers have curves using a sophisticated, proprietary optimization program. The curves are matched to better than 0.15dB to ensure low noise and distortion.

The HF Enhancer is a program-setting HF shelving equalizer that intelligently and continuously analyzes the ratio between broadband and HF energy in the input program material. It can equalize excessively dull material without over-enhancing bright material. It interacts with the five-band compressor to produce sound that is bright and present without being excessively shrill.

The Brilliance setting adds additional gain before the Band 5 multi-band compressor, so it acts like a steep-slope shelving equalizer.

See **Figure 3.5** and **Figure 3.6** for illustrative frequency responses of the shelving and parametric equalizers.

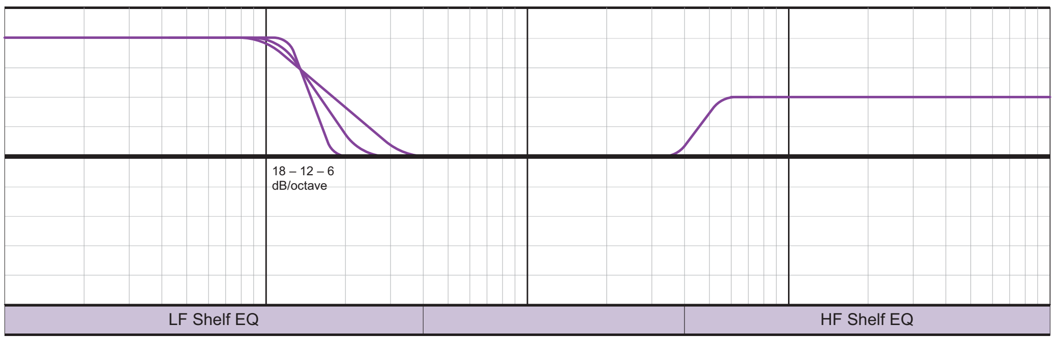


Figure 3.5 LF and Dynamic HF Shelving Equalizer Frequency Response

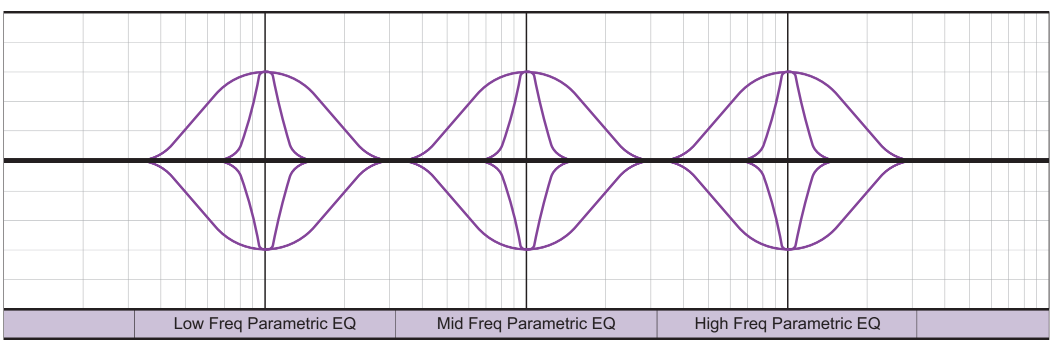


Figure 3.6 Parametric Equalizer Frequency Response for Narrow and Wide Bandwidths

Multiband Compressor/Limiter

The multi-band compressor/limiter operates in five frequency bands. Each band compressor has a Knee and Ratio setting. A soft knee and gentle ratio are particularly useful in production and mastering applications, allowing subtle compression that retains as much of the dynamics of the input program material as the operator desires. (The Soft Knee and Soft Knee MX presets are intended for mastering.) These features are also exploited in the TV 5B DRAMA preset.

Several band-coupling settings allow the gain reduction of a given band's compressor to be affected by the gain reduction in its neighboring band's compressor. These coupling settings allow anything from quasi-wideband compression to fully independent multi-band compression.

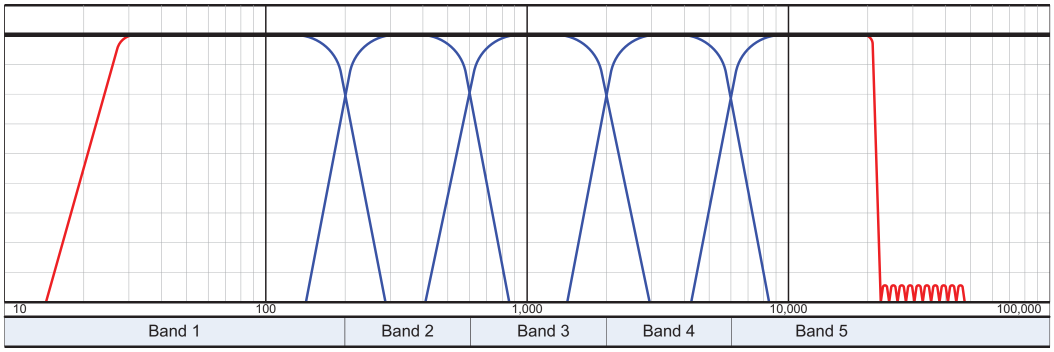


Figure 3.7 Multiband Compressor Crossover Filters (Illustrative)

Each band in the stereo multi-band compressor has two side-chains—one for the left and one for the right channel.

You can separately set the left/right coupling of each band anywhere from 100% stereo coupled to fully independent.

✶ Although the multi-band compressor's architecture has two side-chains, RSAP does not allow two different programs to be routed through a stereo Processor. If you need one or more channels of mono processing, use an instance of mono processing for each. This allows each Processor to have separate setup and processing settings applied to it.

In Stereo mode, there is one Loudness Level meter, AGC GATE indicator, and MB GATE indicator. However, you can still uncouple each band in both the AGC and multi-band compressors to a variable extent—anywhere from perfect stereo coupling to completely uncoupled operation. The coupling setting determines the maximum amount of gain difference permitted between the left and right channels in a given band and therefore the amount of stereo image shift permitted in each frequency band.

Parallel Compression

The parallel compression mode drives downstream processing with the sum of the input and output of the multi-band compressor. As compressor gain reduction increases, the compressor contributes less and less to the overall output. Parallel mode is mainly useful for classical music, preserving the dynamic impact of loud material while making quiet material more audible. Typical applications are preparing material for in-flight entertainment or other noisy environments.

There are two specific settings for parallel mode (in the Multiband tab): Compressor Mode [Inline, Parallel] and Compression Thresh Offset [0dB to –60dB, 1dB steps]. Additionally, in parallel mode the Multiband Drive setting is re-purposed so that it sets the amount of compressor output that is added to the compressor's input. Its setting indicates the amount of amplification of quiet material that occurs when the compressor produces no gain reduction. It does not affect the amount of compressor gain reduction.

The Compression Thresh Offset setting is only active in Parallel mode. It moves all compressor thresholds in the five-band compressor down by the same amount. It is needed because the Multiband Drive setting does not affect the amount of compressor gain reduction and because the individual bands' Compression Threshold settings do not have enough range to achieve the desired amount of gain reduction with all program material.

Parallel compression is usually used to bring up quiet material while minimally affecting loud material. Hence, the compressor threshold must be much lower than normal so that the compressor develops maximum gain reduction with high-level material. Adjust the Compression Thresh Offset setting so that the compressor gain reduction meters are active but on-scale during quiet material that you wish to make louder. Typical settings are –30 to –40 dB.

✶ It is normal for the five-band compressor gain reduction meters to indicate full-scale during loud material. (The maximum gain reduction is clamped internally to 40dB.)

The Compression Thresh Offset setting also scales the MB Gate Threshold and the Down Expander thresholds downward so that these features can be effective with source material that has low intrinsic noise, like most digital recordings. These features can help resist increasing audible noise (such as air conditioning rumble) during the quietest parts of the program.

✶ Parallel compression can be effective and unobtrusive even with faster settings of the Multiband RELEASE setting.

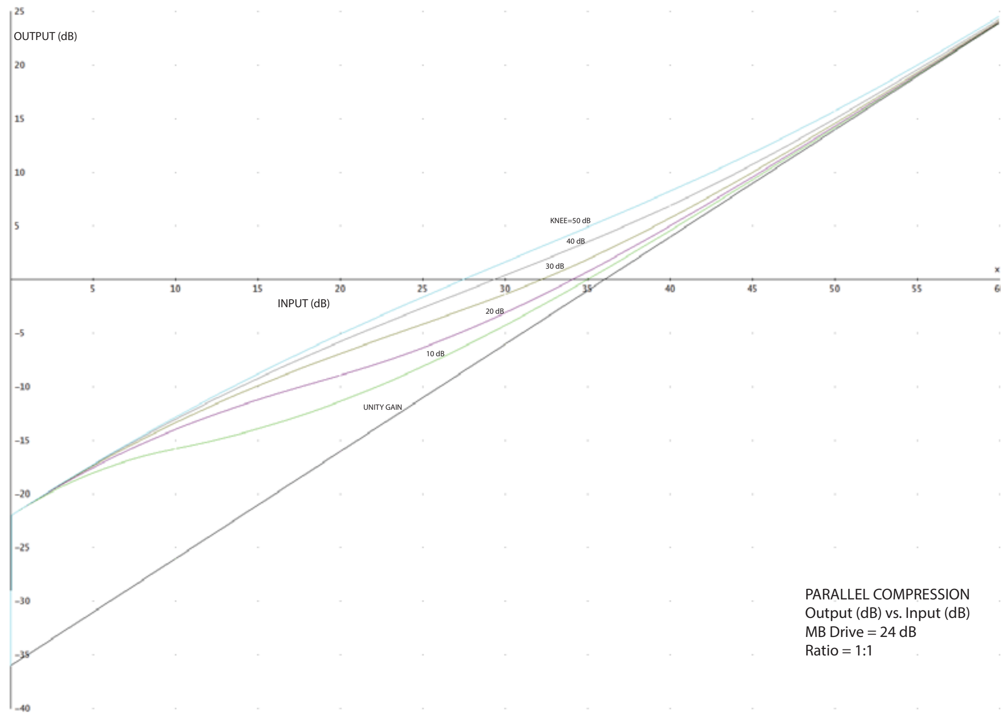


Figure 3.8 Parallel Compression Input vs. Output (dB)

To prevent phase cancellation between the compressor’s output and input, the summation occurs immediately after the compressor’s VCAs, and the five-band crossover is still in the signal path. ALLPASS crossover mode produces phase rotation, so LINEAR crossover mode is most appropriate for the kind of program material typically processed with parallel compression.

LFE Processing

In a surround Processor, the LFE channel has its own dedicated compressor with Attack Time and Threshold settings. Its other settings (such as RATIO and BREAKPOINT) track the settings of the B1 compressor. To constrain build-up of LFE energy, the B1>LFE COUPLING setting clamps the maximum gain of the LFE channel with respect to the band 1 compressor in five-band mode and the Bass band in two-band mode. For example, when this setting is set to 3dB (the default) and the B1 compressor exhibits 12dB of gain reduction, the LFE channel can never have less than 9dB of gain reduction even if the LFE compressor would have produced less than 9dB of gain reduction if uncoupled.

Peak Limiter

RSAP look-ahead peak limiter prevents overshoots by examining a few milliseconds of the unprocessed sound before it is limited. This way the limiter can anticipate peaks that are coming up. The limiter’s side-chain is oversampled to 192kHz to prevent significant overshoot from occurring after sample rate conversion or D/A conversion.

Unlike some familiar compressors and limiters (whose gain reduction is adjusted via threshold settings), the limiter’s threshold is fixed with respect to the input of RSAP digital output level setting so the limiter’s drive level solely determines the gain reduction. Two cascaded gain settings set this drive level. One is MB Limit Drive; the second is a “hidden” setting whose gain is set by the active Target Loudness value.

You can activate our MX peak limiter technology, which introduces fewer audible artifacts than pure look-ahead limiting when a large amount of limiter gain reduction is required. However, it is a CPU-intensive process that adds about 240ms of delay, and is mainly beneficial if the target loudness is above -12 LUFS. (When the target loudness is below -12 LUFS, the peak limiter usually operates lightly.)

The MX limiter has two overshoot compensator modes: HARD and SOFT. HARD is very CPU-intensive and is mainly useful when you need to create very high loudness, such as that found in some “hyper-compressed” CDs. Additionally, SOFT CAN be cleaner sounding at the expense of producing some audible gain pumping. The choice of HARD or SOFT modes must be made by ear.

✶ HARD mode is only available when the pre-emphasis is FLAT.

In stereo mode and with Optimix disabled, only the left and right MX limiters are active, saving CPU cycles.

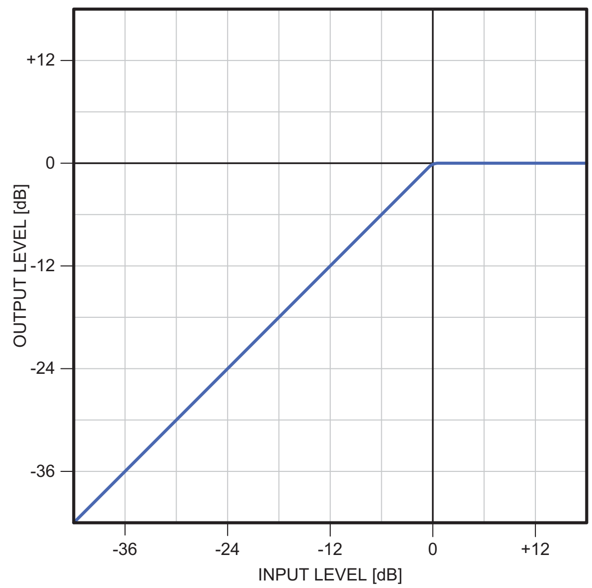


Figure 3.9 Peak Limiter Output vs. Input

Loudness Control

The CBS Loudness Controller and BS.1770 Safety Limiter cooperatively and automatically setting subjective loudness.

Useful in sound-for-picture applications, particularly when government regulations like the ***CALM Act*** require setting of subjective loudness, the CBS Loudness Controller follows the multi-band compressor in the signal flow. This placement reduces the drive level into the peak limiter when the loudness setting produces gain reduction. This minimizes peak-limiter-induced artifacts.

The loudness setting takes into account setting settings that affect the peak limiter so that the loudness setting approximately monitors the loudness at the limiter’s output, not the loudness at the multi-band compressor’s output. It does so by scaling its side-chain drive by two gain factors: (1) the MB Limit Drive setting and (2) the RSAP active Target Loudness value, which is determined either globally or by the active preset depending on the Target Loudness setting of the active preset. The Target Loudness value is the same as value of “dialnorm”. The RSAP contains a BS.1770 “integrated” meter with an integration time of 10 seconds. To comply with these standards, Target Loudness is customarily set to -24 in the United States and -23 in the EU. The Loudness Controller constrains the loudness of most commercials well enough to eliminate viewer annoyance. It works by constantly monitoring the subjective loudness of the RSAP output. The subjective loudness is a single value that represents the listener’s impression of the loudness in the listening room. It takes into account the contribution of all stereo and surround channels.

When subjective loudness would otherwise exceed the threshold set by the Loudness Threshold setting, the CBS Loudness Controller reduces the gain of material above 200Hz, preventing loudness from exceeding the threshold. To prevent the loudness setting from causing too much dynamic bass boost, you can use the Loudness Controller

Bass Couple setting to limit the maximum difference between the gain of the band below 200Hz and the band above 200Hz. For example, when this setting is set to 3 dB and the loudness setting's gain reduction is 10dB, the gain reduction below 200Hz will be 7dB. However, if the loudness setting's gain reduction is 2dB, the gain reduction below 200Hz will be 0 dB because the difference is now less than 3dB.

The loudness setting is triggered mainly by program material that has a lot of energy between 1 and 7kHz, which is the ear's most sensitive range. The five-band compressor can automatically re-equalize such program material so that it does not induce unnatural-sounding gain reduction in the loudness setting, and will decrease extremely sibilant program material before the loudness setting receives it.

The loudness setting's attack and release times are tuned to match the loudness integration times of the ear and are program-adaptive. Only the attack time is user-adjustable.

If you feel that the Loudness Controller is not setting the loudness of commercials or other subjectively loud program material sufficiently well, you may wish to set the threshold lower, forcing the Loudness Controller to do more work. You may also wish to activate the BS.1770 Safety Limiter.

In the Loudness Gain Reduction meter, the gain reduction of the BS.1770 Safety Limiter (in dB units) is stacked with gain reduction of the CBS loudness setting. The CBS setting gain reduction appears in blue, while the BS.1770 Safety Limiter gain reduction appears in cyan. The CBS Loudness Controller produces both fast and slow loudness setting; the fast setting rides on top of the slow setting. You can easily see this dual-speed operation on the Loudness GR meter. The Loudness Attack setting determines how much fast setting the Loudness Controller produces. As the setting is turned down toward 0%, it allows longer and longer loudness peaks to pass through.

Because of the system topology, the Loudness Level (CBS) meter assumes that the Loudness Attack setting is always set to 100% and does not indicate the effect of lower settings. As long as the setting is set to 50% or higher, this limitation should not have any significant effect on the Loudness Level meter's accuracy, and regardless of setting does not affect the BS.1770 loudness meter's accuracy. However, it is wise to double-check the effect of the Loudness Attack setting on subjective loudness by listening tests and/or use of an external Loudness Level meter.

Moreover, because of their placement in the signal path the CBS loudness setting and meter are unaware of the amount of gain reduction that occurs in the peak limiter. In normal sound-for-picture applications (with target loudness around -23 LUFS), the limiter operates very lightly (if at all) and does not affect the loudness, so loudness setting and metering are very accurate. If the target loudness is high (for example, -8 LUFS, which is a typical Target Loudness value in the radio-style presets), the peak limiter may be active enough to reduce loudness by 1 or 2 LU, and this will cause the Loudness Level meter to over-indicate the loudness by this much. (Again, you can use an external Loudness Level meter to check the actual loudness at RSAP output, and this issue does not affect the built-in BS.1770 meter.)

✶ All target loudness values recommended in ***EBU R-128*** are low enough to cause the loudness meter and setting to be very accurate.

The Loudness Controller may reduce the dramatic effect of certain sounds in entertainment programming, like gunshots, explosions, or screeching tires. Operators may therefore want to turn the Loudness Controller on during commercial breaks and off during normal programming. All sound-for-picture presets have the Loudness Controller on. The easiest way to handle this situation is to start with your preferred preset, turn the Loudness Controller off, and then save the result as a User Preset. Using one of the RSAP remote setting mechanisms, recall the "with loudness setting" and "without loudness setting" presets as desired.

Turning down the Loudness Attack setting provides another way to maintain the dramatic impact of loudness transients in dramatic programming; it can let gunshots and the like through while still constraining long-term loudness to a fixed threshold. While this is an easy solution that does not require your automation system to tell the RSAP when to recall presets, it is not ideal because there are some short-term loudness events, like sibilance, applause, and whistles, that can be annoying to audiences. You can use the Speech-Mode B4 and B5 compressor threshold settings to accomplish the same goal.

Another loudness setting strategy is this: Instead of using two presets with and without loudness setting (as described above), you can create presets with different settings of the Loudness Attack setting (and possibly different settings of the Loudness Threshold setting as well). Try a slow attack (50% or below) for dramatic programming and a faster attack (70%) for commercial breaks. This will maintain some automatic loudness setting for dramatic programming while setting the loudness of commercial breaks more rigorously.

✶ The Loudness Controller operates with reference to an absolute subjective Loudness Threshold that does not adapt to program context. This means that if there is a transition between very quiet program material and a commercial, the commercial may still seem offensively loud even though the Loudness Controller is setting its loudness correctly with reference to other sounds that reach full-scale loudness.

BS.1770 Safety Limiter

Following the CBS Loudness Controller is a BS.1770 Safety Limiter that will prevent a BS.1770-2 (or higher) loudness meter with 10second integration time from indicating higher than the setting of the BS.1770 Threshold setting, which is found in the Multiband tab. The BS.1770 Threshold setting is part of the on-air processing preset, so if you change the setting of this setting from its default value (which we recommend doing), you should save your work as a User Preset.

Because some organizations will disqualify an automatic loudness setting if it causes a BS.1770 meter to read higher than a specified threshold, all of the “TV” factory presets have this setting active with the BS.1770 Threshold setting set to 0 LU. If your organization does not have a strict policy about processing for the BS.1770 meter, we recommend that you edit your preferred preset by setting this setting anywhere from +2 to OFF and then saving the result as a User Preset.

The BS.1770 Safety Limiter receives the output of the CBS Loudness Controller and drives the final peak limiters. The total amount of loudness setting-induced gain reduction is the sum of the gain reduction produced by the CBS Loudness Controller and the gain reduction produced by the BS.1770 Safety Limiter. The gain reduction produced by the BS.1770 Safety Limiter changes slowly, seldom exceeds 2 dB, and is indicated by the cyan section of the Loudness GR meter. The gain reduction produced by the CBS Loudness Controller may change slowly or quickly (depending on the nature of the program material), appears in blue, and rides on top of the BS.1770 gain reduction. The peak reading of the meter thus shows the total gain reduction that both settings produce.

We included this Safety Limiter for customers whose policies require the BS.1770 loudness meter reading to be constrained below specified threshold regardless of how loud human listeners perceive the program to be. Our experience suggests that the BS.1770-2 meter will often overread material with unusually low peak-to-average ratios, like highly produced commercials and promos. Strict reliance on the BS.1770 meter can therefore make such material sound unnaturally quiet compared to surrounding material, so we prefer the sound when the CBS Loudness Controller is used exclusively for loudness setting.

Hardware Overview

This chapter presents information on the Radio and Streaming Audio Processor (RSAP) hardware.

Front Panel Overview

This section provides a general overview of the features of the RSAP front panel.

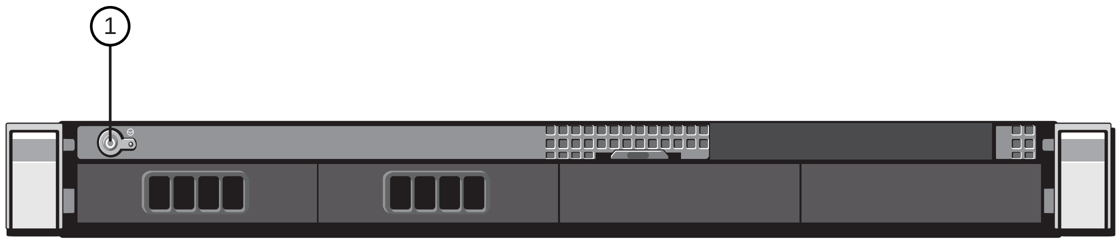


Figure 4.1 RSAP — Front Panel (Door Removed)

1. POWER Button

This is the main power button for the RSAP.

2. VGA Port

Use this port to connect a monitor if you do not have access to a network with DHCP for the initial Ethernet setup, or if you have a need to reset your RSAP static IP address outside of DashBoard.

3. Diagnostic Panel

This panel reports the serial number, network address mode, IP address, and net mask of the RSAP. Status information on the Dell® hardware is also displayed. Refer to the Dell documentation that accompanied the RSAP for details on the type of messages this panel displays.

4. USB Ports

Use each port to connect a keyboard and mouse if you do not have access to a network with DHCP for the initial Ethernet setup, or if you have a need to reset your RSAP static IP address outside of DashBoard.

Rear Panel Overview

This section provides an overview of the connections and cabling designations for the RSAP rear panel.

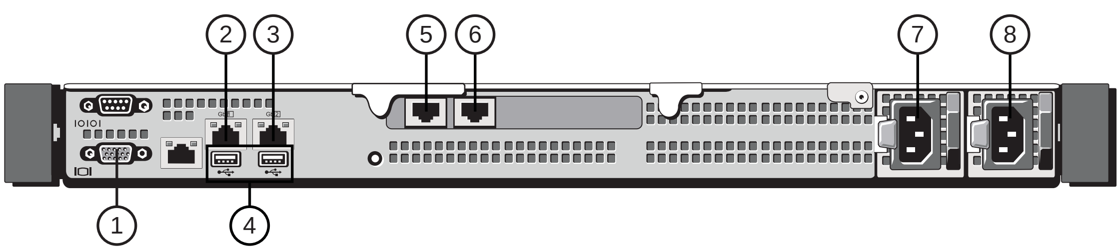


Figure 4.2 RSAP — Rear Panel

1. VGA Port

Use this port to connect a monitor if you do not have access to a network with DHCP for the initial Ethernet setup, or if you have a need to reset your RSAP static IP address outside of DashBoard. If the VGA Port on the front panel is also connected to a monitor, this connection supersedes it.

2. Gb1 Ethernet Port

This standard 10/100/1000 Base-TXRJ45 connector is used to connect the RSAP to your facility network. This connector is required to bridge the external Ethernet network to the local communication bus for monitoring and controlling the RSAP using DashBoard. This port is also used for software upgrades.

3. Gb2 Ethernet Port

This port is not implemented.

4. USB Port(s)

Use each port to connect a keyboard and mouse if you do not have access to a network with DHCP for the initial Ethernet setup, or if you have a need to reset your RSAP static IP address outside of DashBoard.

5. NET1 Port

The NET1 port is populated with a standard 10/100/1000 Base-TXRJ45 module from the factory. This port enables the RSAP to communicate with your PTP Grandmaster and act as a transmitter/receiver for network streams.

6. NET2 Port

The NET2 port is populated with a standard 10/100/1000 Base-TXRJ45 module from the factory. This port enables protection switching as per SMPTE ST 2022-7.

7, 8 Power Supplies

The RSAP comes standard with two power supplies.

Physical Installation

This chapter provides additional information for installing the Radio and Streaming Audio Processor (RSAP), and DashBoard before you can proceed to cabling and configuring your RSAP.

Installing the RSAP in a Rack Frame

Refer to the ***RSAP Quick Start Guide*** that accompanied your device and its mounting kit for installation information.

Connecting the Power Supplies

For redundancy, each power cord should be connected to a separate power source for protection against failure of the A/C power circuit. In the event of one power supply failure, the frame load is transferred to the other redundant power supply.

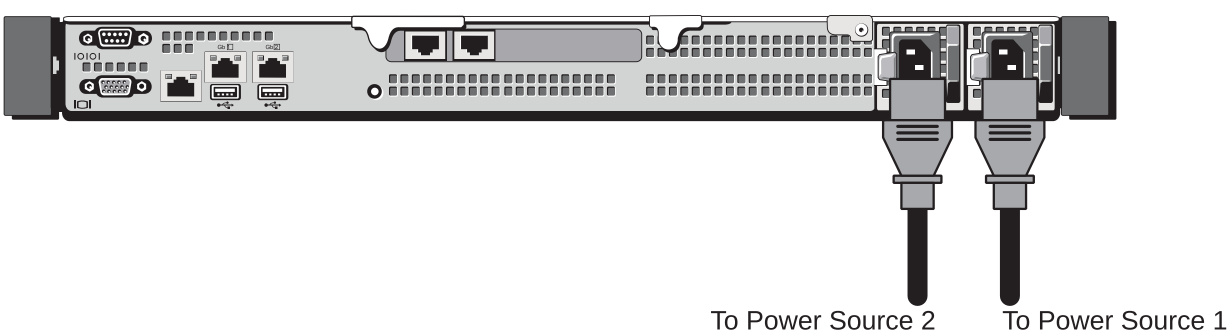
Warning — *In some countries, it may be necessary to supply the correct mains supply cord. Use only an approved IEC 320 C-13 type A/C line cord rated for a minimum 10A at 250V and certified for the country of use.*

The power supply plugs into the right-hand section of the RSAP rear panel. The universal power supply supports all worldwide AC power voltages, and no power adjustments are required.

To connect the RSAP to the power supplies

Warning Hazardous Voltages — *The safe operation of this product requires that a protective earth connection be provided. This protective earth is provided by the grounding conductor in the equipment's supply cord. To reduce the risk of electrical shock to operator and service personnel, this ground conductor must be connected to an earthed ground.*

1. Connect the cable's female IEC connector to the right power socket.
2. Plug the second IEC connector into left power socket.



3. Each AC connector includes a Power-Lock, which is designed to retain the power cable connector. Clip the Power-Lock over the shoulder of the inserted AC Cable end.
4. Connect each supplied power cable's three-prong male connector to an AC outlet.
5. Power on the RSAP by pressing the **POWER** button on the front panel. Refer to "**Front Panel Overview**" on page 31 for the location of this button.

✶ The fans run at full speed for a short period at the startup.

Cabling

If you have questions pertaining to the installation of RSAP, contact us at the numbers listed in the section "**Contacting Ross Video Technical Support**". Our technical staff is always available for consultation, training, or service.

Connecting to a Network

The RSAP is connected to your network via the **Gb1**, **NET1**, and **NET2** ports on the rear panel. Each port has a specific purpose:

- **Gb1** port — This port enables the RSAP to interface with other devices in your facility, and the computer running the DashBoard client.
- **NET1** port — This port enables the RSAP to interface with your facility PTP Grandmaster and enables the ability to receive and transmit network streams.
- **NET2** port — This port enables the RSAP to implement protection switching as per SMPTE ST 2022-7.

Before You Begin

Contact your IT department before connecting to your facility network to ensure that there are no conflicts. They will provide you with an appropriate value for the IP Address, Subnet Mask, and Gateway for your RSAP.

✶ If difficulties or problems are experienced when connecting the RSAP to a network hub, contact your network administrator.

For More Information on...

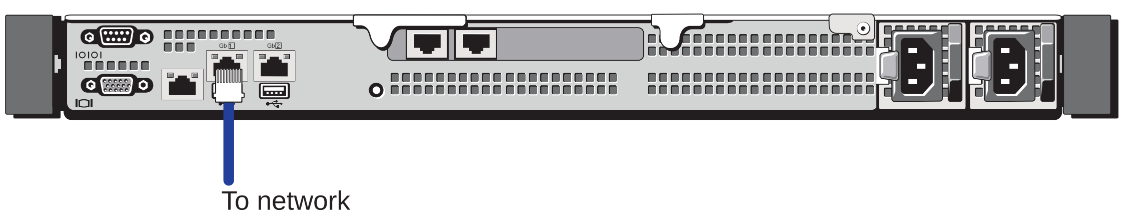
- downloading and installing DashBoard, refer to the ***DashBoard User Guide***.

Connecting to your Facility Network

✶ You will need to ensure that the RSAP is on the same network as your DashBoard client computer.

To connect the RSAP to your facility network

1. Connect one end of a standard RJ45 cable to the **Gb1** port on the RSAP rear panel.
2. Connect the other end of the same RJ45 cable to your Local Area Network (LAN).



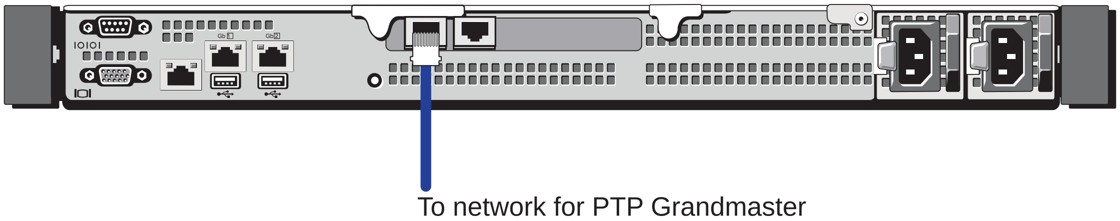
Connecting to the PTP Grandmaster

Connect the **NET1** port on the RSAP rear panel to establish communications with the PTP Grandmaster in your facility. This port is also used by the RSAP to transmit/receive network streams in your system.

To establish a connection for the PTP Grandmaster

1. Connect one end of a standard RJ45 cable to the **NET1** port on the RSAP rear panel.
2. Connect the other end of the same RJ45 cable to your Local Area Network (LAN).
3. Verify that the bottom right LED of the **NET1** port is lit green. This indicates that the connection is 1Gig.

✶ If this LED is not lit green, verify that the settings of your network switch allow 1Gig communication.



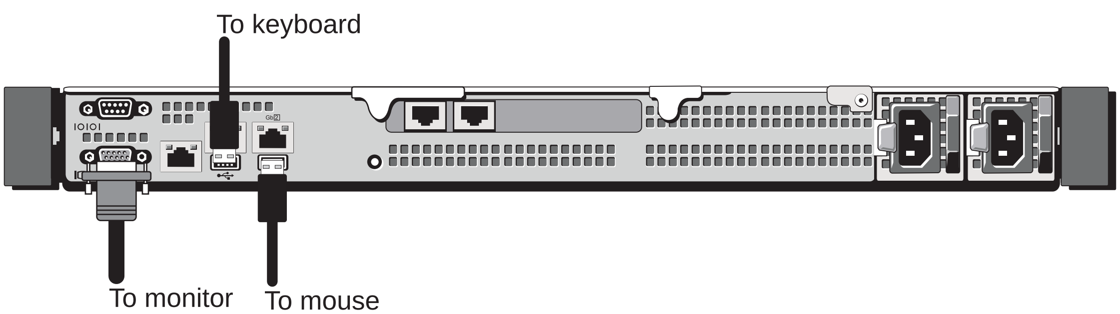
Connecting Peripheral Devices

During initial setup, you can optionally connect peripheral devices (such as a keyboard, mouse, and monitor) to the RSAP to use in establishing the initial IP settings for the RSAP. This is especially useful if your facility network does not support DHCP.

✶ KVM Extenders (customer supplied) are required when the RSAP monitor, keyboard/mouse are located remotely from the rack room. Ensure the cable length does not exceed the recommended standard length for your peripheral device.

To connect peripheral devices to the RSAP

1. Plug a keyboard into the top **USB** port on the RSAP rear panel.
2. Plug a mouse into the bottom **USB** port on the RSAP rear panel.
3. Plug a monitor into the **VGA** port on the RSAP rear panel.



Initial Connection

The Radio and Streaming Audio Processor (RSAP) is configured and operated via its interfaces in DashBoard. Once the RSAP is physically installed and connected to your facility network, you must establish its initial IP address and other network settings using the Command Line Interface (CLI) of the RSAP.

✶ If you have questions pertaining to the operation of RSAP, contact us at the numbers listed in the section **“Contacting Ross Video Technical Support**” on page 14. Our technical staff is always available for consultation, training, or service.

Physical Connections to the RSAP

Before configuring the initial network settings of the RSAP, you must:

- physically connect the RSAP to your network. Refer to “**Connecting to a Network**” on page 35.
- connect a keyboard and monitor to the RSAP chassis. Refer to “**Connecting Peripheral Devices**” on page 36.
- power on the RSAP. Refer to “**Connecting the Power Supplies**” on page 33.

Configuring the Initial Network Settings

There are two methods for configuring the initial network settings of the **Gb1** port on the RSAP: DHCP or Static. Both methods are described in this section.

✶ This section applies to the **Gb1** port only.

Before You Begin

Ensure that a computer running the latest DashBoard client software is installed (version 8.6.0 or higher) and available on the same subnet as the RSAP. The DashBoard client software and user guide are available from our website.

✶ Once the initial IP settings of the RSAP are established, you should verify that the RSAP is discovered by the DashBoard client. Refer to “**Launching DashBoard**” on page 39 for details.

Using DHCP

This section applies if your facility uses DHCP for assigning IP addresses.

To configure the RSAP settings using DHCP

1. Use the monitor and keyboard to access the Command Line interface of the RSAP.
2. At the **login** prompt, type **`ipcfg`** and press **Enter**.
3. At the **Password** prompt, type **`Setup`** and press **Enter**.
4. At the **Change IP configuration?** prompt, type **y** and press **Enter**.
5. At the **Use DHCP? y/N** prompt, type **y** and press **Enter**.

   The RSAP automatically logs out and the RSAP Application restarts.

Using Static Network Settings

If your facility does not use DHCP for assigning IP addresses, you will need to assign the static network settings as determined by your IT Department.

To specify the RSAP static settings

1. Use the monitor and keyboard to access the Command Line interface of the RSAP.
2. At the **login** prompt, type **`ipcfg`** and press **Enter**.
3. At the **Password** prompt, type **`Setup`** and press **Enter**.
4. At the **Change IP configuration?** prompt, type **y** and press **Enter**.
5. At the **Use DHCP? y/N** prompt, type **`N`** and press **Enter**.
6. Enter the IP address, Subnet Mask, Gateway, and DNS values, pressing **Enter** after each entry.

   The RSAP automatically logs out and the RSAP Application restarts.

Manually Adding the RSAP to the Tree View

The Tree View lists all DashBoard Connect devices that the DashBoard client can communicate with. Once you have added the RSAP to the Tree View, you can access its interfaces.

The RSAP does not automatically display the DashBoard Tree View. You must manually add it to the Tree View.

To manually add the RSAP to the Tree View in DashBoard

1. From the main tool-bar in DashBoard, select **File** > **New** > **TCP/IP DashBoard Connect or openGear Device**.

   The **New TCP openGear Frame Connection** dialog opens.
2. In the **IP Address** field, enter the IP Address you assigned to the RSAP in the procedure "**Configuring the Initial Network Settings**" on page 37.
3. Click **Detect Frame Information**.

✶ If you wish to assign a unique name for the RSAP, enter it in the **Display Name** field.

4. Click **Finish** to close the dialog.

   The RSAP node displays in the DashBoard Tree View.
5. Proceed to the section "**Accessing the RSAP Interfaces in DashBoard**" on page 39.

Using DashBoard

This chapter provides instructions for launching DashBoard, and accessing the Radio and Streaming Audio Processor (RSAP) interfaces in DashBoard.

Launching DashBoard

DashBoard must run on a computer that has a physical wired Ethernet connection. Wireless connections do not allow device discovery.

For More Information on...

- downloading and installing the DashBoard client software, refer to the ***DashBoard User Guide***.

To launch DashBoard

1. Ensure that you are running DashBoard software version 8.9.0 or higher.
2. Launch DashBoard by double-clicking its icon on your computer desktop.

Accessing the RSAP Interfaces in DashBoard

The RSAP interfaces are accessed by expanding the RSAP node in the Tree View and selecting a sub-node. Each sub-node represents a set of menus and options for the RSAP.

For More Information on...

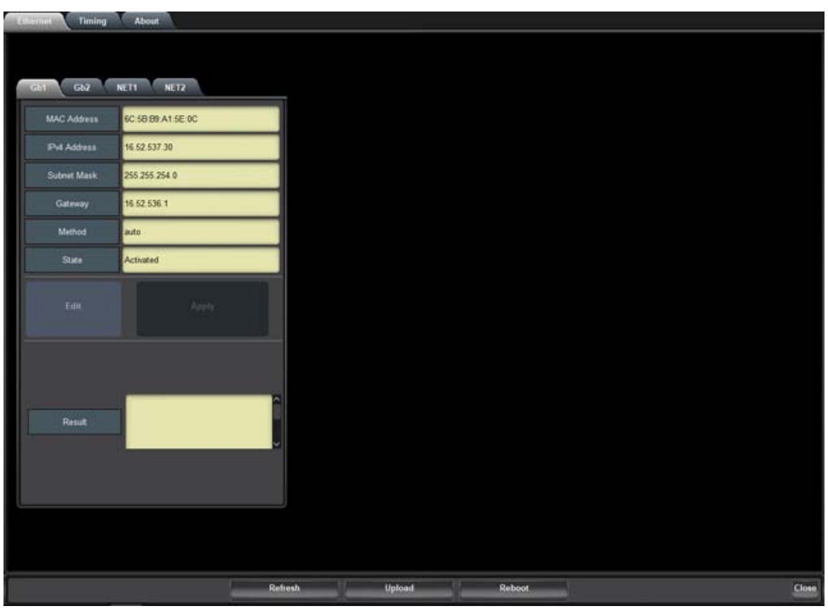
- the RSAP interfaces in DashBoard, refer to the chapter "**DashBoard Interface Overview**" on page 117.

To display the RSAP interfaces in DashBoard

1. Launch DashBoard.
2. In the Basic Tree View of DashBoard, double-click the **RSAP** node.

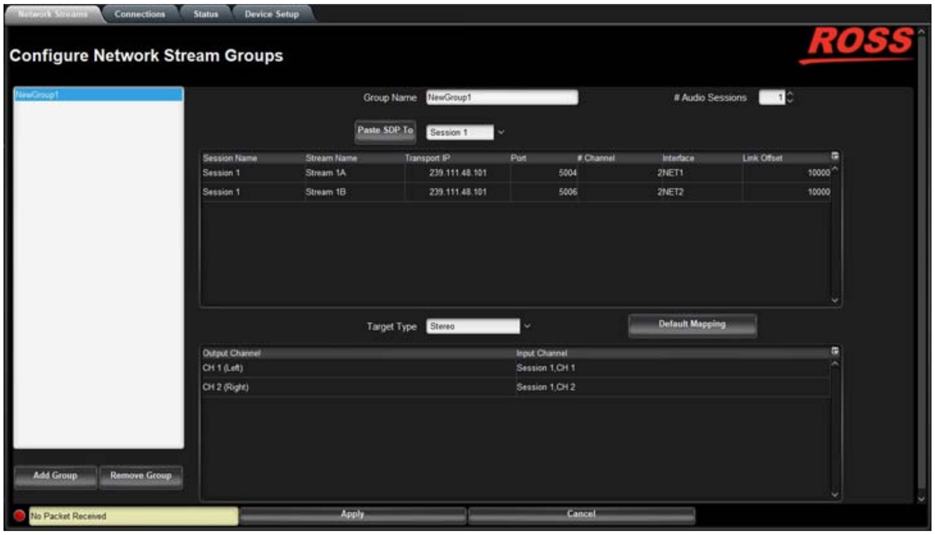
   A list of sub-nodes displays under the RSAP node.
3. Double-click the **Global** sub-node to display its tabs in the right-most pane of the DashBoard window.

   The Global interface enables you to configure the Ethernet settings for each port on the rear panel, and define the timing requirements for your RSAP.



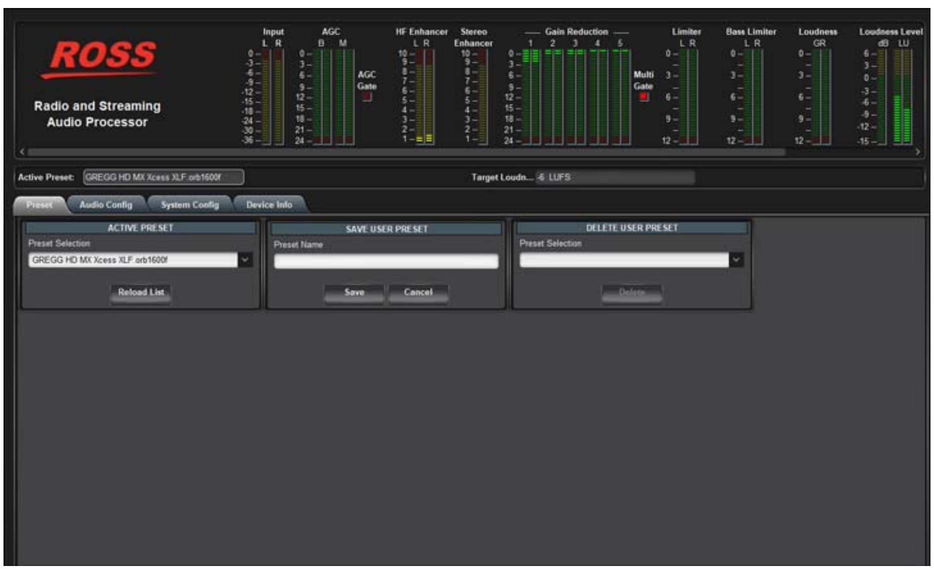
4. Double-click the **AES67 Receivers** sub-node to display its tabs in the DashBoard window.

   This interface enables you to create and manage the IP sessions in your system for the AES67 Receiver.



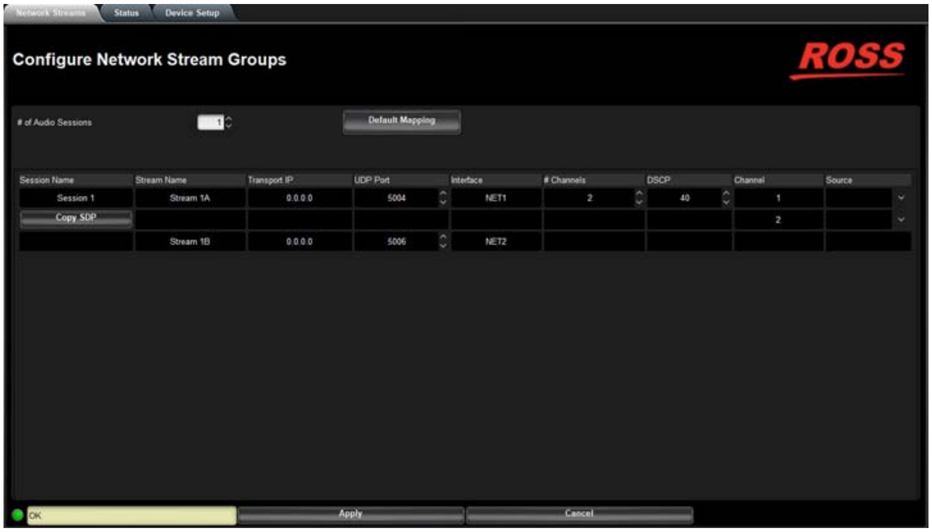
5. Double-click an **RSAP** sub-node to display its tabs in the DashBoard window.

   From this interface you can recall and edit audio processor presets, and monitor the audio output meters.



6. Double-click the **AES67 Senders** sub-node to display its tabs in the DashBoard window.

   This interface provides options for configuring the network streams for the AES67 senders.


Network Streams
Status
Device Setup
Configure Network Stream Groups
# of Audio Sessions
1
Default Mapping
Session Name
Stream Name
Transport IP
UDP Port
Interface
Channels
DSCP
Channel
Source
Session 1
Stream 1A
0.0.0.0
5004
NET1
2
40
1
Copy SDP
2
Stream 1B
0.0.0.0
5006
NET2
OK
Apply
Cancel

Updating the Network Settings

Once the Radio and Streaming Audio Processor (RSAP) is communicating via DashBoard, it is recommended to assign a different static IP address from the factory default values. These default values were used to initially establish a connection point to the RSAP, as outlined in your ***RSAP Quick Start Guide***. This chapter provides instructions for configuring the RJ45 ports of the RSAP for communication with your facility network and streaming devices.

✶ Contact your IT Department for more information on changing these settings.

Network Connections to the RSAP

There are two Ethernet ports that must be configured on the RSAP: **Gb1** and **NET1**. Each port has a specific purpose:

- **Gb1** port — This port enables the RSAP to interface with other devices in your facility, and the computer running the DashBoard client. After the initial connection is established (as outlined in "**Configuring the Initial Network Settings**" on page 37), DashBoard is used to update the network settings for the RSAP as determined by your IT Department and network requirements.
- **NET1** port — This port enables the RSAP to interface with your facility PTP Grandmaster and enables the ability to receive and transmit network streams.

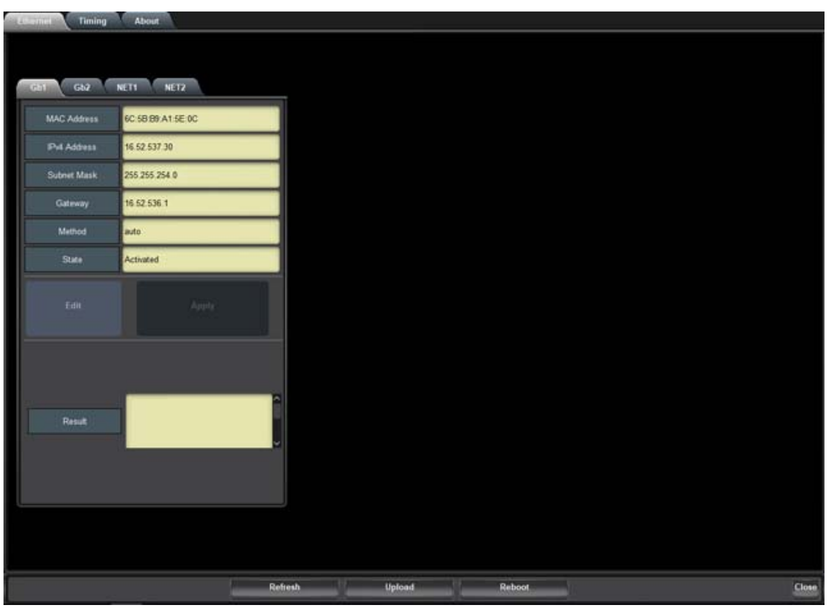
Updating the Gb1 Port Settings

Once the RSAP is communicating via DashBoard, you may wish to assign a different static IP Address from the factory default value (which was used to initially establish a connection point to the RSAP).

✶ The **Gb2** port is not implemented.

To update the Gb1 port settings for the RSAP

1. Display the **Global** interface as outlined in "**To display the RSAP interfaces in DashBoard**" on page 39.
2. Select the **Ethernet** tab.
3. Select the **Gb1** tab.



4. Click **Edit**.

   The fields are now editable on the tab.

5. Use the **Method** menu to specify **Manual**.
6. Use the **Address** field to specify the new static IP Address for the **Gb1** port.
7. Use the **Subnet Mask** field to specify the subnet mask for the NET port.
8. Use the **Gateway** field to specify the gateway for communications outside of the local area network (LAN) the RSAP will use.
9. Click **Apply**.

Updating the NET1 Port Settings

By assigning an IP Address to the **NET1** port, you are able to uniquely identify it on the network and control it via the DashBoard interface. Once these network settings are successfully assigned, you can proceed to configure the **NET1** port for media traffic for the RSAP.

To update the NET1 port settings for the RSAP

1. Display the **Global** interface as outlined in "**To display the RSAP interfaces in DashBoard**" on page 39.
2. Select the **Ethernet** tab.
3. Select the **NET1** sub-tab.
4. Click **Edit**.

   The fields are now editable on the tab.

5. Use the **Method** menu to specify **Manual**.
6. Use the **Address** field to enter the new IP Address for the **NET1** port as provided by your IT Department. This is the IP Address that is used to control and communicate with the NET1 port.
7. Use the **Subnet Mask** field to specify the subnet mask for the NET port.
8. Use the **Gateway** field to specify the gateway for communications outside of the local area network (LAN) the RSAP will use.
9. Click **Apply**.

Configuring the Timing Settings

The Radio and Streaming Audio Processor (RSAP) supports the Precision Time Protocol (PTP) as defined in the IEEE 1588-2008 standard and the SMPTE ST 2059 specification.

Configuring the PTP Settings

From the Timing tab in DashBoard, you can synchronize the RSAP to real-time clocks of other devices in the same network.

✶ There are several criteria that PTP clocks compare to determine who will be master and who will be slave (called the Best Master Clock Algorithm, or BMCA), and they are evaluated in order: Priority1, clock class, accuracy, scaled log variance, Priority2, clock ID (similar to the MAC address). Practically, Priority1 is the only setting configured on all clocks to control the outcome of the Grandmaster election. If Priority1s are equal, the next criterion is evaluated (clock class) and the criteria are evaluated in succession until a Grandmaster is determined.

To update the PTP settings for the RSAP

1. Display the RSAP interfaces in DashBoard as outlined in "**To display the RSAP interfaces in DashBoard**" on page 39.
2. Select the **Timing** tab.

✶ The **Slave Only** box is selected and cannot be edited.

3. Use the **Profile** menu to specify the standard/specification used for PTP.
4. Use the **Domain** field to specify the sub-domain the PTP clock is assigned to.

✶ There can be multiple PTP domains operating concurrently within a network. The domain is a field in all PTP message headers. Messaging between entities are segregated by domain (e.g. The RSAP is an endpoint configured for domain 128 and ignores messages from a neighboring clock configured for domain 127).

5. Use the **Sync Interval** field to specify the number of seconds at which synchronization messages are sent from the master clock to the specified NET port on the RSAP.
6. Use the **Announce Interval** field to specify the rate of announce messages that the specified NET port on the RSAP requests from the master clock during a Unicast session.
7. Use the **Announce Receipt Timeout** field to specify the number of seconds the specified NET port on the RSAP waits for an announce interval message before timing out.
8. Click **Apply Changes** to save the new settings.

Configuring the AES67 Receiver

This chapter outlines the steps required to configuring and allocate the AES67 receiver streams to audio channels on the Radio and Streaming Audio Processor (RSAP).

Overview

From DashBoard you defining audio streams that provide AES67 data (including the IP Address of each stream and the number of audio channels it includes) then map the streams to create Network Stream groups.

The following steps are required:

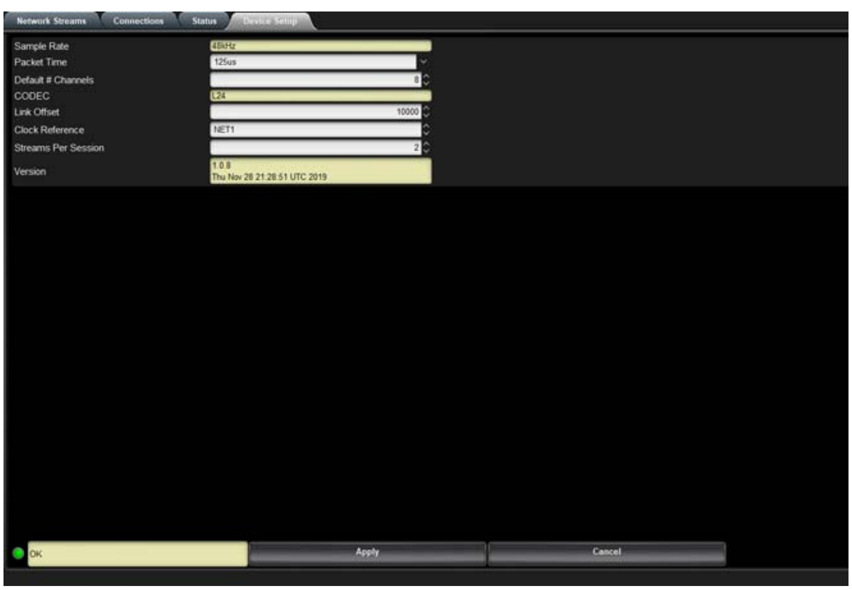
1. Configure the AES67 Receiver global settings.
2. Define the Network Stream group(s) that will provide the AES67 data to be encoded.

Configure the Global AES67 Receiver Settings

There are two settings that impact all AES67 Receiver streams on the RSAP: the default number of (audio) channels, and specify a Link Offset. These settings are located on the Device Setup tab in the AES67 Receiver interface.

To define the default number of audio channels for the AES67 Receiver

1. Display the **AES67 Receiver** interface in DashBoard as outlined in “**To display the RSAP interfaces in DashBoard**” on page 39.
2. Select the **Device Setup** tab.



3. Use the **Default # Channels** field to specify the number of audio channels available to map for any network stream group for the AES67 Receiver.

✶ This number can be updated for each group during configuration.

4. Use the **Link Offset** field to specify the number of nanoseconds (offset) the source will be delayed from the sender’s RTP timestamps.

✶ This allows for non-PTP aligned source to be passed through with fixed latency.

5. Click **Apply** to save your changes.

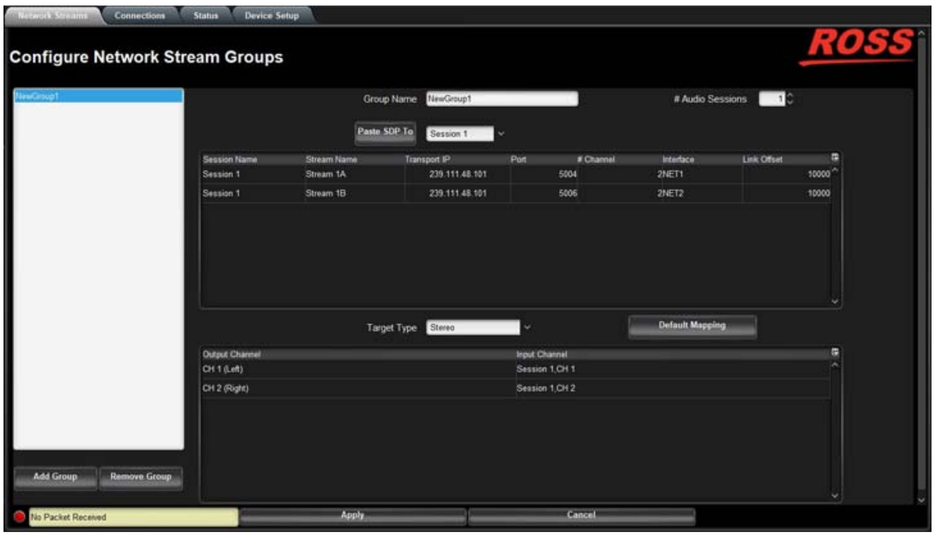
Creating Network Stream Groups for the AES67 Receiver

A network stream group includes a specified number of audio streams, with each stream from a defined IP Address. Further define each group by specifying the type of audio (Stereo or 5.1 Surround) and mapping the input channels. You can create as many network stream groups as required.

Once you have created your network stream groups, you then can allocate them to output channels (as described in "**Routing a Network Stream Group to an Output Channel**" on page 51).

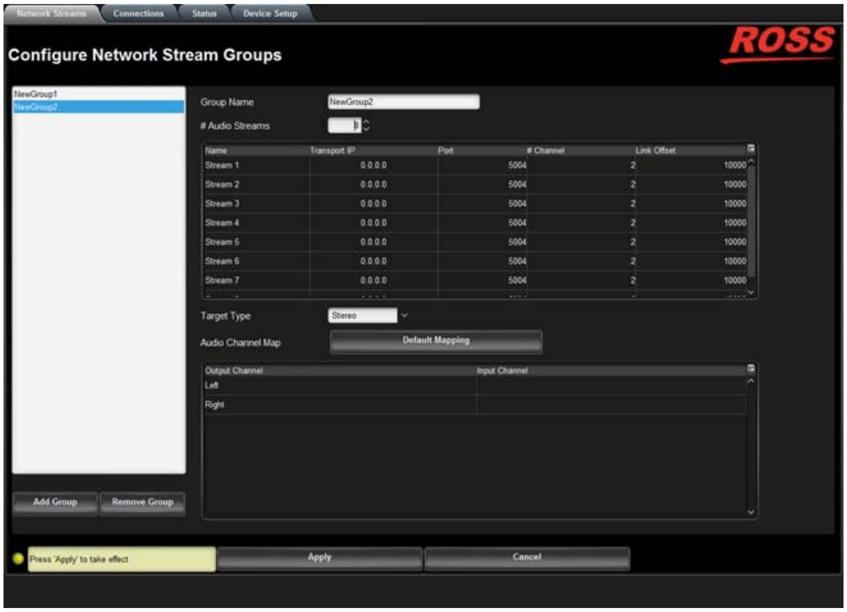
To add a new network stream group

1. Display the **AES67 Receiver** interface in DashBoard as outlined in "**To display the RSAP interfaces in DashBoard**" on page 39.
2. Select the **Network Streams** tab.



3. Click **Add Group**.
   - The fields in the **Network Streams** tab clear.
   - The **Group Name** field displays "**`NewGroup#`**" where # is an auto-generated character.
4. Use the **Group Name** field to specify a unique identifier for the network stream group.
5. Use the **# Audio Streams** field to specify the number of audio streams this network stream group will contain.

   The Group list (on the left of the tab) updates to display a row for each defined network stream group. In the following example, the user specified 8 Audio Streams for the new network stream group.

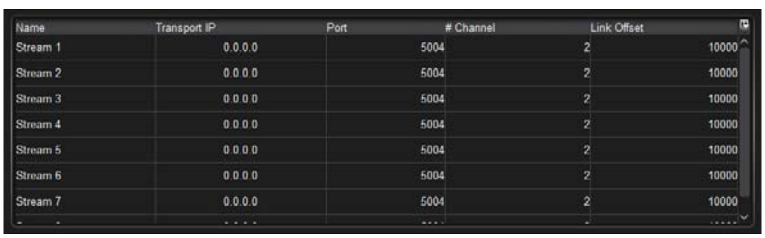


6. Use the **Group Name** field to assign a unique name to the audio stream in the network session.

To define the streams within a group

1. Locate the stream group table on the **Network Streams** tab.

   Each row in the table represents an audio stream within that network group. In the following example, the user created eight audio streams within a single network stream group.



Name	Transport IP	Port	# Channel	Link Offset
Stream 1	0.0.0.0	5004	2	10000
Stream 2	0.0.0.0	5004	2	10000
Stream 3	0.0.0.0	5004	2	10000
Stream 4	0.0.0.0	5004	2	10000
Stream 5	0.0.0.0	5004	2	10000
Stream 6	0.0.0.0	5004	2	10000
Stream 7	0.0.0.0	5004	2	10000

2. Locate the row for the stream you wish to configure.
3. Use the **Name** field to assign a unique name to the audio stream.
4. Use the **Transport IP** field to specify the Multicast IP Address for the audio stream.

✶ Only multicast IP Address of 239.x.x.x and 232.x.x.x can be received by the RSAP. Contact Ross Technical Support if you need additional IP ranges.

5. Use the **Port** field to specify the RTP port for the audio stream.
6. Use the **# Channel** field to specify the maximum number of audio channels in this stream.
7. Use the **Link Offset** field to adjust the relative position of the audio start position as an offset to the reference.
8. Repeat steps 2 to 7 for each audio stream you wish to define.
9. Click **Save** to update save your changes and update the list in the Network Streams tab.

Mapping the Audio Channels to the AES67 Receiver

When defining a network stream group, you can choose to apply the default audio channel map or assign the channels as required by your system.

To apply the default audio map to the network stream group

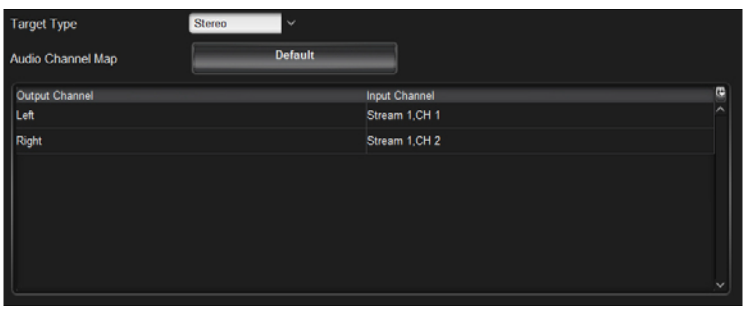
1. Display the **AES67 Receiver** interface in DashBoard as outlined in "**To display the RSAP interfaces in DashBoard**" on page 39.
2. Select the **Network Streams** tab.
3. Use the **Target Type** menu to specify the type of audio channels that will be assigned to the network stream group.
4. Click **Default**.

   The Audio Channel Map table updates to display the default settings for the audio type selected.
5. Click **Apply** to save your settings.

To customize the stereo audio channels assigned to a network stream

1. Display the **AES67 Receiver** interface in DashBoard as outlined in "**To display the RSAP interfaces in DashBoard**" on page 39.
2. Select the **Network Streams** tab.
3. Select **Stereo** from the **Target Type** menu.

   The Audio Channel Map table updates to display rows for the Left and Right in the Output Channel column.



4. Use the **Input Channel** menu to assign a specific stream/channel pair to the output channel of the network stream group.
5. Click **Apply** to save your settings.

To customize the Dolby® Surround audio channels assigned to a network stream

1. Display the **AES67 Receiver** interface in DashBoard as outlined in "**To display the RSAP interfaces in DashBoard**" on page 39.
2. Select the **Network Streams** tab.
3. Select **5.1 Surround** from the **Target Type** menu.

   The Audio Channel Map table updates to displays a row for each channel in the Output Channel column.

4. Use the **Input Channel** menu to assign a specific stream/channel pair to the output channel of the network stream group.
5. Click **Apply** to save your settings.

To customize the Dolby® Surround and Stereo audio channels assigned to a network stream

1. Display the **AES67 Receiver** interface in DashBoard as outlined in “**To display the RSAP interfaces in DashBoard**” on page 39.
2. Select the **Network Streams** tab.
3. Select **5.1 Surround and Stereo** from the **Target Type** menu.

   The Audio Channel Map table updates to displays a row for each channel in the Output Channel column. There are eight channels available; you may need to scroll to the bottom of the table to access the Stereo channels.

Output Channel	Input Channel
5.1 Left	Stream 1, CH 1
5.1 Right	Stream 1, CH 2
5.1 Center	Stream 2, CH 1
5.1 LFE	Stream 2, CH 2
5.1 Left Surround	Stream 1, CH 1
5.1 Right Surround	Stream 1, CH 2
Stereo Left	Stream 2, CH 1

4. Use the **Input Channel** menu to assign a specific stream/channel pair to the output channel of the network stream group.
5. Click **Apply** to save your settings.

Routing a Network Stream Group to an Output Channel

To route the audio streams, you must first select an Output channel then a network stream group. Keep in mind that routing occurs automatically after a Stream Group button is selected.

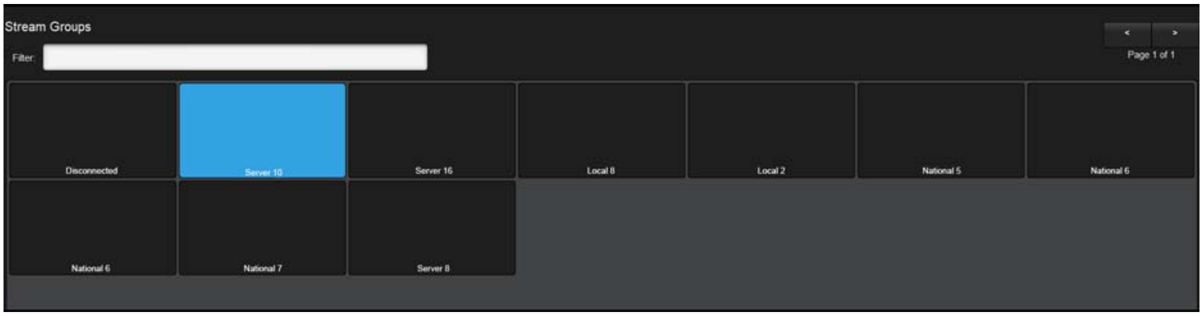
To select an output channel

1. Display the **AES67 Receiver** interface in DashBoard as outlined in “**To display the RSAP interfaces in DashBoard**” on page 39.
2. Select the **Connections** tab.
3. In the **Output Name** row of the **Outputs** area, locate the button for the output channel you wish to allocate the Network Stream Group to.
4. Click the required **Output** button.

To allocate a Network Stream Group to an output channel

1. In the **Stream Groups** area, locate the button for the Network Stream group you wish to route to the selected output channel.

✶ Each button represents a Network Stream Group configured in “**Creating Network Stream Groups for the AES67 Receiver**” on page 48.



2. Click the required **Stream Group** button to perform the switch.

   The selected Network Stream Group is now allocated as the source for the output channel.

Configuring the AES67 Senders

This chapter outlines the steps required for configuring the AES67 sender streams on the Radio and Streaming Audio Processor (RSAP). Each sender stream can be configured independently.

Overview

From DashBoard you specify which audio streams will transmit the encoded audio data to specific RSAP output channels. The following steps are required:

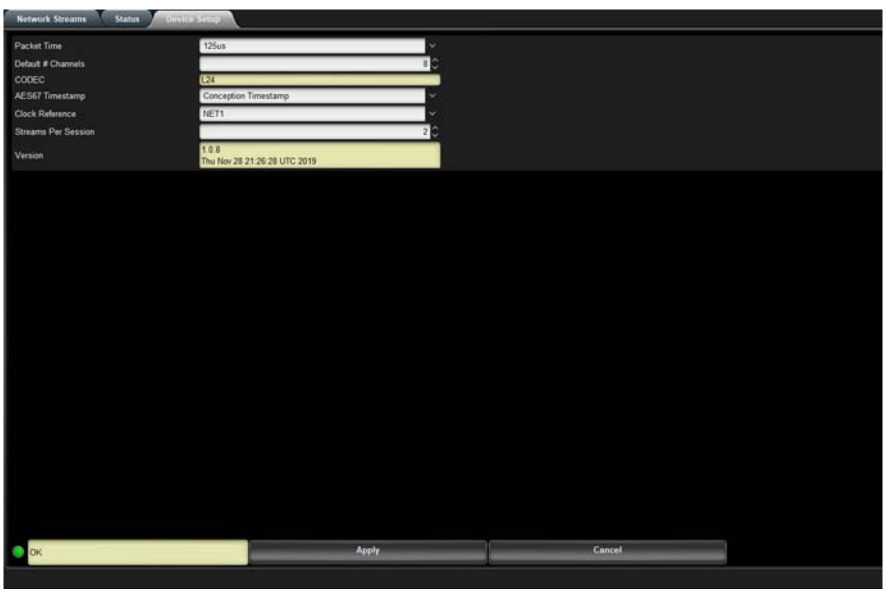
1. Configure the AES67 Sender global settings.
2. Configure the Network Stream groups, including the number of channels in each stream.
3. Map the audio signals to output channels.

Configuring the Global AES67 Sender Settings

There are two settings that impact all AES67 sender streams on the RSAP: the default number of (audio) channels in each stream, and how to process the AES67 timestamps. These settings are located on the Device Setup tab in the AES67 Sender interface.

To configure the global settings for all AES67 sender streams

1. Display the **AES67 Sender** interface in DashBoard as outlined in "**To display the RSAP interfaces in DashBoard**" on page 39.
2. Select the **Device Setup** tab.



3. Use the **Default # of Channels** field to specify the maximum number of audio channels available to map for any network stream group for the AES67 Sender.
4. Specify how to process the AES67 timestamps by selecting an option in the **AES67 Timestamp** area. Choose from the following:
   - **Conception Timestamp** — The AES67 data will include the timestamp that was provided when the RSAP received it.
   - **Processed Timestamp** — The RSAP will include an updated timestamp based on the PTP Grandmaster clock.

Mapping the Audio Channels for the AES67 Sender

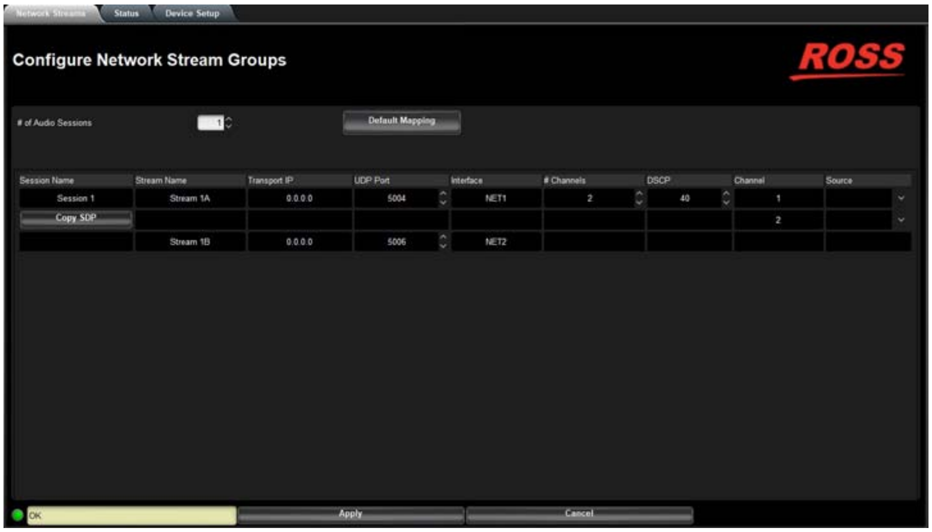
You need to specify the IP encapsulation properties for the active audio for each input signal.

To map the audio channels for the AES67 Sender

1. Display the **AES67 Sender** interface in DashBoard as outlined in “**To display the RSAP interfaces in DashBoard**” on page 39.

   The **Network Streams** tab is automatically selected.

2. Use the **# of Audio Streams** field to specify the total number of streams available to the **AES67 Sender**.



✶ Click **Default Mapping** to use the default values.

3. Use the **Name** field to assign a unique identifier to the stream.

✶ The **Name** is used to help identify the stream within your system and other tabs in the DashBoard window.

4. Use the **Transport IP** field to specify the network socket for the data for the session.
5. Use the **Port** field to specify the source port to connect to the stream. This must match the source you are attempting to connect to.
6. Use the **# Channels** field to specify the number of audio channels in the specified stream.

   The tab updates to display a row for each channel. In the example below, the # Channels field was set to 2.

7. Use the **DSCP** field to specify the Differentiated Service Code Point value for the network layer service for this session.
8. Use the **Channel** field to specify the audio channel within the network stream group you are defining.

Audio Processing Concepts

Before you begin, remember that successful audio processing requires high-quality source material and accurate monitoring. A major potential cause of distortion is excess peak limiting. Another cause is poor quality source material, including the effects of your playback machines, electronics, and studio-to-transmitter link (if any). If the source material is distorted even slightly, that distortion can be exaggerated by RSAP—particularly if a large amount of gain reduction is used. Very clean audio can be processed harder without producing objectionable distortion.

✶ A high-quality monitor system is essential. To modify your sound effectively, you must be able to hear the results of your adjustments.

Overview

Reducing the peak-to-average ratio of the audio increases loudness. If peaks are reduced, the average level can be increased within the permitted modulation limits. The effectiveness with which this can be accomplished without introducing objectionable side effects (such as pumping or inter-modulation distortion) is the single best measure of audio processing effectiveness.

Compression

Compression reduces dynamic range relatively slowly in a manner similar to riding the gain. Limiting and clipping, on the other hand, reduce the short-term peak-to-average ratio of the audio. This reduces the difference in level between the quiet and loud sounds to make more efficient use of permitted peak level limits, resulting in a subjective increase in the loudness of quiet sounds. It cannot make loud sounds seem louder.

Limiting

Limiting increases audio density, making loud sounds seem louder, but can also result in an unattractive busier, flatter, or denser sound. It is important to be aware of the many negative subjective side effects of excessive density when setting settings that affect the density of the processed sound.

Clipping

Sharp peaks does not produce any audible side effects when done moderately. Excessive clipping will be perceived as audible distortion.

Look-ahead Limiting

Prevents overshoots by examining a few milliseconds of the unprocessed sound before it is limited. This way the limiter can anticipate peaks that are coming up.

✶ In Dolby® Digital transmission channels, appropriate setting of the Target Loudness metadata parameter will allow enough headroom to keep peak levels below the threshold of the RSAP’s peak limiters. The best sounding limiting is no limiting at all.

Loudness

This setting prevents the subjective loudness from exceeding a preset threshold.

Distortion in Processing

In a competently designed processor, distortion occurs only when the processor is controlling peaks to prevent the audio from exceeding the peak modulation limits of the transmission channel. The less peak setting that occurs, the less likely that the listener will hear distortion. However, to reduce the amount of peak setting, you must decrease the Drive level to the peak limiter, which causes the average level (and thus, the loudness) to decrease proportionally.

Loudness and Distortion

✶ In AM and FM processing, there can be a direct trade-off between loudness, brightness, and distortion, However, because DAB and netcasting systems do not use pre-emphasis, there is little challenge getting the audio to sound bright and the trade-off is only between loudness and distortion.

Perhaps the most difficult part of adjusting a processor is determining the best trade-off for a given situation. It is recommended to give up ultimate loudness to achieve low distortion. A listener can compensate for loudness by simply adjusting the volume setting. However, a listener cannot make an excessively compressed or peak-limited signal sound clean again.

If processing for high quality is done carefully, the sound will also be excellent on small playback systems. Although such a signal might fall slightly short of ultimate loudness, it will tend to compensate with an openness, depth, and punch (even on small speakers) that cannot be obtained when the signal is excessively squashed.

Speech/Music Detector

The Speech/Music Detector allows the RSAP to change its processing parameters depending on whether the input program material is speech or other material (usually music).

The algorithm is straightforward: Speech is detected if (1) the input is mono, and (2) there are syllabic pauses at least once every 1.5 seconds. Speech with a stereo music background will usually be detected as “music,” or the detector may switch back and forth randomly if the stereo content is very close to the stereo / mono detector’s threshold. Mono music with a “speech-like” envelope may be incorrectly detected as “speech.” Music incorrectly detected as “speech” can exhibit a slight loss of loudness and punch, but mis-detection will never cause objectionable distortion on music.

Speech that is not located in the center of the stereo sound field will always be detected as “music” because the detector always identifies stereo material as “music.”

Processing for Audio Codecs

Professional netcasters committed to providing their audiences with the best audio possible at a given bit-rate should consider the use of a high-performance codec like Modulation Index’s StreamS, which uses standards-based MPEG-4 AAC/HE-AAC/He-AACv2/xHE-AAC. The xHE-AAC is the most efficient codec available at the time of this writing. It can provide good entertainment-quality audio at bit rates as low as 24kbps stereo or 16kbps mono, allowing coverage of audiences listening on dial-up connections or via wireless devices. The lower bit-rates also penetrate crowded networks with fewer audio interruptions.

The RSAP provides the ability to maintain source-to-source spectral consistency. Once you have set up the processing to minimize codec artifacts caused by a given piece of program material, the RSAP will automatically minimize codec artifacts with any program material.

All codecs add peak overshoot to the audio. This because they remove energy that is psycho-acoustically masked by the input audio. It is not unusual for low bit rate codecs to overshoot by 2dB. To minimize the possibility of clipping at the decoder, it is wise to allow 2dB of peak headroom at the encoder. In other words, set the output level of the RSAP to -2dBFS when driving an audio encoder. While there may be a few low-energy overshoots above this, clipping them will not cause audible artifacts.

Depending on the type of program material you are processing, at lowest bit-rates (32kbps and lower with HE-AAC) it is may be advisable to use presets with the MX Limiter disabled because the extreme peak density produced by the MX Limiter may unduly stress the codec. However, disabling the MX Limiter will also reduce punch and transient definition, so you must decide by careful listening to the output of the decoded signal after the codec’s encode/decode cycle.

Some decoders contain a peak limiter that prevents clipping if enabled. However, the subjective quality of any such peak limiter is unpredictable to the netcaster, so it is better to avoid enabling them at all.

Using the RSAP in Radio-Oriented Applications

RSAP can be adjusted so that the output sounds:

- as close as possible to the input at all times (using the Two-Band Protection Limiter preset)
- open but more uniform in Frequency balance (and often more dramatic) than the input (using the Two-Band structure or running the five-band structure with slow release time)
- dense, quite squashed, and very loud (using the five-band structure with faster release times

The dense, loud setup will make the audio seem to jump out of car and table radios, but may be fatiguing and invite tune-outs on higher quality home receivers. The loudness/distortion trade-off explained above applies to any of these setups.

In professional broadcasting environments, you will achieve best results if Engineering, Programming, and Management go out of their way to communicate and cooperate with each other. It is important that Engineering understand the sound that Programming desires, and that Management fully understands the trade-offs involved in optimizing one parameter (such as loudness) at the expense of others (such as distortion or excessive density).

Keep in mind that the listener can easily setting loudness but cannot make a distorted signal clean again. If such excessive processing is permitted to audibly degrade the sound of the original program material, the signal is irrevocably contaminated and the original quality can never be recovered.

Sound for Picture Applications: Controlling Dynamic Range

The most crucial commandment in sound for picture is this: dialog must always be intelligible. Sound for picture is usually heard under less-than-ideal conditions and its dynamic range must be settingled accordingly. Apartment-dwellers must set their volume settings to avoid disturbing neighbors or even other members of the family. At the quiet side, intelligibility of dialog is often impacted by environmental noise such as children playing or a dishwasher going in the kitchen. When one considers that the hearing acuity of a significant portion of the audience is somewhat impaired compared to that of a healthy 20-year-old, one concludes that the dynamic range of dialog must not exceed 15dB if it is to be intelligible to 99% of viewers under common domestic viewing conditions. Feature-film dynamic range is inappropriate for home viewing (except in dedicated home theaters) and the dynamic range of a significant portion of the audio from video source material must be compressed to best serve the audience. The challenge (which RSAP effectively meets) is to compress dynamic range unobtrusively.

RSAP can be adjusted so that the output sounds as close as possible to the input at all times (using the Peak Limiter preset), or so that it sounds open but more uniform in Frequency balance than the input (using considerable interband coupling and a slow release time in the five-band compressor), or so that it sounds dense, quite squashed, and very loud (using faster release times and less band coupling).

In television audio, inconsistent loudness between channels or program elements is annoying, so the dense, loud setup is never appropriate. The RSAP offers two-band and five-band presets (whose names begin with “TV”) that exploit the AGC’s and multi-band compressor’s compression ratio settings to subtly setting dynamic range in sound for picture applications. These presets effectively and unobtrusively maintain dialog intelligibility while retaining a sense of dynamic range, allowing low-level elements to be heard easily. Meanwhile, the CBS Loudness Controller prevents subjective loudness from exceeding a preset ceiling.

The preset tuning settings on the RSAP give you the flexibility to adapt the processing to individual program segments. In most cases, your goal should be to choose the type of processing that best optimizes dynamic range while controlling the loudness of the loudest sounds so that they are not irritating and are consistent with the loudness of other stations or sources. When the RSAP is otherwise set up correctly (so that it is cognizant of the dialnorm metadata you are transmitting to viewers), its TV# presets achieve this goal most precisely by exploiting the loudness setting. The “radio-style” presets are crafted to match the target loudness as well as can be achieved without use of the loudness setting.

If you want more consistent loudness from a “radio-style” preset:

1. Set the Loudness Controller Threshold to -10dB.
2. Increase the MB Limiter Drive setting until the loudness setting gain reduction meter indicates about 3dB of gain reduction with typical program material.

✶ This will often reduce punch with musical programming.

3. Save the result as a user preset.

Protection Limiting

The RSAP has an advanced peak limiter that is more than competitive with the best mastering peak limiters, particularly in MX mode. The Peak Limiter MX preset allows you to use the RSAP as a mastering-style limiter that is capable of a surprisingly favorable trade-off between loudness and undesired artifacts. It is typically possible to make -6 or -7 LUFS masters that sound clean, punchy, and open.

Studio AGC

You can use the RSAP as a studio AGC and STL protection peak limiter. In a typical application, RSAP substitutes for the AGC at the transmitter and provides protection limiting for the STL, which can be flat or pre-emphasized at 50µs or 75µs.

Moreover, because the RSAP supports presets that be recalled by remote setting, it can be automatically synchronized to the presets on-air at a transmitter-side when presets are day-parted. To achieve this match:

1. Recall one of the STUDIO AGC presets. Refer to "**Recalling a Preset**" on page 59 for details.

   There are three such presets, for flat, 50µs, and 75µs pre-emphasized STLs.

2. Configure the Peak Limiter to match the pre-emphasis that your STL uses. Refer to "**Limiters Controls**" on page 109.

3. Edit AGC parameters of the RSAP preset so they are the same as the AGC parameters on-air at the transmitter-side.

4. Save your work as an user preset.

5. Adjust the transmitter-side input reference level so that the RSAP performs the correct amount of multi-band gain reduction (i.e., the same amount of GR that it would have performed if its internal AGC were active).

Using Audio Presets

This chapter outlines how to recall, edit, and save the available presets for audio processing.

Overview

There are two types of presets for the RSAP: Processing and Setup. Both types can be customized and saved under different names. Each instance of the processing is identified by a Processor number or user-generated alias and each Processor “owns” its individual customized presets.

Processing Presets Overview

There are over 50 Factory Processing Presets. They are designed to be compatible with almost any program format.

Each Factory Processing Preset on the Open Preset list is a library of 19 separate presets, selected by using the Less-More setting to adjust the RSAP for less or more processing.

Each set of Factory Preset files consists on one “master” file and several “less/more” files. Master files contain the preset data that is first loaded when you enable a factory preset. Less/more files contain the preset data that is called up when you edit a factory preset via the Control application’s one-knob “less/more” editing procedure. If there is no less/more file for the specific less-more setting you choose, RSAP will automatically generate the data by interpolating between the contents of the two nearest less/more files.

Custom and User Presets

You can create custom “factory” presets that have full Less-More functionality. You can change the settings of a Factory Processing Preset, but if you want to preserve your changes, you must then store those settings as a User Preset, which you are free to name as you wish. You can also create User presets by editing existing user presets and saving the results under a new name. The suffix of User Presets is RSAPUSER. The Factory Preset remains unchanged.

Setup Presets Overview

Setup presets contain setup information, such as input levels, output levels, global Target Loudness, and operate/disable switches for various signal-processing blocks.

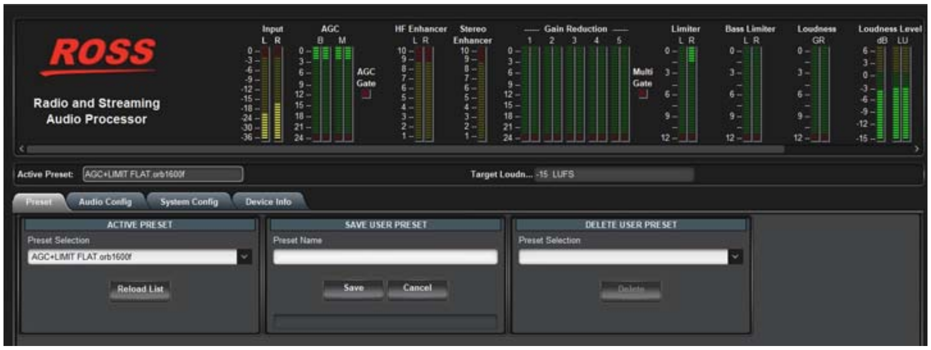
Recalling a Preset

The RSAP comes with presets that make it easy to create a sonic texture that suits your target audience. If you want to customize a preset, it is recommended to start with the options on the Audio Config > Less-More tab. From there you can adjust the other parameters in DashBoard as needed.

To recall a preset

1. Display the **RSAP** interface as outlined in “**To display the RSAP interfaces in DashBoard**” on page 39.

   The **Preset** tab is automatically selected.



2. Use the **Preset Selection menu** in the **Active Preset** area to select the preset to recall.

   The preset is automatically recalled and the **Active Preset** read-only field.

Editing a Preset

You can edit a Factory Preset or an existing User Preset, the process is the same for both preset types. When you modify an existing preset, whether Factory or User, the RSAP server software will automatically generate a temporary User Preset whose name consists of “Modified” appended to the front of the existing preset name. If you do not save your modifications, this temporary preset will remain on the server’s hard Drive until you further modify any preset. Then the temporary preset will be overwritten.You can name them as you wish, limited only by the file naming limits in your operating system. When editing a preset, you can choose to enable/disable the processing settings as required. The RSAP allows you to disable processing settings that you do not need. This can reduce input/output delay. Editing a preset requires you to:

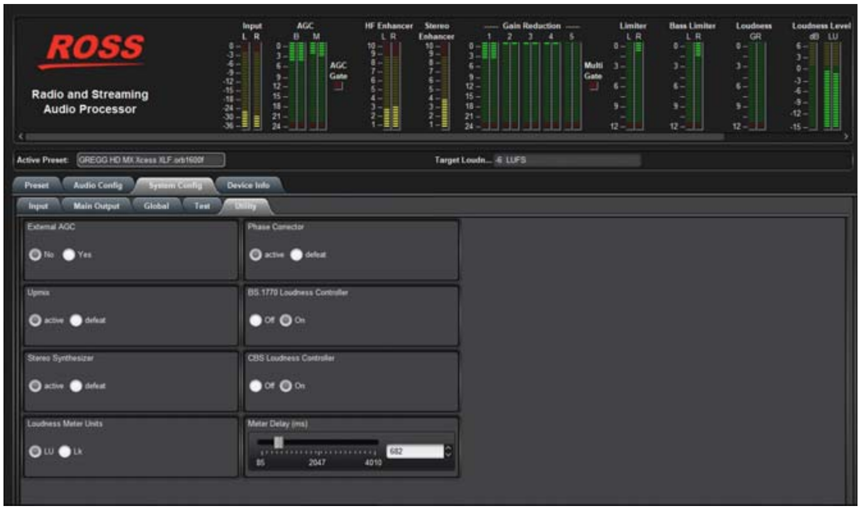
1. Recall a Factory Preset.
2. Use the provided slider(s) to adjusting the setting(s) of a preset to ensure the new value is applied.
3. Save the updated settings as a new User Preset.

Editing the Settings of a Preset

You can store as many User Presets as the RSAP host computer hard drive and operating system can accommodate. User Presets are shown on the “Open Preset” list by the name that you gave them when you saved them. This section briefly summarizes some of the Factory Preset settings you may wish to edit and then save as a User Preset.

To edit the settings of a preset

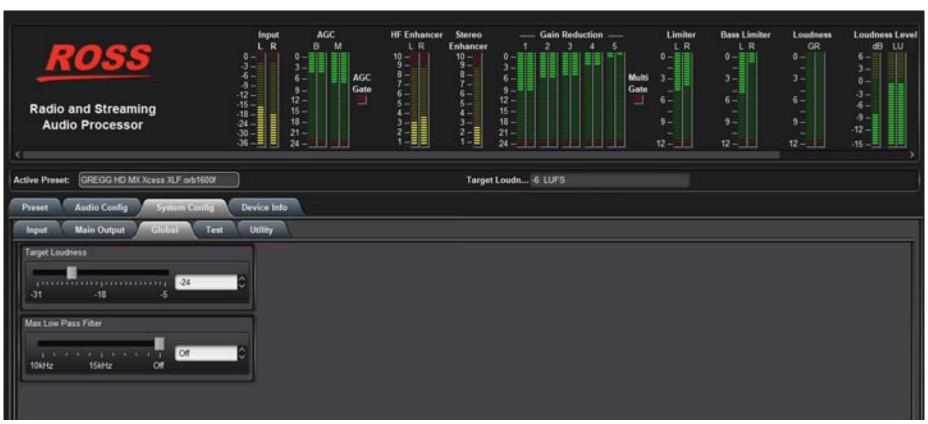
1. Recall a Factory Preset as outlined in “**To recall a preset**” on page 59.
2. Select **System Config** > **Utility**.



3. Enabling the **Stereo Synthesizer** adds a delay of approximately 187ms. This allows its automatic mode sensing to work (by sensing silence in the right channel). Disabling this setting removes the delay.
4. Enabling the **Phase Corrector** adds a 170ms of delay.
5. Enabling the **BS.1770 Loudness Controller** disables the BS.1770 Safety Limiter. Note that the BS.1770 audio meter runs regardless of the setting of this setting.
6. Enabling the **CBS Loudness Controller** disables the CBS loudness setting and loudness meter. The CBS loudness setting/meter relies on a large filter-bank, so it uses a significant amount if memory and disabling it can sometimes help achieve smooth operation on slower CPUs.
7. Use the **Loudness Meter**s to determine the label of the BS.1770 loudness meter and only affects the graphics in DashBoard. LkFS is typically used in the United States and LUFS in Europe.

To configure the global audio processing settings

1. Select **System Config** > **Global**.



2. Set the **Max Low Pass Filter Cutoff Frequency**.

To set the AGC Mode for your installation

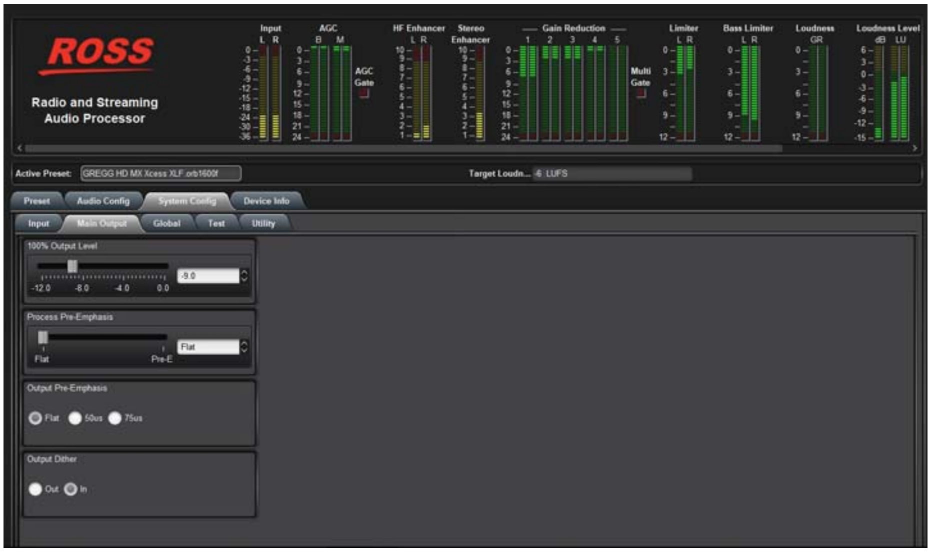
1. Select **Audio Config** > **AGC**.

2. Set the **AGC Mode** to **On**.

   This is recommended during setup to allow the Analog and Digital Input reference level alignment steps (below) to work correctly. After you have finished with these steps, set the AGC Mode appropriately for your installation.

To set the digital output properties for your installation

1. Select **System Config** > **Main Output**.



2. Adjust the **Output Dither** as required.

   RSAP can add “first-order high-pass” dither before any truncation of the output word. The amount of dither automatically tracks the setting of the Word Length setting.

   RSAP’s dither is first-order noise shaped dither that adds less noise in the midrange than white PDF dither. However, unlike extreme noise shaping, it adds a maximum of 3dB of excess total noise power when compared to white PDF dither. It is therefore a good compromise between white PDF dither and extreme noise shaping.

   In many cases, you will not need to add dither because the source material has already been correctly dithered and more than enough input dither is passed through to RSAP’s output. However, particularly if you use the

Noise Reduction feature, the processing can sometimes attenuate input dither so that it is insufficient to dither the output correctly. In this case, you should add dither within RSAP. It is safest to always add dither when operating with 16-bit output.

3. Set the **100% Output Level** to the amount of the peak headroom you need to match the downstream transmission channel.

   In some applications, it is important to leave peak headroom below 0dBFS to accommodate overshoots in the downstream transmission system. For example, when using a lossy codec like HE-AAC, it is wise to allow at least 1.5 dB of peak headroom to compensate for codec-induced overshoots.

   The 100% Output Level setting sets RSAP's maximum peak output level with respect to 0dBFS, which allows you to compensate for transmission channels that introduce peak overshoots. It does not change loudness. For example, if you lower an output setting from 0dBFS to -2dBFS, RSAP automatically reduces the gain following its peak limiter by 2dB and simultaneously increases the Drive into the peak limiter by 2dB. Hence, the average output level does not change but the maximum peak output level is constrained to -2dBFS[1]. This unconventional arrangement results from the RSAP's handling of Target Loudness.

To adjust the loudness using the active Target Loudness setting

✶ This setting is in the active processing preset in the Less-More tab. If the processing preset's Target Loudness is set to GLOBAL, then the RSAP instead uses the global SURROUND Target Loudness value in I/O SETUP > GLOBAL.

1. Select **Audio Config** > **Less-More**.

2. Set the **Target Loudness** parameters to achieve the desired target loudness and output headroom.

   If you have set Target Loudness correctly on the RSAP, you can change the output level setting freely without causing your loudness to be incorrect with respect to other transmissions.

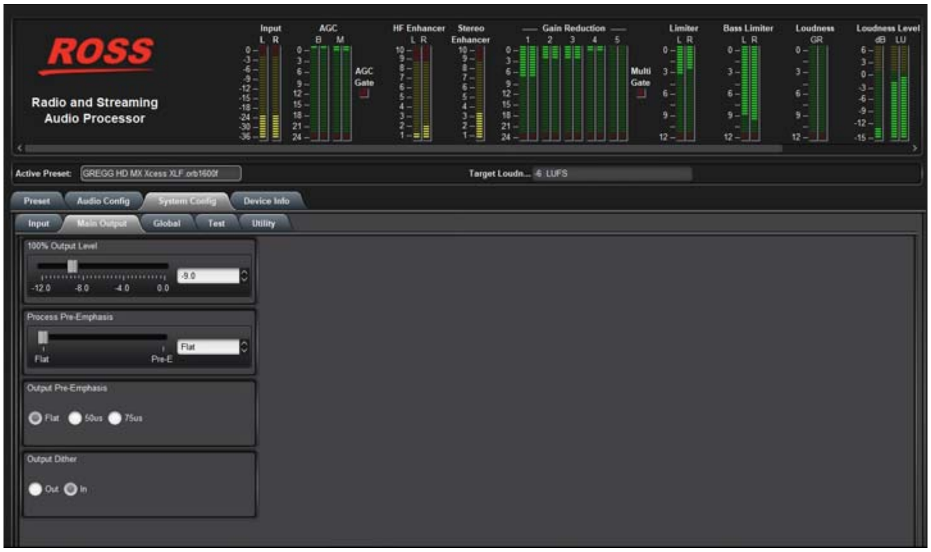
   The Target Loudness and output level settings are calibrated correctly if the transmission channel after the RSAP has unity gain. Because gain > 1 can cause clipping and gain < 1 will cause loss of headroom in the transmission channel, non-unity gain after the RSAP is bad engineering practice.

   The Less-More setting in a processing preset affects texture while changing loudness as little as possible. Set loudness by adjusting the active Target Loudness value. "TV" presets use the global Target Loudness value; other presets usually have local Target Loudness values.

3. Select **System Config** > **Main Output** and set the **Process Pre-Emphasis** curve on which the peak limiting operates.

---

1. If this causes large amounts of gain reduction to occur in the look-ahead limiter, the average level will decrease somewhat. However, this will only happen with pathologically high settings of Target Loudness and will never happen at the settings of Target Loudness typically used in broadcasts (–24 per ATSC A/85 or –23 per EBU R128).



Normally, RSAP will feed transmission channels that do not use pre-emphasis and you should set this setting to FLAT. 50µs and 75µs pre-emphasis are only useful if the RSAP is protecting a transmission link that uses pre-emphasis (like certain satellite up-links) or if you are using a TV# preset to Drive an analog aural transmitter (i.e., one that produces a pre-emphasized FM aural carrier in analog television transmission).

Use FLAT pre-emphasis for DAB, DRM, HD Radio, digital television, net-casts, and any other channel that uses a lossy codec. When in doubt, use FLAT pre-emphasis.

For these applications, you can apply pre-emphasis to the RSAP five-band compressor’s side-chain (to make the five-band compressor “pre-emphasis aware,” allowing it to be used as a high-Frequency limiter) and before the peak limiter so the limiter will setting the peaks of the pre-emphasized audio.

When the Output De-Emphasis setting is set to FLAT, a deemphasis filter after the look-ahead limiter restores “flat” audio at the RSAP output; otherwise, the output is pre-emphasized. When the output is deemphasized, the transmitter Driven by RSAP must restore the pre-emphasis before the signal is transmitted.

When in doubt, choose PRE-E and bypass the transmitter’s pre-emphasis. This prevents potential mismatches between the RSAP output deemphasis filter and the transmitter’s input pre-emphasis filter, which can introduce peak overshoots.

RSAP’s processing is tuned to be most effective with “flat media” and cannot provide “competitive” loudness for pre-emphasized radio channels. Because the RSAP is not sold or licensed as a processor for FM radio transmission, the Hard MX Overshoot Mode is locked out when pre-emphasis is not FLAT.

RSAP can process audio for transmission on a pre-emphasized analog television aural carrier because this application requires lighter processing than FM radio. The TVA# factory presets have been created for this application.

We strongly encourage you to use the MX Limiter if you have configured RSAP to use pre-emphasis. This will produce far fewer gain-pumping artifacts than will non-MX operation.

If lack of CPU power forces you to use the non-MX Limiter, only use it for light-to-moderate protection limiting with a low duty cycle. Otherwise, you may hear pumping on material with a lot of high Frequency energy like sibilance and cymbals. The solution is lowering the MB Final Limit Drive setting until the problem is no longer audible.

If you find that lowering the MB Final Limit Drive causes too much loudness loss, use the Band 5 compressor as a high Frequency limiter to compromise between loudness and limiting artifacts. Set B5 Delta Release to +6 and B5 Stereo Coupling to OFF. Adjust B5 Threshold setting until you see gain reduction on the Band 5 GR meter with problematic material. Continue to lower the B5 Threshold setting until you no longer hear gain pumping. Instead, you will probably hear some high Frequency loss. This loss is less subjectively objectionable than gain pumping.

✶ You can use the AGC+LIMITER and TVA factory presets as examples of how to do this.

To adjust the input reference level setting

✶ This section adjusts the Drive to the RSAP's audio processing so that it operates in its preferred range.

1. Select **Audio Config** > **Less-More**.

2. Set the **Active Preset** to **Gregg HD Med**.
3. Feed normal program material to the RSAP, peaking at the level to which you normally peak program material (typically 0VU if your console uses VU meters).
4. Select **System Config** > **Utility**.
5. Verify that the **External AGC** is set to **No**.
6. Select **System Config** > **Input**.
7. Set the **Input Reference Level** to make the **AGC GAIN REDUCTION** meter indicate an average of 10dB gain reduction when normal levels are applied to the RSAP input. Refer to "**Audio Meters in DashBoard**" on page 113.
8. If the **AGC GAIN REDUCTION** meter averages less than 1 dB gain reduction (higher on the meter), or if the GATE indicator stays on when program material is present, decrease the Input Reference Level. If the AGC GAIN REDUCTION meter averages more gain reduction (lower on the meter), increase the **Input Reference Level**.

   The VU and PPM calibrations are arranged so that the PPM scale is 7dB higher than the VU. This approximates the difference in indication between a VU meter and PPM with dynamic program material.

✶ The Input Reference Level setting is calibrated to comply with industry standard levels. A typical reference level is -18dBFS (VU) for EBU countries. SMPTE standard reference level is -20dBFS (VU). If you are using one of those reference level, it is usually acceptable to set the Input Reference Level setting (VU) to this reference level.

9. When finished, reset the **AGC** to **DISABLED**, if required.

Customizing Processing Presets to Achieve Specified Target Loudness

This section outlines the options available when adjusting Target Loudness.

Radio-Style Presets

Most non-TV presets have local Target Loudness values (i.e. Target Loudness in the Less-More tab is not set to GLOBAL). We chose these values to represent competitive loudness/distortion trade-offs in traditional, non-R128 applications — their loudness is much higher than -23LUFS and represents the typical loudness encountered in non-R128 net-casts.

The fact that the presets have local Target Loudness is a compromise between the needs of users who process for specified target loudness and those who process more traditionally. You can adjust the loudness of any such preset by changing its Target Loudness value to the desired target loudness or to GLOBAL. The GLOBAL setting is convenient for those processing for government-specified target loudness. After you have adjusted Target Loudness, Less-More continues to be available and adjusts texture (usually, the amount of gain reduction in the multi-band compressor) while changing loudness as little as possible.

Because of their limited graphic resolution, the CBS and BS.1770 loudness meters are calibrated relative to the active Target Loudness value: “0” always corresponds to the active Target Loudness value. Hence, adjusting Target Loudness will not change the loudness meter indications. This makes it easy to customize presets. After you have finished your coloration and texture adjustments, fine-tune the Limiter Drive to make the loudness meters indicate “0” on average.

To achieve most dynamic-sounding audio, the CBS and BS.1770 loudness settings are turned off in the radio-style presets. Therefore, they will produce loudness meter indications that are somewhat less consistent than those produced by the “TV” presets. Nevertheless, they will always provide excellent subjective source-to-source consistency.

TV Presets

The “TV” factory presets do not need customization to achieve correct loudness because out of the box, they provide loudness equal to the global Target Loudness value (as set in the Utilities tab). In almost every application, a factory preset (possibly as customized with Less-More, which preserves correct loudness) will suffice for a given application. However, for the sake of completeness, we provide instructions below describing how to customize them while retaining the correct loudness.

If you are processing for specified target loudness in television or other video applications and wish to customize a factory preset, be aware of two major philosophies, one presented in ATSC Recommendation A/85 and the other in EBU Recommendation R128.

ATSC A/85

If you wish to ensure that dialog levels are consistent from one program segment to the next, use the the CBS loudness meter as a reference for adjusting a user preset’s loudness. In this case, we prefer relying on the CBS Loudness Controller to do final loudness setting and disabling the **BS.1770 Safety Limiter**, although you may leave it on if your organization demand strict adherence to BS.1770 meter readings. Consistent loudness of the “anchor element” (usually dialog) is the goal suggested in ***ATSC Recommendation A/85***, which is the basis for CALM Act compliance, and we prefer this goal to strictly relying on the BS.1770 loudness meter.

EBU R128

If you wish to ensure that integrated loudness as measured on a **BS.1770-2** meter is consistent from one piece of program material to the next, use the **BS.1770 Loudness** meter as a reference. In this case, you may wish to enable the **BS.1770 Safety Limiter**, which will constrain BS.1770 integrated loudness with a sliding 10-second integration window from exceeding the setting of the BS.1770 Threshold setting. However, note that R128 encourages mixing to achieve a wide dynamic range, where some material may considerably exceed the target loudness when measured on a “short-term” (three-second integration time; ungated) BS.1770 meter. Moreover, R128 requires online loudness setting to be disabled if upstream material is known to be pre-processed such that the integrated loudness of each program segment (per BS.1770) is identical to the active Target Loudness value with a ±0.5 LUFS window. To do this, use enable the Pass-Through Switch in your active preset and save the result as a user preset.

The CBS and BS.1770 loudness meters’ calibrations track the RSAP’s active Target Loudness value, which is either (1) the Target Loudness value (in the System tab) or (2) the Target Loudness value in the active processing preset if this value is not set to GLOBAL.

To customize Processing Presets to Achieve Specified Target Loudness

1. Make sure that the RSAP's active Target Loudness value is the same as the dialnorm or Target Loudness metadata you are transmitting to consumers' receivers.
   - For U.S. program providers, this ensures that the RSAP-processed transmission will meet the requirements of the ***CALM Act***.
   - ***EBU Tech 3344*** has suggestions about setting target loudness when targeting receivers and player devices that cannot receive dialnorm or Target Loudness metadata from the signals they are reproducing.
2. Recall the **TV 5B GENERAL PURPOSE** preset. Refer to "**To recall a preset**" on page 59 for details.
3. Select the **Setup** tab.
4. Set the **Input Reference Level** to 1 dB of AGC gain reduction with typical input material.
5. Reccall the preset you intend to edit.
6. If the AGC's idle gain (i.e. the gain reduction produced by the AGC when its silence gate is on) is incorrect, adjust it with the **AGC Idle Gain** setting. It is usually adjusted so that the idle gain is equal to the gain reduction that occurs with normal program material. Its normal setting is 0..

✶ If you are using a factory processing preset and you have adjusted the input reference level correctly, there is no need to adjust the AGC Drive or AGC Idle Gain settings.

7. Adjust the Multiband Drive to produce the desired amount of multi-band gain reduction. We recommend about 5dB for dialog at normal levels.
8. If you wish to use the CBS Loudness Controller:
   a. Set the Loudness Threshold to 0dB, which matches the CBS Loudness Controller's threshold to the RSAP Active Target Loudness value.
   b. Set the MB Limit Drive so that the CBS segment of the LOUDNESS GR meter indicates 3dB of gain reduction with dialog at normal levels. (The CBS Loudness Controller's gain reduction appears in blue on the LOUDNESS GR meter.)

      You should see the LOUDNESS LEVEL meter peaking around 0dB when the LOUDNESS GR meter shows that gain reduction is occurring.
9. If you wish to use the BS.1770 Safety Limiter, set the **BS.1770 Threshold** to your preferred level.

   0LU provides the tightest setting as indicated on a BS.1770 meter, but this is likely to cause program material with an usually low peak to RMS ratio to sound too quiet. Most natural sound is produced when the BS.1770 Threshold setting is set to +2 LU or higher.
10. If you do not wish to use the Loudness Controller:
    a. Set the Loudness Threshold to OFF.
    b. Set the BS.1770 LIMIT Threshold to OFF.
    c. Adjust the MB Limiter Drive setting to produce an average of 0LU on the BS.1770 loudness meter with program material at normal levels.

✶ Presets with the loudness setting off will produce wider source-to-source loudness variation (as indicated on the BS.1770 and CBS loudness meters) than presets with the loudness setting on. However, they will sound more dynamic and natural while still achieving good subjective loudness consistency.

✶ If you chose an appropriate Target Loudness value for your transmission, the RSAP's limiting meters should rarely indicate any gain reduction. This means that the target loudness is well matched to the headroom in the transmission system.

11. Set the Pass-Through Gain (optional).

✶ The ***EBU Recommendation R128*** requires on-line loudness settings to be disabled so that they do not change the dynamics of upstream material that is known to meet the R128 requirement: the loudness of each program segment must be within a ±0.5 LU window centered on the active Target Loudness value. This measurement must be done over the entire program segment using a BS.1770-2 (or higher) “integrated” measurement.

✶ You may operate any processing preset in Pass-through mode.

a. Use the Pass-Through Switch to enable or disable pass-though mode

b. Set the Pass-Through Gain setting (in the Less-More tab) as needed.

   When set to 0dB, RSAP provides unity gain from input to output regardless of the settings of any settings except the Pass-Through Gain setting. The Input Reference Level and 100% Output Level settings do not affect the gain. The normal setting is 0dB, because this retains the target loudness of the source material.

12. If you need a pass-through preset you can recall conveniently, set the Pass-Through Switch in any factory preset to IN and save the result as a User Preset.

    MX presets have a different input/output delay than non-MX presets. A Pass-through user preset should have the same MX Limiting setting as the active preset you use when not in Pass-through mode. We recommend starting with your active preset and then editing it to produce your Pass-through user preset.

    To protect the output channel from clipping, the peak limiter remains in-line during Pass-through operation and the 100% Output Level setting operates normally to specify the maximum peak level at the output with respect to 0dBFS. If you set the Pass-Through Gain setting higher than 0 dB or if the 100% Output Level setting is set lower than 0dBFS, it is possible that you will observe limiter gain reduction. If you need to use substantial amounts of gain reduction, you will get best results by starting with an MX preset.

Adjusting the Target Loudness

The RSAP uses the contemporary concept of “target loudness” to increase listener satisfaction by minimizing the need for listeners to readjust their volume settings when changing between different broadcast stations or netcast streams. If you specify the target loudness (via the Target Loudness setting), the RSAP will produce the desired loudness.

What is Target Loudness?

Target loudness is also known as “dialnorm,” particularly in sound-for-picture applications. The term dialnorm is short for “dialog normalization,” and was first specified by Dolby® Laboratories as part of the metadata for their AC3 (later, “Dolby Digital”) codec. The dialnorm was originally intended to allow motion pictures to be exhibited theatrically with fixed and predictable dialog loudness while allowing other program elements to be louder or quieter than dialog according to the artistic demands of the material. Dolby has since updated their recommended loudness measurement practices to use the BS.1770 standard.

The advantage of enforcing target loudness is that it causes the loudness of various program elements in a broadcast or netcast to be roughly consistent with each other, so that it is less likely that users will need to adjust their volume settings between program segments. The eventual goal is to achieve consistent loudness across all media, like FM radio and netcasts, supported by a given receiver or player device.

By specifying a global Target loudness value, the user can be sure that any processing preset automatically produces the desired target loudness. For the sake of consistency and because the world is moving in the direction of specifying target loudness for all deliverables, the RSAP incorporates the concept of target loudness in all audio processing in two ways:

- It allows the target loudness to be assigned to each processing preset individually. This allows “radio-style” presets to be designed in the traditional way by balancing loudness against distortion, as loudness is specified by a given preset’s local Target loudness value. Most “radio-style” factory presets have a target 1-18 loudness of -8 LUFS6, although this can easily be adjusted to suit your goals.
- It allows the target loudness to be assigned globally (as part of the system parameters) so that it applies to any preset whose Target Loudness value is set to GLOBAL. When you edit a processing preset so that its Target Loudness value is GLOBAL, this causes the loudness produced by the preset to be the same as all other presets with GLOBAL Target Loudness. All of the RSAP’s sound-for-picture presets have GLOBAL Target Loudness, and all radio-style presets can be easily edited to have GLOBAL Target Loudness.

The RSAP loudness meters indicate loudness relative to the active target loudness, such that “0” on the meter corresponds to the target loudness.

Automatic Loudness Control

To make automatic loudness setting as straightforward and dependable as possible, the RSAP adjusts the output level setting but does not change loudness; it only sets the amount of headroom between 0dBFS and the maximum peak output level that the audio processing produces. This allows you to adjust the processing to compensate for downstream overshoots from codecs without changing loudness. (Instead, the processing produces more peak limiting.) For example, for the HE-AAC codec it is recommended to allow 1.5dB of peak headroom by setting the output level setting to -1.5dBFS.

When working with Dolby Digital® transmission systems and other systems having a specified target BS.1770 loudness, it is recommended to specify what value of Dolby Digital dialnorm metadata (which is the same as BS.1770 target loudness) you are transmitting to your audience, or the target loudness of the receiver if it is not metadata-aware. This will prevent your transmission from being too loud or quiet compared to other correctly set up transmissions.

In a bitstream that includes dialnorm metadata, setting Target Loudness to a less negative value automatically turns down the home receiver’s volume setting, so Radio and Streaming Audio Processor output level must be turned up by the same amount to maintain a constant loudness at the receiver. Because it is placed before RSAP’s look-ahead limiter, RSAP’s hidden Target Loudness gain setting achieves this while allowing RSAP’s look-ahead limiter to prevent digital clipping in the downstream transmission chain regardless of RSAP’s Target Loudness setting. This arrangement allows the user to set the correct loudness at RSAP’s output solely by adjusting the RSAP active Target Loudness value—it is unnecessary to adjust any other settings within a factory processing preset.

For recommendations regarding player devices that are not Target Loudness-aware, see ***EBU - TECH 3344: Practical Guidelines*** for distribution systems in accordance with EBU R128.

Setting the Output Loudness

To set the RSAP’s output loudness, adjust its Target Loudness value (either within the active processing preset or globally), or adjust the MB Limiter Drive setting in the active processing preset.

Adjusting Target Loudness changes output loudness without changing the indication on RSAPs loudness meters (which show loudness relative to the active Target Loudness value) or the amount of gain reduction in the loudness setting, if turned on. The peak limiter’s gain reduction will change. This is the preferred method if the RSAP’s loudness setting is active because it has the smallest effect on the sonic texture of the RSAP’s audio processing.

Adjusting MB Limiter Drive changes the loudness meters’ indications and the amount of gain reduction in the loudness setting and peak limiter. If RSAP’s loudness settings are turned off (as they are in most radio-style presets), you may have to tweak the MB Limiter Drive to fine-tune the loudness to your program material, such that RSAP loudness meters peak around “0.”

Your transmission’s loudness will automatically be correct if:

- The Loudness Threshold and BS.1770 Safety Limiter Threshold settings are set to “0.” (If these are OFF, you may have to tweak the MB Limiter Drive setting as explained above.)
- you have adjusted the RSAP Input Reference Level so that the processing operates with normal amounts of gain reduction (typically 10dB of AGC gain reduction) and;
- you have adjusted the RSAP’s Target Loudness to match the dialnorm metadata you are send to your audience or the target loudness on player devices that do not accept dialnorm metadata.

The RSAP Target Loudness value can be set in two places:

- There is a global setting in the active I/O Setup, which can be overridden by a setting in Less-More tab of the active processing preset. All sound-for-picture factory processing presets are configured to use the global Target Loudness setting specified in the active Setup.
- All “radio-style” processing presets have local Target Loudness settings, which makes their loudness/distortion/punch trade-off independent of the setting of global Target Loudness. You can make any

"radio-style" preset produce the global target loudness by setting the preset's local Target Loudness value to GLOBAL and then saving the result as a User Preset.

✶ A given preset's Less-More setting continues to be available after you have changed the preset's Target Loudness.

To better match other streams in "radio-style" applications, you may wish to set a target loudness that is substantially higher than -23LUFS. If you see a substantial amount of gain reduction on RSAP LIMITER GR meters, we recommend enabling the MX Limiter (in the Distortion tab of the active processing preset). This can substantially improve subjectively quality by reducing or eliminating audible peak-limiting artifacts. However, it will approximately double the CPU usage for that Processor.

If you are using a mono-mode Processor, its BS.1770 and CBS loudness meters will read about 3dB lower than they do in stereo mode. This is because there is only one audio channel contributing to the loudness measurement. However, the mono output is duplicated on the left and right channels of the output sound device, and if the mono signal is played out through both channels, this will increase the loudness by 3dB.

You must use your best judgment as to how to set up loudness in mono; it depends on listening context.

To set the target loudness for the Active Preset

1. Display the **RSAP** interface as outlined in "**To display the RSAP interfaces in DashBoard**" on page 39.
2. Select **Audio Config** > **Less-More**.

3. Use the **Target Loudness** slider to set the target loudness value for the active preset.

To set the global target loudness for the RSAP

1. Display the **RSAP** interface as outlined in "**To display the RSAP interfaces in DashBoard**" on page 39.
2. Select **Audio Config** > **Less-More**.
3. Use the **Target Loudness** slider to set the Target Loudness to Global.
4. Select **System Config** > **Global**.

5. Use the **Target Loudness** slider to adjust the overall Target Loudness for the RSAP.

✶ This impacts all Target Loudness values and will override the Active Preset value even if a different preset is selected.

Customizing RSAP Sound

The subjective setup settings on RSAP give you the flexibility to customize your station's sound. Nevertheless, as with any audio processing system, proper adjustment of these settings consists of balancing the trade-offs between loudness density, and audible distortion. The following sub-sections provide the information you need to adjust RSAP settings to suit your format, taste, and competitive situation.

When you start with one of the Factory presets, there are two levels of subjective adjustment available to you to let you customize the Factory preset to your requirements: Basic Control and Advanced Control.

✶ The Equalization, Stereo Enhancer, and Less-More labs access the Basic Modify settings. The remaining tabs show the Advanced Modify settings, logically organized by functionality.

Basic Control

The single Less-More setting changes many different subjective setup setting settings simultaneously. Most RSAP users will never need to go beyond the Less-More level of setting.

The Less-More setting has been designed keep loudness constant when adjusted. Instead, turning it up typically changes texture by increasing the multi-band compressor Drive (and this the amount of multi-band compressor gain reduction), while decreasing the peak limiter Drive appropriately.

To change the loudness for a preset, change the value of Target Loudness, which you can do by editing the local value of Target Loudness in the processing preset (Less-More tab) or setting that value to GLOBAL and then changing the global Target Loudness value. Changing the value of Target Loudness will change the Drive level into the peak limiter (and hence, will change its gain reduction). This does not change the indication of the loudness meter because it indicates loudness relative to the active Target Loudness value.

In the "video" presets, the Less-More setting also sets the average amount of dynamic range setting provided by the processing. As you go from less to more, the loudness of loud sounds will stay about the same but the loudness of quieter sounds will increase. Very quiet material like background sounds, quiet underscoring, hiss, and hum will not be pumped up.

All User Presets are created by modifying Factory Presets or by further modifying Factory Presets that have been modified previously. It is advised to set the Less-More setting to achieve a sound as close as possible to your desired sound before you make further modifications at the Advanced Modify level. This is because the Less-More setting gets you close to an optimum trade-off between loudness and artifacts, so any changes you make are likely to be smaller and to require resetting fewer settings.

The Less-More affects only the dynamics processing (compression and peak limiting) — the equalization and stereo enhancement are decoupled from Less-More. You can therefore change EQ or stereo enhancement and not lose the ability to use Less-More. When you create a user preset, RSAP will automatically save your EQ and stereo enhancement settings along with your Less-More setting. When you recall the user preset, you will still be able to edit your Less-More setting if required.

Advanced Control

Use the Advanced Control settings if:

- you want to create a signature sound for your broadcast or netcast that is far out of the ordinary;
- your standards differ from the provided Less-More settings;
- you are using RSAP in mastering or production applications

You can customize or modify any subjective setup values to create a sound as required. You can then save the settings in a User Preset and recall it later. This sort of customization is usually unnecessary for sound-for-picture but can be very useful for radio, production, and mastering applications.

Compressor attack time, release time, and threshold settings are available. We recommend that you create custom presets at the Advanced Modify level only if you are experienced with audio processing sound design and if you are willing to take the time to double-check your work on many different types of program material.

In production and mastering applications, you will usually be working with one piece of program material at a time. Here, you can use all of the Advanced Control settings to get the sound you want without being concerned about how your settings will sound with other material.

✶ Once you have edited the dynamics parameters if a preset in Advanced Modify, Less-More setting is no longer available in Basic Modify. As noted above, we strongly recommend using the Less-More setting to achieve a sound as close as possible to your desired sound before you make further modifications at the Advanced Modify level.

Gain Reduction Metering

Because it uses floating point processing, the RSAP has essentially unlimited amounts of available gain reduction. However, the meter should never exceed 25dB gain reduction if RSAP has been set up for a sane amount of gain reduction under ordinary program conditions. If any AGC or compressor gain reduction meter reads full-scale, this is usually a sign that you are using too much gain reduction, which can cause unpleasant compression artifacts.

The peak limiter gain reduction meters have 12dB of range. If you are using the MX Limiter, it is common for these meters to go to fully scale briefly without audibly objectionable consequences. However, you must assess this by ear for the program material you are processing.

Factory Programming Presets

Factory Programming Presets are the "factory recommended settings" for various program formats or types. The Factory Programming Presets are starting points to help you get on the air quickly.

You can easily edit any of these presets with the Less-More setting to optimize the trade-off between loudness and distortion according to the needs of your format, although this is often unnecessary. The presets are editable because other sound designers may have different requirements than the preset default settings. Start with one of these presets. Spend some time listening critically to your sound. Listen to a wide range of program material typical of your format and listen on several types of audio systems (not just on your studio monitors). Then, if you wish, customize your sound using the information in the Protection Limiter, Two-Band and Five-Band sections that follow.

Each factory preset has full Less-More capability. This section describes the presets, including the source presets from which they were taken and the nominal Less-More setting of each preset. Some of the Five-Band presets appear several times under different names because we felt that these presets were appropriate for more than one format; these can be identified by a shared source preset name.

✶ Each named preset is actually 19 presets that can be accessed via the Less-More setting. Try using this setting to trade off the amount of dynamic range reduction against processing artifacts and side effects. Once you have configured the Less-More settings, save your edited preset as a User Preset.

You can choose a preset other than the one named for your programming if you believe this other preset has a more appropriate sound. Also, if you want to fine-tune the Frequency balance of the programming, use Basic Modify and make small changes to the Bass, Mid EQ, and HF EQ settings. The RSAP enables changes in EQ (and stereo enhancement) without losing the ability to use Less-More settings.

✶ Less-More is still available for the unedited preset if you want to go back to it.

Protection, AGC and Mastering Presets

These presets are a useful starting point, but we expect that any mastering engineer would tweak them for each piece of program material being mastered.

Table 14.1 Protection, AGC, and Mastering Presets

Preset Name	Source Preset	Less-More	Loudness
AGC+ LIMIT FLAT	AGC+ LIMIT FLAT	5.0	-15 LUFS
AGC+ LIMIT PRE-E	AGC+ LIMIT 50us	5.0	-15 LUFS
ROCK MASTERING MXH	ROCK MASTERING MXH	5.0	-6 LUFS
ROCK MASTERING MX	ROCK MASTERING MX	5.0	-6 LUFS
PEAK LIMIT MX	PEAK LIMIT MX	1.0	-16 LUFS
SOFT K	SOFT Knee	5.0	-12 LUFS
SOFT Knee MX	SOFT Knee MX	5.0	-12 LUFS

AGC+LIMIT [FLAT, PRE-E]

These presets allow RSAP to serve as a studio AGC, substituting for the AGC at a radio or television transmitter and providing protection limiting for the STL that links the output of RSAP to the input of the RSAP at the transmitter.

If Proc Pre-Emphasis is FLAT, use the AGC+LIMIT FLAT preset; otherwise, use AGC+LIMIT PRE-E. These presets are identical, except that AGC+LIMIT PRE uses the MX Limiter and AGC+LIMIT FLAT does not. (The MX Limiter is much less vulnerable to audible gain pumping caused by large amounts of pre-emphasized high Frequency energy.)

✶ It is common to adjust the AGC attack times, release times, etc. to your preference and then save the result as a user preset.

Peak Limiter MX

The Peak Limiter MX preset turns off all processing other than the peak limiter to implement a fast true-peak protection limiter with very high performance. In the LIMITER tab, you may choose either SOFT or HARD MX overshoot limiter modes. HARD can be useful if you need to produce maximum loudness. The highest usable loudness available in HARD mode is approximately -4 LUFS, but we recommend using -6 LUFS or lower to achieve best quality. Attempting to go above -6 LUFS quickly sacrifices bass punch.

The subjective differences between SOFT and HARD are subtle. SOFT sounds cleaner, but can produce some audible gain pumping when Driven very hard. HARD is closer to a “clipper sound,” so it is not quite as clean sounding but does not produce gain pumping.

The loudness meters continue to work, and continue to be calibrated such that 0 is equal to the setting of the Target Loudness setting. However, they are located before the peak limiters, so if you are driving the peak limiters hard so that their limiting action alone changes loudness, you should monitor the output of RSAP with an external loudness meter.

✶ If you use HARD mode with the OPTIMIX SWITCH set to ACTIVE, you may encounter audio stuttering. The stuttering will normally stop if you change the setting that caused it. Clearing this issue may require stopping and restarting the Service.

SOFT Knee

These presets (in MX and non-MX versions) are zeroed-out starting points for mastering applications, particularly where soft knee compression is desired. These presets must be manually tweaked to complement the program material being processed.

These presets are phase-linear. They set all equalization flat and turn off the AGC. The multi-band compressor (two-band or five-band) is set to supply a very soft-knee compression characteristic with approximately 5-10dB of gain reduction. The ratio for a given compressor starts out at 1:1 and ends up at :1 when the input level is 20dB above threshold.

Mastering engineers will certainly want to adjust the compression thresholds and band coupling to complement the program material. The ratio and knee settings are adjustable separately for each band's compressor. For example, one might want to use a low ratio and soft knee in bands 1-4 while using a higher ratio and/or harder knee in Band 5.

The RSAP powerful equalization section is also available. Additionally, these presets set the look-ahead limiter Drive setting conservatively, which ensures highest quality. However, the RSAP look-ahead limiter can be Driven quite hard without objectionable side effects, so the RSAP can create competitively loud masters. For "loudness war" masters, start with the MX version.

ROCK MASTERING [MX, MXH]

These presets demonstrate how RSAP can be used as a mastering processor to make clean, bright, detailed, punchy, and very loud masters. They are based on the SOFT Knee MX preset, but with substantial customization. They work best when fed mixes that have not previously had dynamics processing (such as wide-band compression) applied to the entire mix.

✶ Rock Mastering MXH uses HARD MX LIMIT OVERSHOOT MODE.

Radio-Style Presets

Most presets with names that are specific to a programming format (like "Modern Rock") duplicate other presets, called the "Source Presets" in **Table 14.2** All of the presets with format-specific names use the MX Limiter. To create a non-MX format-specific preset, use the non-MX version of the format-specific preset's "Source Preset."

Keep the following naming conventions in mind when referencing **Table 14.2**:

- If a preset name contains "XLF," the subharmonic synthesizer is active.
- If a preset name contains "MX," the MX peak limiter is active in SOFT overshoot limiting mode.
- If a preset name contains "MXH," the MX peak limiter is active in HARD overshoot limiting mode.

Table 14.2 Radio-style Presets

Preset Name	Source Preset	Less-More	Loudness
Adult Contemp MX MED	GREGG HD MX Med XLF	9.0	−7 LUFS
Adult Contemp MX OPEN	GREGG HD MX Open XLF	9.0	−10 LUFS
Alternative MX	GREGG HD MX Med XLF	9.0	−7 LUFS
Ambient MX LIGHT	GREGG HD MX Light	7.0	−11 LUFS
Ambient MX OPEN	GREGG HD MX Open XLF	9.0	−10 LUFS
Chill MX LIGHT	GREGG HD MX Light	7.0	−11 LUFS
Chill MX OPEN	GREGG HD MX Open XLF	9.0	−10 LUFS
CHR MX HEAVY	GREGG HD MX Xcess XLF	9.0	−6 LUFS
CHR MX MED	GREGG HD MX Med XLF	9.0	−7 LUFS
CLASSICAL-5 BAND MX	CLASSICAL-5 BAND MX	7.0	−14 LUFS
CLASSICAL-5 BAND	CLASSICAL-5 BAND	7.0	−14 LUFS
CLASSICAL-5 BAND Parallel	CLASSICAL-5 BAND Parallel	5.0	−14 LUFS
CLASSICAL-5B+AGC MX	CLASSICAL-5B+AGC MX	5.0	−14 LUFS

Table 14.2 Radio-style Presets

Preset Name	Source Preset	Less-More	Loudness
CLASSICAL-5B+AGC	CLASSICAL-5B+AGC	5.0	−14 LUFS
CLASSICAL-5B+AGC Parallel	CLASSICAL-5B+AGC Parallel	5.0	−14 LUFS
Classic Hits MX	GREGG HD MX Heavy XLF	10.0	−7 LUFS
Classic Rock MX	GREGG HD MX Med XLF	9.0	−7 LUFS
Country MX	GREGG HD MX Med XLF	9.0	−7 LUFS
Dance MX	GREGG HD MX XXLF	9.0	−8 LUFS
Easy Listen MX LIGHT	GREGG HD MX Light	7.0	−11 LUFS
Easy Listen MX OPEN	GREGG HD MX Open XLF	9.0	-10 LUFS
GOLD MX	GOLD MX	9.5	−8 LUFS
GOLD	GOLD	5.0	−10 LUFS
Gold MX HEAVY	GREGG HD MX Heavy XLF	10.0	−7 LUFS
GREGG HD MX Med Dry	GREGG HD MX Med Dry	9.0	−9 LUFS
GREGG HD MX Med Dry XLF	GREGG HD MX Med Dry XLF	9.0	−9 LUFS
GREGG HD MX Heavy	GREGG HD MX Heavy	7.0	−7 LUFS
GREGG HD MX Heavy XLF	GREGG HD MXH Heavy XLF	10.0	−5 LUFS
GREGG HD MXH Heavy XLF	GREGG HD MX Heavy XLF	10.0	−7 LUFS
GREGG HD MX Light	GREGG HD MX Light	7.0	−11 LUFS
GREGG HD MX Light XLF	GREGG HD MX Light XLF	7.0	−11 LUFS
GREGG HD MX Med	GREGG HD MX Med	9.0	−7 LUFS
GREGG HD MX Med XLF	GREGG HD MX Med XLF	9.0	−7 LUFS
GREGG HD MX Open	GREGG HD MX Open	9.0	-10 LUFS
GREGG HD MX Open XLF	GREGG HD MX Open XLF	9.0	-10 LUFS
GREGG HD MX Xcess	GREGG HD MX Xcess	9.0	−6 LUFS
GREGG HD MX Xcess XLF	GREGG HD MX Xcess XLF	9.0	−6 LUFS
GREGG HD MXH Xcess XXLF	GREGG HD MXH Xcess XXLF	10.0	–7 LUFS
GREGG HD MX XXLF	GREGG HD MX XXLF	9.0	−8 LUFS
GREGG HD MX XXXLF	GREGG HD MX XXXLF	10.0	–8 LUFS
GREGG HD Med Dry XLF	GREGG HD Med Dry XLF	9.0	−11 LUFS
GREGG HD Med Dry	GREGG HD Med Dry	9.0	−11 LUFS
GREGG HD Heavy	GREGG HD Heavy	7.0	−9 LUFS
GREGG HD Heavy XLF	GREGG HD Heavy XLF	10.0	−9 LUFS
GREGG HD Light	GREGG HD Light	7.0	−11 LUFS
GREGG HD Light XLF	GREGG HD Light XLF	7.0	−11 LUFS
GREGG HD Med	GREGG HD Med	9.5	−9 LUFS
GREGG HD Med XLF	GREGG HD Med XLF	9.0	−9 LUFS
GREGG HD Open	GREGG HD Open	9.0	−10 LUFS

Table 14.2 Radio-style Presets

Preset Name	Source Preset	Less-More	Loudness
GREGG HD Open XLF	GREGG HD Open XLF	9.0	−10 LUFS
GREGG HD Xcess	GREGG HD Xcess	9.0	−8 LUFS
GREGG HD Xcess XLF	GREGG HD Xces XLF s	9.0	−8 LUFS
GREGG HD XXLF	GREGG HD XXLF	9.0	-10 LUFS
Hip-Hop MX	GREGG HD MX XXLF	9.0	−8 LUFS
Indie MX	GREGG HD MX Med XLF	9.0	−7 LUFS
Jazz MX	POP MX OPEN FLAT	9.0	−10 LUFS
Lounge MX LIGHT	GREGG HD MX Light	9.0	−10 LUFS
Lounge MX OPEN	GREGG HD MX Open XLF	9.0	−10 LUFS
Modern Rock MX	GREGG HD MX Med XLF	9.0	−7 LUFS
NEWS-TALK MX	NEWS-TALK MX	7.0	-10 LUFS
NEWS-TALK	NEWS-TALK	7.0	−10 LUFS
Oldies MX	GREGG HD MX Heavy XLF	10.0	−7 LUFS
POP MX XLF	POP MX XLF	9.0	−8 LUFS
POP MX FLAT EQ	POP MX FLAT EQ	9.0	−8 LUFS
POP MX LIGHT FLAT	POP MX LIGHT FLAT	7.0	−11 LUFS
POP MX OPEN FLAT	POP MX OPEN FLAT	9.0	−10 LUFS
POP XLF	POP XLF	9.0	−10 LUFS
POP FLAT EQ	POP FLAT EQ	9.0	−10 LUFS
POP LIGHT FLAT	POP LIGHT FLAT	7.0	−11 LUFS
POP OPEN FLAT	POP OPEN FLAT	9.0	−10 LUFS
R and B MX	GREGG HD MX XXLF	9.0	−8 LUFS
Rap MX	GREGG HD MX XXLF	9.0	−8 LUFS
Reggae MX	GREGG HD MX XXLF	9.0	−8 LUFS
Rock MX	GREGG HD MX Med XLF	9.0	−7 LUFS
Smooth Jazz MX LIGHT	GREGG HD MX Light	7.0	−11 LUFS
Smooth Jazz MX OPEN	GREGG HD MX Open XLF	9.0	-10 LUFS
Soul MX	GREGG HD MX Heavy XLF	10.0	−7 LUFS
Soul MX XXLF	GREGG HD MX XXLF	9.0	−8 LUFS
SPORTS MX	SPORTS MX	7.0	−10 LUFS
SPORTS	SPORTS	7.0	− 10 LUFS
Standards MX LIGHT	POP MX LIGHT FLAT	7.0	−11 LUFS
Standards MX OPEN	GREGG HD MX Open XLF	9.0	−10 LUFS
Urban	GREGG HD MX XXLF	9.0	−8 LUFS

CLASSICAL

As their names imply, the CLASSICAL 5-BAND and CLASSICAL 5-BAND+AGC presets (in MX and non-MX versions) are optimized for classical music, gracefully handling recordings with very wide dynamic range and sudden shifts in dynamics. They use heavy inter-band coupling to prevent large amounts of automatic re-equalization, which could otherwise cause unnatural stridency and brightness in strings and horns and which could pump up very low Frequency rumble in live recording venues.

The Five-Band preset disables the AGC, using only the five-band compressor for gain reduction. It also disables phase rotation and uses the phase-linear five-band crossover to ensure the most transparent Five-Band sound available.

CLASSICAL-5B+AGC uses the AGC set for 2:1 compression ratio. Because of the AGC, it affects more of the total dynamic range of the recording than does the CLASSICAL-5 BAND preset. However, the AGC provides extremely smooth and unobtrusive compression because of the gentle ratio and window gating. This preset uses the Five-Band compressor very lightly with a fast release time as a peak limiter. The AGC does almost all of the compression.

CLASSICAL-5B disables the AGC and exploits RSAP’s soft-knee five-band compression with an initial compression ratio of 1:1. Quiet material is gently compressed with a very low compression ratio. The compression ratio increases as the source material gets louder (see Figure 3-16 on page 3-70). Very quiet material is typically amplified by 10dB. This level-dependent compression ratio provides very smooth, subtle compression.

CLASSICAL-5B+AGC PARALLEL and CLASSICAL-5 BAND PARALLEL use the compressor’s parallel mode to bring up low-level passages while preserving more and more of the music’s dynamics as it gets louder (see Parallel Compression). These presets can be very subtle and musical while still allowing the quiet parts of the music to be heard clearly in noisy environments.

For these presets, Less-More sets the amount of amplification of quiet parts of the program. While neither Parallel preset uses the MX Limiter (because the low target loudness typically used in classical operations usually causes little or no peak limiting), you can enable the MX Limiter without losing Less-More functionality.

Because these presets pass the dynamics of loud parts of the music through essentially unmodified, you must use care not to overDrive the peak limiter. These presets are safest to use with low target loudness (–16 LUFS and below), where overdriving the peak limiters is less likely. Because of this level sensitivity, we expect that most users will want to customize the Parallel presets by adjusting the Compression Thresh Offset setting and possibly the Target Loudness and/or MB Final Limit Drive settings.

CLASSICAL-5B+AGC PARALLEL has the AGC turned on to prevent the peak limiters from being grossly overDriven if the input is unusually loud. However, because of the AGC’s slow attack time and 2:1 compression ratio, loud transients can set through CLASSICAL-5 BAND PARALLEL has the AGC turned off, so operators must do appropriate gain riding to prevent annoying loudness inconsistencies between different recordings. In facilities where the program is played out via an automation system, it may be sufficient to statically normalize the gain of individual elements so that the BS.1770 Integrated Loudness of the entire element meets a target value, typically –16 to –20 LUFS. With classical music formats, the target loudness of speech material should be 3 to 6 LU below the target loudness of music.

Because the CLASSICAL presets preserve a significant amount of the dynamic range present in the source material (including speech), it is wise to use a separate microphone processor to ensure appropriate voice/music balance.

GOLD [MX]

GOLD and GOLD MX are loud and “hi-fi”-sounding while still respecting the limitations and basic flavor of the recordings from the era of the 1950s through 1970s. It is the only RSAP preset that uses a considerable amount of fixed equalization to achieve a bright, punchy, “present” sound without over-exaggerating high frequencies. The highs in recordings of this era are often noisy, distorted, or have other technical problems that make them unpleasant sounding when the processor over-equalizes them in an attempt to emulate the high Frequency balance of recently recorded material.

Because they provide an example of using fixed parametric equalization, these presets can be useful as a guide for further customization. (Most of the RSAP radio-style presets do not use fixed equalization other than shelving bass boost.)

GREGG

The GREGG family of presets was designed by Greg Ogonowski. Most use a 200Hz band1/band2 crossover Frequency to achieve a bass sound similar to the classic five-band Gregg Labs FM processors. Dynamically, most of these presets produce a considerable increase in bass energy below 100 Hz and a decrease of bass energy centered at 160 Hz. This bass sound works particularly well with speakers having good bass response.

The GREGG HD MX # presets include peak limiter technology to be loud without the usual peak limiter artifacts. There are also several non-MX presets, which can be used when the target loudness is lower and it is important to maximize the number of instances of RSAP that can run on a given computer; these presets cut CPU usage approximately in half compared to the MX presets.

Most of the GREGG presets are designed to achieve extremely punchy bass, even with program material that lacks bass. GREGG presets use the subharmonic synthesizer in conjunction with bass equalization to achieve this. The processing is tuned dynamically (using the bass limiter and Band 1 of the multi-band compressor) to prevent bass from becoming excessive if the program material already has sufficient bass.

As is always true with presets, the only way to judge them and decide if they are appropriate for your goals is to audition them with program material typical of your format.

GREGG HD and GREGG OPEN HD are good general-purpose presets for popular music programming if you wish to save CPU cycles by not using the MX Limiter. The OPEN version uses a slower multi-band release time and increases density less.

Sonically, GREGG HD and GREGG OPEN HD are not competitive with the GREGG HD MX # presets unless the target loudness is -16 LUFS or lower (above that, the MX presets will sound brighter, more detailed, and punchier). However, GREGG HD and GREGG OPEN HD will not produce overtly objectionable artifacts (such as audible gain pumping) unless the target loudness is -8 LUFS or higher.

GREGG HD MX HEAVY, MEDIUM, and LIGHT are good general-purpose presets for popular music programming if you want an excellent trade-off between loudness and artifacts. As you go from LIGHT to HEAVY, RSAP will produce more and more density and source-to-source consistency. GREGG HD MX MEDIUM is the factory default preset in a new RSAP installation.

GREGG HD MX OPEN uses a SLOW multi-band release to achieve a combination of open, relaxed sound and good source-to-source consistency.

GREGG HD MX XCESS is useful if you want a very consistent, high-energy presentation.

GREGG HD MX XXLF goes for extreme bass punch and impact for formats like Urban and Dance, and was originally created for European customers seeking an “over the top” bass texture.

GREGG HD MXH Heavy XLF uses HARD MX clipping to achieve extremely high loudness. With turned-down down the LIMITER ATTACK settings, It is purposely designed to exhibit some compressor pumping, giving it a more “processed” sound than the other GREGG presets.

GREGG HD MX XXXLF is similar to GREGG HD MX XXLF. This preset generates a lot of synthetic bass energy and is targeted towards “Gold” and “Oldies” formats where the original program material lacks bass impact.

GREGG HD MXH Xcess XXLF was derived from GREGG HD MX Xcess, but uses HARD MX Limiting and, like GREGG HD MX XXXLF, generates generous amounts of synthetic bass and is optimized for “Gold” and “Oldies” formats.

NEWS-TALK

This preset (in both MX and non-MX versions) is based on the fast multi-band release time setting so it can quickly perform automatic equalization of substandard program material, including telephone.

It is useful for creating a uniform, intelligible sound from widely varying source material, particularly source material that is “hot from the field” with quality.

The MX Limiter has little value for speech programming, as the non-MX look-ahead limiter is very clean on speech. NEWS-TALK MX exists mostly for the convenience of users having mixed formats, streaming music for part of the day and news or talk at other times. It allows you to use an MX preset for the music programming and switch between the music and news-talk presets without audible glitches caused by a sudden change in input/output delay when enabling the MX Limiter. The MX Limiter adds about 240ms of delay when active.

POP MX

The POP MX family of presets offer a different bass texture compared to the GREGG presets. In general, they do not push low bass as hard, and may therefore be preferred to the GREGG presets if most of the audience is listening on player devices with small loudspeakers having no significant response below 100Hz or so.

POP+SUBHARM MX is appropriate for programming that mixes recordings from different eras. It uses the subharmonic synthesizer to add low bass to material that lacks it. As such, it is the POP preset most appropriate for player devices with good low-Frequency response.

POP MX FLAT EQ, POP MX LIGHT FLAT, and POP MX OPEN FLAT use no bass boost or subharmonic synthesis, although they still exploit the ability of the multi-band compressor to automatically re-equalize material that lacks bass. They use a combination of the multi-band compressor and HF Enhancer to re-equalize dull-sounding material automatically. The LIGHT version uses SLOW multi-band release time to minimize density build-up caused by the multi-band compressor. OPEN is very similar to LIGHT, except that OPEN uses the multi-band compressor's LINEAR-PHASE crossover mode, whereas LIGHT uses ALLPASS These presets may be preferred if the musical style of the program material calls for modest amounts of bass. Examples include traditional country and smooth jazz.

SPORTS

Similar to NEWS-TALK except the AGC Release (AGC Release Time) is slower and the Gate Thresh (Gate Threshold) is higher. This recognizes that most sports programming has very low signal-to-noise ratio due to crowd noise and other on-field sounds, so the preset does not pump this up as the NEWS-TALK preset would tend to do.

Sound for Picture Presets

✶ A/V Sync input/output delay varies from about 450 to 1000 ms, depending on which features re enabled. Each time the RSAP Service is stopped and restarted, the delay may vary by as much as 25ms due to use of the Windows MME sound model for the input device, and this will make it difficult to achieve consistent A/V sync.

The sound-for-picture presets have GLOBAL Target Loudness. They use non-MX Limiter mode because little or no peak limiting is required for typical target loudness used in video applications, so it is pointless to waste CPU cycles on MX Limiting. Most of these presets have the CBS Loudness Controller turned on. It is set so that loudness is constrained to a level consistent with the active Target Loudness value. The Target Loudness value is the same as the "target loudness" as measured by an ITU-R BS.1770-2 (or higher) Integrated loudness meter (with a 10-second rolling windows of integration) in units of LkFS (***ATSC Rec. A/85***) and LUFS (***EBU Rec. R128***). To comply with these standards, Target Loudness is customarily set to -24 in the United States and -23 in the EU.

To constrain the reading of a BS.1770 loudness meter (included with RSAP) to a fixed threshold, almost all "TV" presets have the BS.1770 Threshold setting set to 0 LK. However, if your organization does not have a strict policy about processing for the BS.1770 meter, we recommend that you edit your preferred preset by setting this setting anywhere from +2 to OFF and then saving the result as a User Preset. For a more detailed discussion, see BS.1770 Safety Limiter.

✶ If you wish to use a given preset with the CBS loudness setting disabled, set the Loudness Threshold setting (in the Multiband tab) to OFF and save the result as a User Preset.

Table 14.3 Sound for Pictures Presets

Preset Name	Source Preset	Less-More	Loudness
TV 5B DRAMA COUPLED	TV 5B DRAMA COUPLED	5.0	GLOBAL
TV 5B DRAMA	TV 5B DRAMA	5.0	GLOBAL
TV 5B GEN PUR W NR	TV 5B GEN PUR W NR	5.0	GLOBAL
TV 5B GEN PURP NOLC	TV 5B GEN PURP NOLC	5.0	GLOBAL
TV 5B GEN PURPOSE	TV 5B GEN PURPOSE	5.0	GLOBAL
TV 5B NEWS	TV 5B NEWS	5.0	GLOBAL
TV 5B OPTICAL FILM	TV 5B OPTICAL FILM	5.0	GLOBAL
TV 5B SPORTS	TV 5B SPORTS	5.0	GLOBAL
TV AGC+LC	TV AGC+LC	5.0	GLOBAL
TV AGC+LC+DS	TV AGC+LC+DS	5.0	GLOBAL

TV 5B-GEN PUR W/NR (TV Five-Band General Purpose with Noise Reduction)

This preset provides effective dynamic range setting and “automatic re-equalization” of most dramatic material. It uses the Loudness Controller to setting loudness tightly. It applies single-ended noise reduction to the material, which will reduce unwanted noise like hiss, hum, or stage rumble. However, it will also reduce ambiance. If the program material is carefully produced (as are most contemporary feature-film soundtracks), you may wish to use TV 5B-GEN PURPOSE (which does not apply noise reduction), or, if the material is so well produced that it would not benefit from “automatic re-equalization,” use TV 2B-GEN PURPOSE.

TV 5B-GEN PURPOSE (TV Five-Band General Purpose without Noise Reduction)

This preset is identical to TV 5B-GEN PUR W/NR except that the single-ended dynamic noise reduction system is off.

TV 5B GEN PURP -LC (TV Five-Band General Purpose without Loudness Controller)

This preset is the same as TV 5B-GEN PURPOSE except the Loudness Controller is turned off and the MB Drive setting is backed off by 3dB to achieve approximately the same loudness as TV 5B-GEN PURPOSE. Because the Loudness Controller is inactive, this preset has more dynamic punch than TV 5B-GEN PURPOSE. The five-band processing makes the audio spectrum more consistent than does TV 2B-GEN PUR-LC, so TV 5B-GEN PURP -LC settings loudness better than TV 2B-GEN PURP -LC even though the Loudness Controller is inactive.

TV 5B-DRAMA (TV Five-Band Drama)

This preset uses the RSAP’s soft knee compression and AGC RATIO setting to regulate loudness while still preserving some of the dynamic range of the original mix. In addition, the five-band compressor automatically re-equalizes material that may otherwise sound spectrally unbalanced. The center channel compressor effectively desses dialog without punching holes in the remaining channels. Coupling between the center channel and remaining channels allows the center channel’s level to be boosted automatically by as much as 3 dB with respect to the other channels if this is needed to help intelligibility.

You may prefer this preset to TV 2B-DRAMA because of its effective, unobtrusive de-essing and because it is more resistant to spectral gain inter-modulation, which is program material in one Frequency range’s audibly pumping material in a different range.

Because it preserves some dynamic range, it is important that the input level to the RSAP not be too far awry. Usually, network feeds will meet this requirement but local play-out of older material that has not been checked for loudness by a long-term Loudness Level meter like ITU-R BS.1770 may not work well. Use one of the general purpose presets for this kind of material. This preset fully exploits the Loudness Controller, which usually shows slight gain reduction with dialog at normal levels.

TV 5B-DRAMA COUPLD (TV Five-Band Drama Coupled)

This preset is similar to TV 5B DRAMA but uses more interband band coupling in the five-band compressor and a slower multi-band compressor release time so it performs less automatic re-equalization of the program material.

TV 5B-NEWS (TV Five-Band News)

This preset rides gain more quickly than the general-purpose presets. Its AGC release time is faster, so it will bring up low-level material more quickly. It is designed for live news programs where input levels may be quite unpredictable. It also automatically re-equalizes substandard audio (which is quite common in live news broadcasts). The dynamic single-ended noise reduction is turned on.

TV 5B-SPORTS (TV Five-Band Sports)

This preset is similar to TV 5B-NEWS, except the AGC release time is slower to resist pumping up crowd noise.

TV 5B OPTICAL FILM (TV Five-Band Optical Film)

This preset is designed to make the best of the low-quality audio provided with optical film sound tracks (particularly 16mm). The gate threshold is quite high to avoid pumping up hiss, thumps, and other optical artifacts. The threshold of the single-ended dynamic noise reduction system is also high so that this system can reduce artifacts as far as possible. Release times are slow, because we assumed that material encoded on optical film has already been carefully level-controlled to accommodate the very limited dynamic range of the medium, so little gain riding is therefore required from the RSAP.

Using the Test Modes

The System Config > Test tab allows you to switch between OPERATE, BYPASS, and TONE. When you switch to BYPASS or TONE, RSAP saves the preset you had on-air and will restore it when you switch back to OPERATE. Even if you had been editing a preset and did not yet save these changes as a User preset, you will not lose the edits you made. The available Tone waveforms are SINE, SQUARE, and PINK NOISE.

The pink noise spectrum is accurate to ±0.05dB, 20-20,000Hz.

“Squarewaves” are available up to 1kHz.

Your RSAP is a band-limited system (like any digital system). Because of the Gibbs phenomenon, no band-limited system can produce true squarewaves without overshoot. Instead, your RSAP generates useful squarewave-like waveforms without overshoot by applying a waveform whose sample values periodically switch between +1 and –1 to a non-overshooting pulse-shaping low-pass filter. This eliminates overshoot at the cost of increasing the rise time of the “square-wave” edges.

The main purpose of these waveforms is to test the transmission system following the RSAP to determine if the path introduces overshoot, tilt, or ringing into the “square-wave.” This is particularly important if you are using an analog signal path after the RSAP because the –3dB Frequency of the path must be 0.15Hz or lower to prevent 50 Hz squarewaves from overshooting more than 1% and increasing peak levels.

To prevent aliasing and ensure that only odd-order harmonics are generated, the source waveform is generated at 192kHz sample rate and has an identical, integer number of samples in its “+1” and “–1” segments. Hence, output frequencies that are not sub-multiples of 96kHz (like 315Hz) will differ from their labeled values by a few percent. This causes no problems when testing transmission systems.

Troubleshooting

This section provides some basic troubleshooting methods.

Hard MX Overshoot mode cannot be enabled

The HARD MX Overshoot mode is only available when the pre-emphasis is set to FLAT in the STEREO OUTPUT.

HARD MX is a very CPU-intensive process and will cause audio stuttering if the host computer is insufficiently powerful. To reduce CPU load, try disabling Optimix and the CBS Loudness Controller in the Utilities tab.

✶ If the output sound device is stereo, Optimix will be disabled automatically.

Meters freeze momentarily but audio continues to be processed normally

This is by design. The software thread settingling the meters is given lower than “normal” priority in Windows to prevent the meters from interrupting important threads that maintain audio continuity. Quickly changing setting settings will sometimes temporarily cause the meters to freeze because the setting setting messages have a higher priority than the meters.

Poor peak level setting

Thanks to its “true-peak” limiter, which anticipates and settings peak levels following an ideal reconstruction filter in the analog domain, RSAP audio processing usually settings its output peak levels to an accuracy of 0.2 dB at any output sample rate—in principle, sample rate conversion is similar to reconstruction.

However, codecs like HE-AAC have intrinsic peak overshoots and you must allow headroom for these to prevent audible clipping in player devices. 1.5dB is typically sufficient, as the remaining clipping will have a low duty cycle and is unlikely to be audible. The RSAP Output Level setting directly settings headroom without affecting loudness. For example, to allow 1.5dB of headroom for codec overshoots, set this setting to -1.5dB.

An analog connection can cause analog-domain overshoot if the connection is not phase linear and has a low-Frequency cutoff of greater than 0.15Hz (at -3dB).

Audible distortion

Make sure that the problem can be observed on more than one sound system and at several locations.

Verify that the source material at the RSAP audio inputs is clean. Heavy processing can exaggerate even slightly distorted material, pushing it over the edge into unacceptability.

The subjective adjustments available to the user have enough range to cause audible distortion at their extreme settings. Advancing the MB Final Limit Drive setting too far will inevitably cause distortion. (Distortion is very probable if gain reduction in the final limiter frequently exceeds 8 dB.) Setting the Less-More setting beyond “9” will cause audible distortion of some program material.

The MX peak limiter has several settings that affect the trade-off between bass punch and IM distortion between the bass and midrange. Try turning up Bass Pre-limiting and Bass Limiting (move them more towards to right of the tab). Setting Bass Pre-limit Mode to HARD and setting the MX Limiter Threshold setting slightly below “0” can also help.

Audible noise in processed audio

Excessive compression will always exaggerate noise in the source material.

RSAP reduces this problem with its compressor gate, which freezes the gain of the AGC and compressor systems whenever the input noise drops below a level set by the Gate Threshold setting, preventing noise below this level from being further increased.

There are two independent silence gates in each processing channel of the RSAP. The first affects the AGC and the second affects the Multiband Compressor. Each has its own threshold setting.

In sound for picture, the setting of the Gate Threshold setting is quite critical if you want the processing to be undetectable to the audience. If this setting is set too low, then the RSAP will pump up quiet sounds like ambiance and underscoring to unnaturally high levels..

If you are using the RSAP with a sound-card having an analog input, the overall noise performance of the system is usually limited by the overload-to-noise ratio of the sound-card’s analog-to-digital converter.

In digital radio applications, if an analog studio-to-transmitter link (STL) is used to pass unprocessed audio to RSAP, the STL noise level can severely limit the overall noise performance of the system because compression in RSAP can exaggerate the STL noise. For example, the overload-to-noise ratio of a typical analog microwave STL may only be 70-75dB. In this case, it is wise to use another RSAP to perform the AGC function prior to the STL transmitter and to setting the STL's peak modulation. This will optimize the signal-to-noise ratio of the entire transmission system. An uncompressed digital STL will perform much better than any analog STL.

Shrill, harsh sound; excessive sibilance

Excessively high settings of the HF GAIN setting can cause this problem. It can also be caused by excessively high settings of the B5 Threshold (Band 5 Compression Threshold) setting. In the latter case, you are first likely to notice the problem as harsh sibilance on voice.

Gain pumping when high Frequency energy is present

This will occur with many non-MX factory presets when the pre-emphasis of the RSAP is set to 50µs or 75µs.

The gain pumping happens because the pre-emphasis creates a large high Frequency boost before the look-ahead limiter, so the look-ahead limiter must produce large amounts of gain reduction to setting peak levels.

To correct this problem, enable the MX Limiter (in the Distortion tab), and, if necessary, turn down the FINAL Limiter Drive setting. Note that enabling the MX Limiter may cause CPU usage to increase.

Compared to the look-ahead limiter that performs peak limiting when the MX Limiter is disabled, the MX Limiter is much less susceptible to gain pumping.

System will not pass line-up tones at full output level/100% modulation

This is normal in OPERATE mode. Sine waves have a very low peak-to-average ratio by comparison to program material. The processing thus automatically reduces their peak level to bring their average level close to that of program material, promoting a more consistent and well-balanced sound quality.

To pass line-up tones transparently, use the Bypass Mode with the Bypass Gain set to 0dB.

PASS-THROUGH mode will pass tones very close to 100%, but the peak limiter of the RSAP remains in the signal path.

Speech/music detector toggles continuously between Speech and Music with a low-Frequency test tone

This can occur when the main high-pass filter and speech high-pass filters are set differently. If the test Frequency is midway between the two filter frequencies, switching between them will cause the tone level to change abruptly, fooling the speech/music detector into thinking that this is a speech waveform. This issue never arises with program material.

Dialog is muffled in TV applications

Set the B3>B4 Coupling and B4>B5 Coupling settings to a higher value. This will allow the processing to apply more dynamic high Frequency boost. In the surround processing, adjust these settings in the center channel compressor.

Set the Center (surround) or Speech Mode (stereo) B4 Compressor Threshold setting to a higher value (i.e., closer to 0 dB). This will produce less gain reduction in the presence region.

Adjust the Equalizer > HF Enhance setting as required. For most TV programming, moderate settings (like 3dB) are appropriate because they minimize the possibility of increasing noise.

General dissatisfaction with subjective sound quality

RSAP is a complex processor that can be adjusted for many different tastes. For most users, the factory presets, as augmented by the gamut offered by the Less-More setting for each preset, are sufficient to find a satisfactory “sound.” However, some users will not be satisfied until they have accessed other Modify Processing settings and have adjusted the subjective setup settings in detail to their satisfaction.

Automatic Gain Controls

This chapter outlines how to use the RSAP as a studio Automatic Gain Control AGC (including peak limiting) to protect a studio-to-transmitter link (STL), optimally using the STL’s native dynamic range.

Overview

In radio applications, it is common to have an external Automatic Gain Control (AGC) at the studio side of a studio-to-transmitter link to protect the link from overload. Most of the processing structures in RSAP setting level with a preliminary AGC internal to RSAP. If you are using an external AGC device (such as another RSAP) in front of the RSAP, set the first RSAP’s internal AGC to DISABLED. This is to ensure that the internal and external AGCs do not conflict with each other and that they do not simultaneously increase gain (resulting in increased noise).

The AGC smoothly rides gain before the multi-band compressor. A linear-phase crossover is available and the left and right channels are always stereo-coupled using RMS summation.

Figure 15.1 Example of the AGC Controls in DashBoard

✶ Disable the AGC when you want to create a preset with minimal processing (such as a CLASSICAL preset). The AGC is also disabled when you are using a studio level controller to protect a transmission link before the RSAP. However, in this case it is recommended to disable the AGC via the System Config > Global tab.

Master Delta Threshold

AGC Maximum Delta GR approximates the maximum gain difference permitted between the two stereo channels of the AGC (Stereo processing only). Set it to “0” for perfect stereo coupling. In surround processing, the RMS sum of all audio channels settings the gain reduction of the AGC, which is identical in all channels.

Drive

AGC Drive setting adjusts signal level going into the slow dual-band AGC, therefore determining the amount of gain reduction in the AGC. This setting also adjusts the “idle gain”—the amount of gain reduction in the AGC section when the structure is gated. (It gates whenever the input level to the structure is below the threshold of gating.)

The total amount of gain reduction in the Five-Band structure is the sum of the gain reduction in the AGC and the gain reduction in the five-band compressor. The total system gain reduction determines how much the loudness of quiet passages will be increased (and, therefore, how consistent overall loudness will be). It is determined by the setting of the AGC Drive setting, by the level at which the console VU meter or PPM is peaked, and by the setting of the Multiband Drive (compressor) setting.

Idle GR

The "idle gain" is the target gain of the AGC when the silence gate is active. Whenever the silence gate turns on, the gain of the AGC slowly moves towards the idle gain.

The idle gain is primarily determined by the AGC Drive setting—a setting of 10 dB will ordinarily produce an idle gain of -10dB (i.e., 10dB of gain reduction). However, sometimes you may not want the idle gain to be the same as the AGC Drive setting. The AGC Idle Gain setting allows you to add or subtract gain from the idle gain setting determined by the AGC Drive setting.

You might want to do this if you make a custom preset that otherwise causes the gain to increase or decrease unnaturally when the AGC is gated. For example, to make the idle gain track the setting of the AGC Drive setting, set the AGC Idle Gain setting to zero. To make the idle gain 2dB lower than the setting of the AGC Drive setting, set the AGC Idle Gain setting to -2.

Gate Threshold

This setting determines the lowest input level that will be recognized as program by the RSAP; lower levels are considered to be noise or background sounds and cause the AGC or multi-band compressor to gate, effectively freezing gain to prevent noise breathing.

In sound for picture, the setting of the gate threshold settings are quite critical if you want the processing to be undetectable to the audience. If this setting is set too low, then the RSAP will pump up quiet sounds such as ambiance and underscoring to unnaturally high levels.

There are two independent silence-gating circuits in RSAP. The first affects the AGC and the second affects the multi-band compressor (two-band or five-band). Each has its own threshold setting.

In the five-band structure, the multi-band silence gate causes the gain reduction in bands 2 and 3 of the five-band compressor to move quickly to the average gain reduction occurring in those bands when the gate first turns on. This prevents obvious midrange coloration under gated conditions, because bands 2 and 3 have the same gain.

The multi-band gate also independently freezes the gain of the two highest Frequency bands (forcing the gain of the highest Frequency band to be identical to its lower neighbor), and independently sets the gain of the lowest Frequency band according to the setting of the DJ BASS boost setting (in the Equalization tab). Thus, without introducing obvious coloration, the gating smoothly preserves the average overall Frequency response "tilt" of the multi-band compressor, broadly maintaining the "automatic equalization" curve it generates for a given piece of program material.

✶ If the MB GATE THR (Gate Threshold) is set to OFF, the DJ BASS setting is disabled.

Release

The AGC Release ("AGC Master Release") setting provides an adjustable range from 0.5 dB/second (slow) to 20 dB/second (fast). The increase in density caused by setting the AGC Release setting to fast settings sounds different from the increase in density caused by setting the Five-band's Multiband Release setting to FAST. You can trade the two off to produce different effects.

Unless it is purposely increased (with the AGC Release setting), the Automatic Gain setting (AGC) that occurs in the AGC prior to the multi-band compressor makes audio levels more consistent without significantly altering texture. Then the multi-band compression audibly changes the density of the sound and dynamically re-equalizes it as necessary: booming bass is tightened; weak, thin bass is brought up; highs are always present and consistent in level.

The various combinations of AGC and compression offer great flexibility:

- Light AGC + light compression yields a wide sense of dynamics, with a small amount of automatic re-equalization.
- Moderate AGC + light compression produces an open, natural quality with automatic re-equalization and increased consistency of Frequency balance.
- Moderate AGC + moderate compression gives a denser sound, particularly as the release time of the multi-band compressor is sped up.
- Moderate AGC + heavy compression (particularly with a FAST multi-band release time) results in a “wall of sound” effect, which may cause listener fatigue.

Adjust the AGC (with the AGC Drive setting) to produce the desired amount of AGC action, and then fine-tune the compression and clipping with the Five-Band structure’s settings.

Bass Threshold

This feature determines the compression threshold of the bass band in the AGC. It can be used to set the target spectral balance of the AGC.

As the AGC B CPL setting is moved towards “100%,” the AGC BASS Threshold setting affects the sound less and less.

The interaction between the AGC BASS Threshold setting and the AGC B CPL setting is a bit complex, so we recommend leaving the AGC BASS Threshold setting at its factory setting unless you have a good reason for readjusting it.

Bass Attack

AGC Bass Attack sets the attack time of the AGC bass compressor (below 200Hz).

Bass Release

AGC Bass Release sets the release time of the AGC bass compressor.

Master Attack

AGC Master Attack sets the attack time of the AGC master compressor (above 200Hz).

Window Threshold Window Release

This feature determines the size of the floating “slow zone” window in the master band of the AGC. (The Bass band is not windowed.)

The window works by slowing down changes in the AGC gain reduction that are smaller than the Window Size. The window has 2:1 asymmetry around the current AGC gain reduction. For example, if the Window Size is set to 4dB, the window extends 4dB in the release direction and 2dB in the attack direction.

✶ The default setting for the Window Size is 3dB.

If the AGC needs to respond to a large change in its input level by making a gain change that is larger than the window, then the AGC’s attack and release settings determine the AGC’s response time. However, if the change in input level is smaller than the Window Size, the Window Release setting determines the attack and release times. This is usually much slower than the normal AGC time constants. This prevents the AGC from building up density in material whose level is already well established.

The previous explanation was somewhat simplified. In fact, the window has “soft edges.” Instead of switching abruptly between time constants, the attack and release times morph smoothly between the setting of the Window Release setting and the setting of the AGC master release and attack settings.

Ratio

This feature determines the compression ratio of the AGC. The compression ratio is the ratio between the change in input level and the resulting change in output level, both measured in units of dB.

RSAP compressor can be operated at a compression ratio as low as 2:1. This can add a sense of dynamic range and is mostly useful for subtle fine arts formats like classical and jazz.

Bass Coupling

This setting clamps the amount of dynamic bass boost (in units of dB) that the AGC can provide.

The AGC processes audio in a master band for all audio above approximately 200Hz and a bass band for audio below approximately 200Hz. The AGC Master and Bass compressor side-chains operate without internal coupling. The gain reduction in the BASS audio path is either the output of the Bass compressor side-chain or the output of the Master band side-chain. The AGC BASS Coupling setting sets the switching threshold. For example, if the AGC BASS Coupling setting is set to 4dB and the master gain reduction is 10dB, the bass gain reduction cannot decrease below 6dB even if the gain reduction signal from the Bass compressor side-chain is lower. However, the audio path bass gain reduction can be larger than the master gain reduction without limit. In the previous example, the bass gain reduction could be 25dB.

The default setting of the AGC BASS Coupling setting is 0 B, which allows the AGC bass band to correct excessive bass as necessary but does not permit it to provide a dynamic bass boost.

Stereo Controls

This chapter outlines the stereo enhancer and synthesizer features of the RSAP.

Stereo Enhancer Controls

The stereo enhancer increases the energy in the stereo difference signal (L-R) whenever a transient is detected in the stereo sum signal (L+R). This increases the sense of envelopment and space while maintaining excellent downmix compatibility.

Figure 16.1 Example of the Stereo Controls in DashBoard

The stereo enhancer's gating operates under two conditions:

- The two stereo channels are close to identical in magnitude and phase. In this case, the enhancer assumes that the program material is actually mono and thus suppresses enhancement to avoid creating undesired channel imbalance.
- The ratio of L-R / L+R of the enhanced signal tries to exceed the threshold set by the L-R / L+R Ratio Limit setting. In this case, the enhancer prevents further enhancement in order to prevent excess L-R energy, which can sound unnatural and which can increase multi-path distortion in FM broadcasting.

The Stereo Enhancer meter indicates the amount of gain (in dB) that applied to the L-R signal.

✶ It is not recommended to use stereo enhancement with low bit-rate codecs. At low bit-rates, these codecs use various parametric techniques for encoding the spatial attributes of the sound field. Stereo enhancement can unnecessarily stress this encoding process.

The stereo enhancer has the following settings:

- Amount sets the maximum spatial enhancement.
- Enhancer In / Out bypasses the stereo enhancer. OUT is equivalent to setting the Amount to 0.
- L-R / L+R Ratio Limit sets the maximum amount of enhancement to prevent multi-path distortion. However, if the original program material exceeds this limit with no enhancement, the enhancer will not reduce it.

Stereo Synthesizer Controls

The stereo synthesizer creates an artificial stereo difference signal (L-R) by passing the mono input through a multistage all-pass filter. After matrixing with the original mono input (which is the L+R signal) to produce the synthesized left and right channels, the result is a "complementary comb filter" whose notches are spaced in Frequency in an approximately logarithmic manner.

Table 16.1 Stereo Synthesizer Controls

Name	Range
Mono > Stereo Algorithm	Bypass, Auto, Upmix
In / Out Wide	Narrow
Mono > Stereo Separation	0 to 10

In AUTOMATIC mode, synthesis from audio on the left input channel will occur if silence is detected on the right input channel. A 187 ms look-ahead delay in the audio path compensates for the delay built into the detector to prevent false triggering on extremely brief right channel pauses. Silence gating prevents triggering unless there is activity on the left channel.

The stereo synthesizer has the following settings.

Mode

This setting allows you to manually bypass the synthesizer or force it to always upmix from material on the left channel. AUTO forces up-mixing when silence is detected on the right input channel, as described above.

Stereo Algorithm

This specifies the number of notches in the complementary comb filters. WIDE produces fewer notches than NARROW. WIDE produces the most dramatic effect on music, while NARROW prevent speech from becoming unnaturally wide-sounding, and is preferable in sound-for-picture applications.

Separation

This sets the gain applied to the allpass filter chain. 0 suppresses synthesis, while 10 causes the magnitudes of the L+R and L-R signals to be identical.

✶ Assuming equal gains in the L+R and L-R channels, the mathematics of the process require the phase difference of the L and R channels to be 90 degrees at the frequencies where their magnitudes are equal. This can produce an effect that some people dislike. If the output of the synthesizer is applied to RSAP’s phase correction algorithm, this will remove the “phasiness” at higher frequencies at the expense of putting bumps of up to 3 dB in the Frequency response of the mono sum (assuming that Stereo Separation is set to 10). Because of this compatibility issue, we recommend instead simply turning down the Stereo Separation setting until the result sounds comfortable and convincing. For example, selecting 7 where the L-R signal is 70% of the L+R signal is a good compromise value.

Equalizer Controls

This chapter summarizes the available equalization settings of the audio processor.

Overview

Any equalization that you set will be automatically stored in any User Preset that you create and save. For example, you can use a User Preset to combine an unmodified Factory Programming Preset with your custom equalization. Of course, you can also modify the Factory Preset (with Basic or Advanced Modify) before you create your User Preset.

In general, you should be conservative when equalizing modern, well-recorded program material. This is particularly true with general-purpose video programming.

✶ Except for BASS GAIN, most of the factory presets use less than 3dB of equalization.

Figure 17.1 Example of the Equalizer (EQ) Controls in DashBoard

Low Band Equalizer

This setting is a specially designed equalizer whose boost and cut curves closely emulate those of a classic analog parametric equalizer with conventional bell-shaped curves (within 0.15dB worst-case). This provides warm, smooth, “analog-sounding” equalization.

LF Freq determines the center Frequency of the equalization, in Hertz. The range is 20-500Hz.

LF Gain determines the amount of peak boost or cut (in dB) over a 10 dB range.

Width determines the bandwidth of the equalization, in octaves. The range is 0.8-4.0 octaves. If you are unfamiliar with using a parametric equalizer, 1.5 octaves is a good starting point. These curves are relatively broad because they are designed to provide overall tonal coloration, rather than to notch out small areas of the spectrum.

The LF parametric can be used in the mid-bass region (100-300Hz) to add “warmth” and “mellowness” to the sound when boosting. When cutting, it can remove a “woody” or “boxy” sound.

The equalizer has constant “Q” curves. This means that the cut curves are narrower than the boost curves. The width (in octaves) is calibrated with reference to 10dB boost. As you decrease the amount of EQ gain (or start to cut), the width in octaves will decrease. However, the “Q” will stay constant. “Q” is a mathematical parameter that relates to how fast ringing damps out. (Technically, we are referring to the "Q" of the poles of the equalizer transfer function, which does not change as you adjust the amount of boost or cut.)

The curves in RSAP's equalizer were created by a so-called "minimax" ("minimize the maximum error" or "equal-ripple") IIR digital approximation to the curves provided by an analog parametric equalizer. Therefore, unlike less sophisticated digital equalizers that use the "bilinear transformation" to generate EQ curves, the shapes of RSAP's curves are not distorted at high frequencies.

Mid Band Equalizer

Midrange Parametric Equalizer is a parametric equalizer whose boost and cut curves closely emulate those of an analog parametric equalizer with conventional bell-shaped curves.

Mid Freq determines the center Frequency of the equalization, in Hertz. The range is 250-6000Hz.

Mid Gain determines the amount of peak boost or cut (in dB) over a 10dB range.

Mid Width determines the bandwidth of the equalization, in octaves. The range is 0.8-4.0 octaves. If you are unfamiliar with using a parametric equalizer, one octave is a good starting point.

With Five-Band presets, the audible effect of the midrange equalizer is closely associated with the amount of gain reduction in the midrange bands. With small amounts of gain reduction, it boosts power in the presence region. This can increase the loudness of such material substantially. As you increase the gain reduction in the midrange bands (by turning the Multiband Drive setting up), the MID GAIN setting will have progressively less audible effect. The compressor for the midrange bands will tend to reduce the effect of the MID Frequency boost (in an attempt to keep the gain constant) to prevent excessive stridency in program material that already has a great deal of presence power. Therefore, with large amounts of gain reduction, the density of the presence region energy will be increased more than will the level of energy in that region.

Because the 3.7kHz band compressor is partially coupled to the gain reduction in the 6.2kHz band in most presets (as set by the B4>5 Coupling setting), tuning MID FREQ to 2-4kHz and turning up the MIDGAIN setting will decrease energy in the 6.2kHz band—you will be increasing the gain reduction in both the 3.7kHz and 6.2kHz bands. You may wish to compensate for this effect by turning up the BRILLIANCE setting.

With Two-Band presets, the midrange equalizer will behave much more as you might expect because the two-band structure cannot automatically re-equalize midrange energy. Instead, increasing midrange energy will moderately increase the Master band's gain reduction.

✶ Use the MID FREQ equalizer with caution. Excessive presence boost tends to be audibly strident and fatiguing. Moreover, the sound quality, although loud, can be irritating. We suggest a maximum of 3dB boost, although 10dB is achievable. In some of our factory music presets, we use a 3dB boost at 2.6kHz to bring vocals more up-front.

High Band Equalizer

This setting is an equalizer whose boost and cut curves closely emulate those of an analog parametric equalizer with conventional bell-shaped curves.

High Freq determines the center Frequency of the equalization, in Hertz. The range is 1-15kHz

High Gain determines the amount of peak boost or cut over a 10dB range.

High Width determines the bandwidth of the equalization, in octaves. The range is 0.8-4.0 octaves. If you are unfamiliar with using a parametric equalizer, one octave is a good starting point.

Excessive high Frequency boost can exaggerate hiss and distortion in program material that is less than perfectly clean. We suggest no more than 4dB boost as a practical maximum, unless source material is primarily from high-quality digital sources. In several of our presets, we use this equalizer to boost the upper presence band (4.4 kHz) slightly, leaving broadband HF boost to the Brilliance and/or HF Enhance settings.

All Band Equalizer

This feature is similar to the other parametric sections but can be swept over the entire audio band, from 20 Hz to 20 kHz.

Bass Gain

This sets the amount of bass boost (dB) at the top of the shelf.

Bass Frequency

This sets the Frequency where shelving starts to take effect.

Bass Slope

This sets the Slope (dB/octave) of the transition between the top and bottom of the shelf.

Because the Five-Band structure often increases the brightness of program material, some bass boost is usually desirable to keep the sound spectrally well balanced. Adjustment of bass equalization must be determined by individual taste and by the requirements of your format. Be sure to listen on a wide variety of consumer systems—it is possible to create severe distortion on poor quality speakers by over-equalizing the bass.

The moderate-Slope (12dB/octave) shelving boost achieves a bass boost that is more audible on smaller radios, but which can sound boomier on high-quality systems. The steep-Slope (18dB/octave) shelving boost creates a solid, punchy bass from the better consumer systems with decent bass response. The 6dB/octave shelving boost is like a conventional tone setting and creates the most mid-bass boost, yielding a “warmer” sound. Because it affects the mid-bass Frequency range, where the ear is more sensitive than it is to very low bass, the 6dB/octave Slope can create more apparent bass level at the cost of bass “punch.”

You must choose the characteristic you want by identifying your target audience and the receivers they are most likely to be using. Regardless of which curve you use, we recommend a +2 to +5dB boost for most formats. Larger amounts of boost will increase the gain reduction in the lowest band of the multi-band compressor, which may have the effect of reducing some frequencies. So be aware the large fixed bass boosts may have a different effect than you expect because of the way that they interact with the multi-band compressor. (The GREGG presets use this effect purposely to create a dynamic cut in the mid-bass.)

Subharmonic

The subharmonic synthesizer generates subharmonics of fundamental frequencies in the 50-90 or 60-120Hz range, as selected by the Sub Harmonic Cutoff Frequency setting. The subharmonics are one octave below the frequencies from which they are generated (i.e., 25 to 45Hz or 30-60Hz) and track the levels of their generating frequencies. Subharmonic Injection can be varied from 0 to 100% of the level of the generating Frequency. A good starting point for popular music formats is 50%.

If input program material below 50 or 60Hz is present, the subharmonic synthesizer automatically reduces the level of the synthesized subharmonics to prevent excess build-up of energy below 50 or 60Hz (i.e. the amount of Subharmonic Injection will be lower than the setting of the Subharmonic Injection setting).

To prevent introducing unnatural coloration in male speech, the subharmonic synthesizer is disabled when the RSAP’s automatic speech/music detector detects speech. When you use the 120 Hz cutoff Frequency (which can cause “thumps” behind low-pitched male speech), it is wise to ensure that raw speech is panned to the exact center, which is one important criterion that the speech/music detector uses to detect speech.

It is important to understand that material below 50 Hz takes up lots of peak level to produce significant loudness. Moreover, the close spacing of psychoacoustic “equal loudness” curves 3 below 50 Hz means small changes in amplitude lead to large change in subharmonic loudness. The unpredictability of receiver bass response means that the perceived loudness of subharmonics is highly receiver-dependent.

Because of the amount of peak level they use up, subharmonics will always make a broadcast sound quieter for a given amount of processing artifacts/distortion—they use up peak level that otherwise could be dedicated to audio to which the ear is more sensitive. (This is an inevitable effect of the equal-loudness curves.) For all these reasons, it is wise not to overdo subharmonic synthesis.

✶ When you apply an L+R sine wave at a Frequency between 50 to 90Hz to the RSAP's input, the RSAP's speech/music detector will detect this as "speech," so no subharmonics will be produced. To test the subharmonic synthesizer with tone, there must be at least 2 B of level difference between the left and right inputs. In other words, be sure that the RSAP's speech/music detector is indicating "music" in the RSAP PC GUI.

Brilliance

The Brilliance sets the Drive to Band 5. The Band 5 compressor/limiter dynamically settings this boost, protecting the final limiter from excessive HF Drive. We recommend a maximum of 4dB of Brilliance boost and most people will prefer substantially less.

HF Enhancer

This feature is a program-adaptive, 6 dB/octave shelving equalizer with a 4kHz turnover Frequency. It constantly monitors the ratio between high Frequency and broadband energy and adjusts the amount of equalization in an attempt to make this ratio constant as the program material changes. It can therefore create a bright, present sound without over-equalizing material that is already bright.

High Frequency Enhancer is a mix setting for the High Frequency Enhancer that sets the amount of enhanced high frequencies being mixed into the input and clamps the maximum enhancement to the setting's setting.

Sensitivity Trim (v0.9.9 and higher) trims the ratio of high-Frequency to wide-band energy produced by HF Enhancer at a given setting of the High Frequency Enhancer setting. Higher settings increase the amount of enhancement. The default is 0dB, which retains the behavior of the previous High Frequency Enhancer. This had only one setting: High Frequency Enhancer.

Threshold Trim trims the High Frequency Enhancer input level above which full HF enhancement occurs. Higher settings produce lower amounts of enhancement at low levels. Default is 0 dB, which produces the same behavior as previous versions of the HF enhancer.

✶ The HF Enhancer is Driven by the AGC, so its input level tracks the AGC's output level.

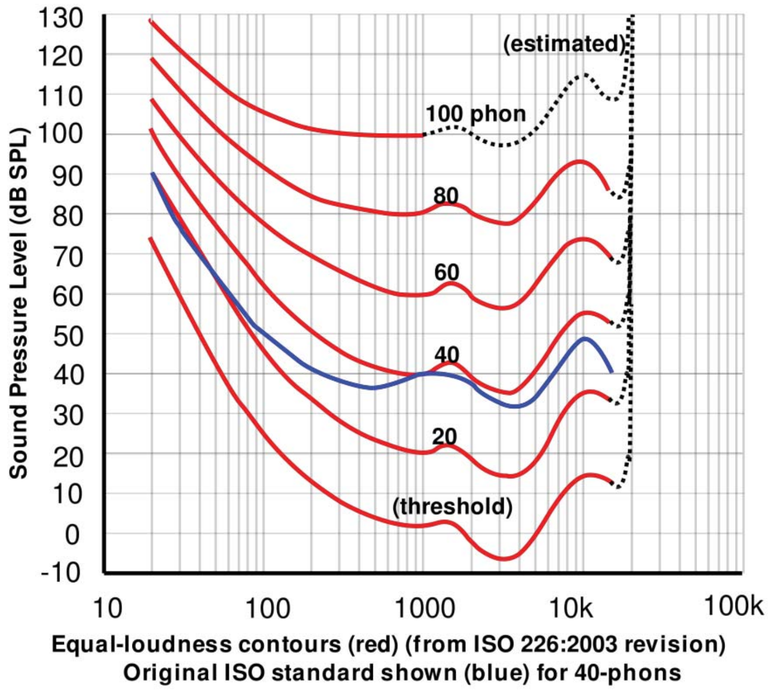


Figure 17.2 Equal Loudness Curves

High Pass Filter

This setting determines if a sweepable 6, 12, 18 or 24 dB/octave high-pass filter will be placed in-circuit before other processing. The 6, 12 and 18dB/octave filters have a Butterworth response and the indicated Frequency is where the filter is 3dB down. The 24dB/octave filter has a Chebychev response with 0.01dB of passband ripple. The indicated high-pass Frequency of this filter is the passband edge of the filter. (For example, if the filter is set to 20Hz, the high filter's response will be flat ±0.01dB to 20Hz. The high-pass filter is useful for reducing low Frequency noise, particularly when RSAP is being used for production or mastering.

Two setting sets (each containing Frequency and Slope) are available, one for music mode and one for speech mode. These can be switched via RSAP's automatic speech/music detector.

It is recommended to make sure that the main and speech-mode high-pass filter settings are the same unless you specifically want to apply additional high-pass filtering to speech. Like all automatic speech/music detectors, the one in your RSAP can occasionally make mistakes, so it is also recommended in mixed-format programming to set the filter no higher than 70 Hz. This way, if the RSAP mistakenly identifies music as speech (or decides that voice-over-music is speech), it will not cause an obvious dropout of bass frequencies.

✶ An L=R test tone whose level changes depending on whether speech or music is detected can repeatedly toggle the speech/music detector between "speech" and "music" modes. This is because the sudden jump in tone level is "speech-like" and triggers the speech/music detec-tor's syllabic detector. An example is a 100 Hz tone when the main high-pass filter Frequency is 50 Hz and the speech high-pass filter Frequency is 200 Hz. This issue never occurs with program material.

Bass Boost

("DJ Bass Boost") setting determines the amount of bass boost produced on some male voices. In its default OFF position, it causes the gain reduction of the lowest Frequency band to move quickly to the same gain reduction as its nearest neighbor when gated. This fights any tendency of the lowest Frequency band to develop significantly more gain than its neighbor when processing voice because voice will enable the gate frequently. Each time it does so, it will reset the gain of the lowest Frequency band so that the gains of the two bottom bands are equal and the response in this Frequency range is flat. This is particularly desirable for most video programming.

When this setting is not set to OFF, gating causes the gain reduction of the lowest Frequency band to move to the same gain reduction (minus a gain offset equal to the numerical setting of the setting) as its nearest neighbor when gated. You can therefore set the maximum gain difference between the two low Frequency bands, producing considerable dynamic bass boost on voice. This setting might be appropriate for news and sports.

The difference will never exceed the difference that would have otherwise occurred if the lowest Frequency band were gated independently. The amount of bass boost will be highly dependent on the fundamental Frequency of a given voice. If the fundamental Frequency is far above 100Hz, there will be little voice energy in the bottom band and little or no audio bass boost can occur even if the gain of the bottom band is higher than the gain of its neighbor. As the fundamental Frequency moves lower, more of this energy leaks into the bottom band, and you hear more bass boost. If the fundamental Frequency is very low (a rarity), there will be enough energy in the bottom band to force significant gain reduction, and you will hear less bass boost than if the fundamental Frequency were a bit higher.

✶ This setting is only available in the Five-Band structure.

✶ If the GATE THRESH (Gate Threshold) is set to OFF, the DJ BASS boost setting is disabled.

Phase Rotator

Phase Rotator determines if the phase rotator will be in-circuit. The purpose of the phase rotator is to make voice waveforms more symmetrical. Because it can slightly reduce the clarity and definition of program material, we recommend leaving it OUT unless program material is mainly speech, where it may result in cleaner sound because it can substantially reduce the amount of gain reduction that RSAP’s look-ahead limiter produces on speech waveforms.

Multiband Controls

This chapter describes the Multiband settings of the audio processor.

Overview

Following the AGC is an equalization section, a five-band compressor, a dynamic single-ended noise reduction system, an output mixer (for the five bands), and a peak limiter.

When the input is noisy, you can sometimes reduce the noise by enabling the single-ended noise reduction system. Functionally, the single-ended noise reduction system combines a broadband downward expander with a program-dependent low-pass filter. This noise reduction can be valuable in reducing audible hiss, rumble, or ambient studio noise. We use it for the news and sports factory presets.

The Five-Band structure does not have a separate Loudness Controller because its Five-Band compressor automatically re-equalizes the spectral balance of various pieces of program material in a way that tends to make their loudness more consistent.

The Five-Band Structure's Full and Advanced Setup Controls The tables below summarize the Five-band and Band Mix settings in the dynamics section. The AGC, Equalizer, Stereo Enhancer, and Clipper settings are common to both the Two-Band and Five-Band structures and are discussed in their own sections.

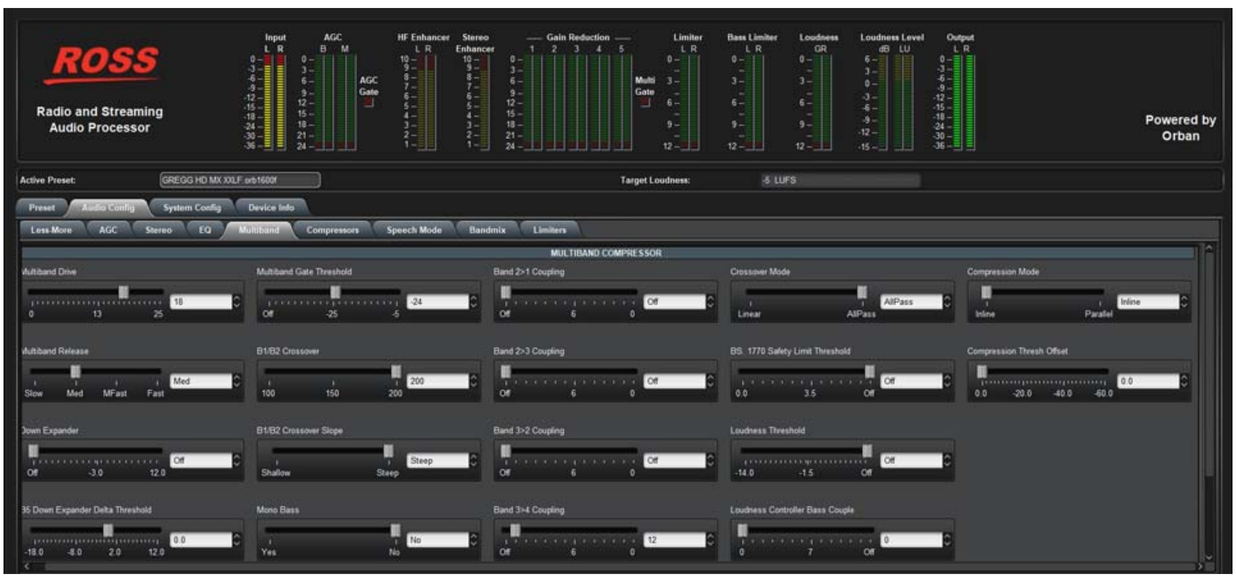


Figure 18.1 Example of the Multiband Controls in DashBoard

Multiband Drive

Multiband Drive setting adjusts the signal level going into the five-band compressor, and therefore determines the average amount of gain reduction in the five-band compressor. Range is 25dB. It works differently in INLINE and PARALLEL compressor modes. In INLINE mode, the setting sets the Drive into the multi-band compression and thus the amount of gain reduction that it produces.

In PARALLEL mode, the Multiband Drive setting is re-purposed so that it sets the amount of compressor output that is added to the compressor's input. Its setting indicates the amount of amplification of quiet material that occurs when the compressor produces no gain reduction. It does not affect the amount of compressor gain reduction; set this with the Compression Thresh Offset setting, which only works in parallel mode. In PARALLEL mode, it is a mix setting located after the multi-band compressor and sets how much of the compressor's output is added to its input signal before being passed to the next processing stage.

Adjust the Multiband Drive setting to your taste and programming requirements.

Used lightly with a slow or medium release time, the Five-Band compressor produces an open, re-equalized sound that is appropriate for most video programming. The Five-Band compressor can increase audio density when operated at a fast or medium-fast release because it acts more and more like a fast limiter (not a compressor)as the release time is shortened. With fast and medium-fast release times, density also increases when you increase the Drive level into the Five-Band compressor because these faster release times produce more limiting action. Increasing density can make loud sounds seem louder, but can also result in an unattractive busier, flatter, or denser sound. It is very important to be aware of the many negative subjective side effects of excessive density when setting settings that affect the density of the processed sound.

Because the RSAP's AGC algorithm uses sophisticated window gating, it is preferable to make the AGC do most of the gain riding (instead of the five-band compressor), because the AGC can ride gain quickly without adding excessive density to program material that is already well defined. The five-band compressor is typically used with 5 to 10 dB of average gain reduction so it can perform automatic re-equalization of material that the AGC has already defined without adding excessive density to the audio or re-equalizing to an unnatural extent.

The Multiband Drive interacts with the Multiband Release. With slower release time settings, increasing the Multiband Drive setting scarcely affects density. Instead, the primary danger is that the excessive Drive will cause noise to be increased excessively when the program material becomes quiet. You can minimize this effect by enabling the single-ended noise reduction and/or by carefully setting the Multiband GATE Threshold setting to freeze the gain when the input gets quiet.

When the release time of the Five-Band compressor is set towards FAST, the setting of the Multiband Drive setting becomes much more critical to sound quality because density increases as the setting is turned up. Listen carefully as you adjust it.

With these fast release times, there is a point beyond which increasing the Five-Band compressor Drive will no longer yield more loudness, and will simply degrade the punch and definition of the sound. Instead, let the AGC do most of the work. Because excessive loudness is an irritant in sound for picture, there is almost never any reason to push processing to the point where it degrades the audio. We recommend no more than 10dB gain reduction as shown on the meters for Band 3. More than 10dB, particularly with the fast release time, will often create a wall of sound effect that many find fatiguing.

To avoid excessive density with fast Five-Band release time, we recommend using no more than 5dB gain reduction in band 3, compensating for any lost loudness by speeding up the AGC Release instead.

Multiband Release

This setting can be switched to any of seven settings. To understand how to adjust this setting for video programming, see the discussion above under MB Drive.

Multiband Gate Threshold

This setting determines the lowest input level that will be recognized as program by RSAP; lower levels are considered to be noise or background sounds and cause the AGC or multi-band compressor to gate, effectively freezing gain to prevent noise breathing.

There are two independent gating circuits in RSAP. The first affects the AGC and the second affects the five-band compressor. Each has its own threshold setting.

The multi-band silence gate causes the gain reduction in bands 2 and 3 of the fiveband compressor to move quickly to the average gain reduction occurring in those bands when the gate first turns on. This prevents obvious midrange coloration under gated conditions, because bands 2 and 3 have the same gain.

The gate also independently freezes the gain of the two highest Frequency bands (forcing the gain of the highest Frequency band to be identical to its lower neighbor), and independently sets the gain of the lowest Frequency band according to the setting of the DJ BASS boost setting (in the Equalization tab). Thus, without in troducing obvious coloration, the gating smoothly preserves the average overall Frequency response "tilt" of the five-band compressor, broadly maintaining the “automatic equalization” curve it generates for a given piece of program material.

✶ If the MB GATE is set to OFF, the DJ BASS setting (in the Equalization tab) is disabled.

Crossover Mode

This feature sets the structure of the five-band crossover to ALLPASS or LINEAR.

In LINEAR mode, the delay through the five-band compressor is essentially constant at all frequencies. Accordingly, each crossover band is down 6 dB at the crossover Frequency.

In ALLPASS mode, the magnitude response through the crossover is flat when all bands have the same amount of gain reduction, but the five-band compressor applies an overall phase rotation to the sound—different frequencies are delayed by different amounts of time. Each crossover band is down 3 dB at the crossover Frequency.

In general, LINEAR sounds more “precise” than ALLPASS and preserves transient detail better. However, ALLPASS can sound “fuller” or “bigger” than LINEAR. The audible difference between the two is often subtle and neither is “correct”; choosing a crossover type is a matter of preference. Most loudspeakers, including many highly regarded models, use allpass crossovers, and given this fact, it is important not to assume that LINEAR is always correct just because it introduces no phase distortion.

B1/B2 Crossover

The Band 1 to Band 2 Crossover Frequency sets the crossover Frequency between Bands 1 and 2 to either 100Hz or 200 Hz. It significantly affects the bass texture, and the best way to understand the differences between the two crossover frequencies is to listen.

B1/B2 Crossover Slope

This setting determines the selectivity of the filter used to separate the frequencies applied to the Band 1 and Band 2 crossover side-chains, which compute the gain applied to the audio in each Frequency band. The setting does not affect the shape of the crossover in the audio path, only in the side-chain, where it determines how much the Band 1 and Band 2 gain reductions are affected by frequencies outside the main passband of a given band. For example, when this setting is set to STEEP and the B1/B2 CROSSOVER setting is set to 200Hz, a strong signal at 100Hz (which is outside the main Frequency range covered by Band 2) will produce less gain reduction in Band 2 than it would if the setting was set to SHALLOW.

Subjectively, a setting of STEEP produces a more consistent bass texture between various program elements than does a setting of SHALLOW. STEEP is the preferred choice if you want consistent, punchy bass texture. On the other hand, SHALLOW preserves more of the original bass coloration of the source because it produces less gain reduction difference between Band 1 and Band 2 than does STEEP.

Mono Bass and Mono Bass Crossover

This setting causes the two input channels to be blended to mono below a Frequency set by the Mono Bass Crossover setting. This function is implemented via a linear phase high-pass filter applied to the stereo difference channel (L-R). A compensating delay in the stereo sum channel makes the transition between subjectively perfect separation and mono as narrow-band as possible.

If there is significant bass energy in the L-R channel, this process will reduce the overall bass heard by the listener. Because the process precedes the multi-band compressor, this will tend to restore this loss of bass by producing less gain reduction in Band 1. This will be most effective if the B1/B2 CROSSOVER Slope setting is set to STEEP and you use 100 Hz B1/B2 CROSSOVER.

Sometimes, it is useful to use the bass shelving equalizer in the EQ section to produce a bit of bass boost to compensate statically for bass loss caused by the mono bass process.

Down Expander

This setting determines the level below which the single-ended noise reduction system's downward expander begins to decrease system gain and below which the high frequencies begin to become low-pass filtered to reduce perceived noise. There are three options: the MB Down Expander and SPEECH MB Down Expander settings set the expansion threshold in Bands 1-4 for Music and Speech modes, while the B5 Down Expander Delta Threshold setting offsets the expansion threshold in Band 5 with respect to the active MB Down Expander threshold for both Speech and Music modes. Activate the single-ended dynamic noise reduction by setting these settings to a setting other than OFF.

The single-ended noise reduction system combines a broadband downward expander with a program-dependent low-pass filter. These functions are implemented by causing extra gain reduction in the multi-band compressor. You can see the effect of this extra gain reduction on the gain reduction meters. The maximum expander gain reduction achievable is constrained to 6dB in Bands 2-5 and 18 dB in Band 1, as this range often contains hum and/or rumble that can benefit from extra noise reduction.

Ordinarily, the gating on the AGC and multi-band limiter will prevent objectionable build-up of noise and you will want to use the single-ended noise reduction only on unusually noisy program material. Modern commercial recordings will almost never need it. We expect that its main use will be in talk-oriented programming, including sports.

You will get best results if you set the MB Down Expander of the noise reduction system to complement the program material you are processing. The MB Down Expander should be set higher when the input is noisy and lower when the input is relatively quiet. The best way to adjust the MB Down Expander is to start with the setting set very high. Reduce the setting while watching the gain reduction meters. Eventually, you will see the gain increase in sync with the program. Go further until you begin to hear noise modulation — a puffing or breathing sound (the input noise) in sync with the input program material. Set the MB Down Expander higher until you can no longer hear the noise modulation. This is the best setting.

Obviously, the correct setting will be different for a sporting event than for classical music. It may be wise to define several presets with different settings of the MB Down Expander and to recall the preset that complements the program material of the moment.

✶ It is virtually impossible to achieve undetectable dynamic noise reduction of program material that is extremely noisy to begin with, because the program never masks the noise. It is probably wiser to disable the dynamic noise reduction with this sort of material (traffic reports from helicopters and the like) to avoid objectionable side effects. You must let your ears guide you.

Band 5 is particularly critical for noise reduction because much of the Down Expander's utility lies in hiss reduction. Hiss has most of its energy in band 5, while program material typically has less energy in this band, so the B5 Down Expander setting's setting is critical to removing hiss while minimizing removal of desired program energy.

Band 5 is uncoupled from the lower bands so the Band 5 downward expander can produce less gain reduction than other bands. This can help prevent loss of desired high Frequency material in the program.

Band 3>4 Coupling

This setting determines the extent to which the gains of Band 4 (centered at 3.7kHz) and Band 5 (above 6.2kHz) are determined by and follow the gain of Band 3 (centered at 1kHz). Set towards 100% (fully coupled) this setting reduces the amount of dynamic upper midrange boost, preventing unnatural upper midrange boost. The gain of Band 5 is further affected by the B4>B5 CPL setting.

Excessive HF energy is one cause of audibly objectionable artifacts in low bit-rate codecs. The B3>B4 CPL AND B4>B5 CPL settings can be very useful in reducing such artifacts and setting them for large amounts of coupling will minimize RSAP's ability to increase high Frequency energy dynamically.

Band 4>5 Coupling

B4>B5 CPL (“Band 4>5 Coupling”) settings the extent to which the gain of Band 5 (6.2 kHz and above) is determined by and follows the gain of band 4.

The sum of the high Frequency limiter setting signal and the output of the B4>B5 CPL setting determines the gain reduction in Band 5. The B4>B5 CPL setting receives the independent left and right band 4 gain setting signal. Range is 0 to 100% coupling.

B3>B2 Coupling and B2>B3 Coupling

This setting determines the extent to which the gains of Bands 2 and 3 track each other. When combined with the other coupling settings, these settings can adjust the five-band processing to be anything from fully independent operation to quasi-wideband processing.

B2>B1 Coupling

This setting determines the extent to which the gain of Band 1 (below 100Hz or 200Hz, depending on crossover setting) is determined by and follows the gain of Band 2 (centered at 400Hz). Set towards 100% (fully coupled), it reduces the amount of dynamic bass boost, preventing unnatural bass boost. Set towards 0% (independent), it permits frequencies below 100Hz (the “slam” region) to have maximum impact in modern rock, urban, dance, rap, and other music where bass punch is crucial. Accordingly, it can be useful in music video oriented formats.

Loudness Threshold

This setting specifies the maximum subjective loudness allowed by the CBS Loudness Controller with reference to the input of the RSAP’s MB look-ahead limiter.

Compressor Controls

This chapter outlines the compression threshold settings for the audio processor.

Figure 19.1 Example of the Compressor Controls in DashBoard

Multiband Delta Release

Delta Release settings are differential settings. They allow you to vary the release time in any band of the Five-Band compressor/limiter by setting an offset between the Multiband Release setting and the actual release time you achieve in a given band. For example, if you set the Multiband Release setting to medium-fast and the BAND 3 Delta GR setting to -2, then the Band 3 release time will be the same as if you had set the Multiband Release setting to medium and set the BAND 3 Delta GR setting to 0. Thus, your settings automatically track any changes you make in the Multiband Release setting. In our example, the release time in Band 3 will always be two “click stops” slower than the setting of the Multiband Release setting.

If your setting of a given Delta Release setting would otherwise create a release slower than “slow” or faster than “fast” (the two end-stops of the Multiband Release setting), the band in question will instead set its release time at the appropriate end-stop.

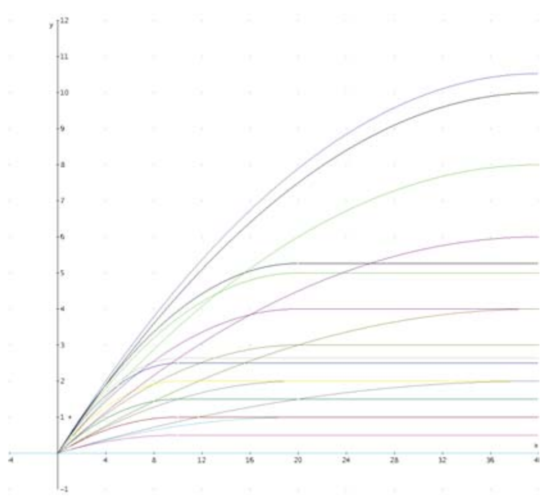


Figure 19.2 Output Level in dB (y) for a Given Input Level in dB (x) at Various Settings of the KNEE and RATIO

Multiband Compressor Threshold

These settings specify the compression threshold for music and speech in each band (following RSAP's automatic speech/music discriminator), in units of dB. We recommend making small changes around the factory settings to preserve the internal headroom built into the processing chain. These settings will affect the spectral balance of the processing above threshold, but are also risky because they can significantly affect the amount of distortion produced by the back-end clipping system.

B1-B5 Attack

You can use these settings to set independent Frequency balances for music and speech.

Attack (Time); Speech B1-B5 Attack settings set the speed with which the gain reduction in each band responds to level changes at the input to a given band's compressor for music and speech respectively, following RSAP's automatic speech/music detector. These settings are risky and difficult to adjust appropriately. They affect the sound of the processor in many subtle ways. The main trade-off is "punch" (achieved with slower attack times) versus distortion and/or pumping produced in the look-ahead limiter (because slower attack times increase overshoots that the look-ahead limit must eliminate). The results are strongly program dependent and must be verified with listening tests to a wide variety of program material.

Because there are separate settings for music and speech, you can set attack times faster for speech (to minimize look-ahead limiter artifacts) and slower for music (to maximize punch and transient definition).

The ATTACK time settings are calibrated in arbitrary units that very approximately correspond to milliseconds. Higher numbers correspond to slower attacks.

Multiband Limiter Attack

These settings allow you to set the limiter attack anywhere from 0 to 100% of normal in the Five-Band compressors, each of whose gain reduction has a fast-release (limiter) and slow-release (compressor) component. Because the limiter and compressor characteristics interact, you will usually get best audible results when you set these settings in the range of 70% to 100%. Below 70%, you will usually hear gain pumping because the compressor function is trying to create some of the gain reduction that the faster limiting function would have otherwise achieved. If you hear pumping in a band and you still wish to adjust the limiter attack to a low setting, you can sometimes ameliorate or eliminate the pumping by slowing down the compressor attack time in that band.

✶ These settings do not impact the final peak limiter.

B1-B5 Compression Ratio

This feature sets the compression ratio of a given band at its thresholds of compression. Beyond threshold, the ratio increases with increased gain reduction until it becomes ..:1 at the amount of gain reduction (in dB) set by the B[X] Compressor Knee setting. When you adjust these settings, the thresholds of the multi-band compressors change automatically so that the total amount of gain reduction stays approximately the same. (This automatic adjustment is internal to the RSAP's DSP; the MB THRESH settings' displayed settings do not show it.)

To achieve a classic soft knee characteristic, set the Compression Ratio setting to 1:1 and set the Compression Knee setting to the gain reduction in dB at which you wish the compression ratio to level off to ..:1. The maximum setting produces the softest knee. Setting the Knee to 0dB produces a classic hard knee curve with ..:1 compression ratio regardless of the setting of the corresponding Compression Ratio setting.

Multiband Release Rate Breakpoint

The release rate (measured in dB/second) in the RSAP's compressors is constant when the gain reduction is higher than the setting's setting, and exponential when the gain reduction is lower than the setting's setting. When the release is exponential, the release rate is proportional to the amount of gain reduction.

Compression-induced audio density remains constant when the gain reduction is above the Breakpoint setting. When the gain reduction is below the Breakpoint setting, density decreases proportionally to the amount of gain reduction. For example, if the Breakpoint is set to 10dB, the release rate (in dB/second) will be constant when the gain reduction is above 10dB. Between 10dB and 0dB gain reduction, the release rate will slow down more and more.

The calibration of the Breakpoint settings is only accurate when Knee = 0dB and/or Ratio = infinity:1 — i.e., when the compression ratio is essentially infinite. When the ratio is less than infinite, the effective breakpoint of the compressor will be lower than Compressor Breakpoint setting.

The main use of the Compressor Breakpoint setting is to prevent the compressor from objectionably increasing audio density when using low compression ratios and a significant amount of gain reduction—for example, 10 dB. The Breakpoint setting is best adjusted by ear. If you find that density increases too much as gain reduction increases, lower the Compressor Breakpoint setting. If you want more density at high amounts of gain reduction, increase the Compressor Breakpoint setting. A value of 10dB is a good starting point for this setting.

Multiband Max Delta GR

This settings defines the maximum permitted gain difference between the left and right channels for each band in the multi-band limiter. The five-band processing chain uses a full dual-mono architecture, so the channels can be operated anywhere from fully coupled to independent. We recommend operating Band 1-4 fully coupled (BAND 1-4 MAX Delta GR = 0) for best stereo image stability. However, audio-processing experts may want to experiment with lesser amounts of coupling to achieve a wider, “fatter” stereo image at the cost of some image instability.

B5 MAX Delta GR is set OFF most factory presets. This permits Band 5 to be used as a fast-operating high Frequency limiter that works independently on the left and right channels. This prevents gain reduction in one channel from causing audible spectral modulation on the other channel. However, the additional stereo difference channel energy created by independent operation can adversely affect certain low bitrate codecs (like WMA). It is wise to do careful listening tests through the codec to determine if it sounds better with B5 MAX Delta GR = 0 dB.

Bandmix Controls

This chapter outlines output mix controls of the audio processor.

Figure 20.1 Example of the Bandmix Controls in DashBoard

Output Mix

Because these settings mix after the band compressors, they do not affect the compressors' gain reductions and can be used as a graphic equalizer to fine-tune the spectral balance of the program material over a 6dB range. Their range has been purposely limited because the only gain setting element after these settings is the look-ahead limiter, which can produce pumping or distortion if over-driven. The thresholds of the individual compressors have been tuned to prevent audible distortion with almost any program material. Large changes in the Frequency balance of the compressor outputs will change this tuning, leaving RSAP more vulnerable to unexpected audible distortion with certain program material.

You can also get a similar effect by adjusting the compression threshold of the individual bands. This is comparably risky with reference to look-ahead limiter overload, but unlike the MB Band Mix settings, the threshold adjustments do not affect the Frequency response when a given band is below threshold and is thus producing no gain reduction.

B1-B5 On/Off

This setting allows you to mute any combination of bands in the five-band compressor and permit you to "solo" any individual band. This can be useful when you are designing new user presets.

Limiters Controls

This chapter summarizes the bass clip threshold, bass pre-limit mode, multiband final limit drive, Transient Enhance, and MX Limiter settings.

Figure 21.1 Example of the Limiters Controls in DashBoard

Overview

Regardless of whether MX mode is ON or OFF, a bass pre-limiter stage settings low Frequency peaks ahead of the main peak limiter, while a “true-peak”-aware lookahead limiter implements final peak setting. This limiter typically constrains the true peak overshoot (i.e., the maximum peak value that would appear after an ideal digital-to-analog converter) to 0.2dB or less.

Bass Clip Threshold

This feature sets the threshold of the bass pre-limiter with respect to the threshold of the final peak limiter. The Bass Limiter gain reduction meters show the amount of bass clipping or limiting in each audio channel.

✶ The Speech Mode > Bass Clip Threshold setting overrides the Bass Clip Threshold setting when RSAP automatically detects speech.

Bass Pre-limit Mode

This sets the operation of the bass pre-limiter to HARD, MEDIUM, or SOFT. This setting only works when the MX Limiter is ON. When the MX Limiter is OFF, the bass clip mode is always MEDIUM.

HARD

This setting produces the most harmonic distortion. This can be useful if you want maximum bass punch, because this setting allows bass transients (like kick drums) to make square waves. The peak level of the fundamental component of a square wave is 2.1dB higher than the peak level of the flat top in the square wave. Therefore, this allows you to get low bass that is actually higher than 100% modulation (or 0dBFS)—the phase of the harmonics produced by the clipping works to reduce the peak level of the overall program. The squarewaves produced by this clipper are low-pass-filtered to reduce the audibility of the higher clipper-generated harmonics. Nevertheless, the downside is that material with sustained bass (including speech) can sound somewhat less clean than it will with the MEDIUM or SOFT settings.

MEDIUM

This setting uses more sophisticated signal processing than HARD to reduce distortion substantially.

SOFT

This setting uses the most sophisticated look-ahead signal processing to reduce distortion further.

✶ The trade-off of using MEDIUM or SOFT bass clipping is less bass punch compared to HARD.

Multiband Final Limit Drive

This setting adjusts the level of the audio driving the peak limiter, thereby adjusting the peak-to-average ratio of the processed audio. The MB Final Limit Drive setting primarily determines the loudness/distortion trade-off.

Turning up the MB Final Limit Drive setting Drives the look-ahead limiter harder, reducing the peak-to-average ratio, and increasing the loudness of the output.

When the amount of limiting is increased, the audible inter-modulation distortion caused by limiting increases, even though special algorithms minimize the increase compared to less sophisticated designs. Lower settings reduce loudness, of course, but result in a cleaner sound.

The MX peak limiter uses a variety of tactics to adapt its operation intelligently to the program material applied to it. Compared to the look-ahead peak limiter in non-MX mode, the M X peak limiter produces a different set of artifacts when over-driven.

Depending on how you set the other settings, these artifacts may include loss of bass punch, harsh clipper-like distortion, “soft” IM distortion, and/or excessive density that can cause fatigue. If you intend to make adjustments to the Final Limit Drive setting, it is wise to familiarize yourself with these artifacts by purposely overdriving the peak limiter via the MB Final Limit Drive setting and listening to what happens with different types of program material.

When using pre-emphasis, it is usually necessary to turn down the MB Final Limit Drive setting to prevent the peak limiter from causing objectionable artifacts.

✶ You may find it illuminating to recall several Factory Presets, adjust the Less-More settings to several points in its range, and then examine the trade-offs between the release time and Final Limit Drive made by the factory programmers. However, note that all Factory Presets were created to complement FLAT pre-emphasis. As explained above, you must turn down the Final Limit Drive setting when using pre-emphasis.

Transient Enhance

This setting is mainly useful in mastering. This setting allows you to insert an audio delay in the side-chain of the five-band compressor. By delaying the gain setting signal, this allows attack transients to pass through the multi-band compressor uncompressed, which can increase punch. There is a trade-off between this setting and the activity of the look-ahead limiter, which will have to eliminate attack transients exceeding the look-ahead limiter’s threshold. For any material, there will be an optimum setting for the Transient Enhance setting that provides the most punch without triggering peak limiter artifacts.

MX Limiter

When MX Limiter is ON, additional peak setting occurs between the bass pre-limiter and the look-ahead limiter, and the look-ahead limiter is used very lightly for final overshoot setting only. When the peak limiter section is being Driven heavily, the MX peak limiter, which uses a psychoacoustic model, provides decreased audible distortion, more transient punch and a crisper, more open high Frequency texture. When MX Limiter is OFF, the final look-ahead limiter implements all peak setting and is more likely to cause audible gain pumping when Driven heavily.

The MX Limiter has two downsides: its sophisticated processing almost doubles the amount of CPU power required compared to non-MX mode, and it adds several hundred milliseconds of input-to-output delay. (The delay is required so that the MX Limiter has time to make intelligent decisions about how to process.) Disabling the MX Limiter can allow more instances of processing to be run on the CPU and will decrease delay.

When the target loudness is below -12 LUFS, the MX Limiter is unlikely to provide significant audible advantages because the overall peak limiter section is not being Driven hard, so the final lookahead limiter (which is itself quite capable) is unlikely to produce audible artifacts when used alone. If you want to minimize CPU usage and/or delay, it is wise to perform listening tests to see if the MX Limiter provides audible advantages with your program material. In mastering applications, on the other hand, neither CPU usage nor delay are usually relevant, so you may wish to save time by always enabling the MX Limiter because it will never degrade the sound compared to not using it.

When the target loudness is below -22 LUFS (as is typical is sound-for-picture applications), the MX Limiter never provides audible advantages because at this target loudness, the peak limiter rarely produces any gain reduction at all.

The only exception to these recommendations is when pre-emphasis is enabled. When you use RSAP in a pre-emphasized mode, this inserts a Frequency-dependent high Frequency boost before peak limiter. This boost can be as large as 20dB at 20kHz. The peak limiter handles this boost far more smoothly when the MX Limiter is active, so we strongly advise always using the MX Limiter when you use pre-emphasis.

The settings described below are only active when the MX Limiter is ON. They allow you to trade off loudness, distortion, and bass energy. Increasing bass increases the likelihood that audible IM distortion between bass and other program elements will occur. Depending on the program material you are processing, you may prefer to have a cleaner sound (with less bass) or a sound with more distortion but punchier bass.

MX Limiter Threshold

This sets the threshold of the MX Limiter with respect to the threshold of the final look-ahead limiter, used for overshoot setting. This setting is normally set to 0dB. When the peak limiter is Driven very hard (as determined bythe setting of the MB Final Limit Drive setting), setting the MB LIMITER Threshold slightly below 0dB can reduce gain pumping artifacts that would otherwise be caused by the final look-ahead limiter.

Bass Pre-limiting

The MX bass pre-limiter can intelligently reduce the bass applied to the main peak limiter to reduce or prevent audible IM distortion. It does so when the pre-limiter’s analysis of the program material indicates that this action is needed to prevent or minimize audible IM distortion between the bass (125 Hz and below) and other program elements in the main peak limiter. The Bass Pre-limiting setting allows you to specify the maximum amount of bass reduction that can occur. Lower settings increase bass punch but do not protect against IM distortion as effectively as higher settings do.

There are two settings, one for Speech mode and one for Music mode, allowing you to have separate settings depending on whether RSAP automatically detects speech or music input.

Bass Limiting

Like the bass pre-limiter, the main peak limiter can automatically reduce bass when it detects potentially audible IM distortion. The Bass Limiting setting allows you to limit the amount of potential bass reduction at the expense of a possible increase in IM distortion. The scale shows the maximum amount of dynamic bass cut that can be produced, in dB.

Distortion Control

This setting determines the amount of audible distortion that the main peak limiter is permitted to create. Higher settings can increase loudness and punch at the expense of audible clipping distortion. Lower settings are cleaner but may reduce punch and loudness. We prefer it at 0, which is its cleanest setting. All MX factory presets use this setting.

The best way to familiarize yourself with the effects of this setting is by listening extensively to different types of program material while experimenting with different settings of the setting. Because the MX peak limiter uses advanced algorithms, the loudness/distortion/brightness/punch trade-offs are also different and it is worthwhile to take the time to get a feel for the MX Limiter's capabilities.

MX Overshoot Limit Mode

This setting determines the algorithm used to eliminate overshoots produced by the main part of the MX processing. HARD uses a complex, computationally intensive process that shares certain characteristics with clipping, while SOFT uses a look-ahead limiter. In general, low bit-rate codecs will handle SOFT mode better than HARD mode.

HARD produces punchier transients and is often preferred when the absolute maximum loudness is desired. It adds about 200 ms of delay compared to SOFT, and this is easy to hear when one is switching back and forth between HARD and SOFT modes. (Switching between modes produces audible discontinuities and should not be done "on-air.")

For applications where the MX Limiter is not Driven to its limits, SOFT may be preferred because it has less distortion and loads the CPU significantly less. As always, you should choose based on program material and processing goals. Subjectively, it might be said that HARD is more "rock 'n' roll," while SOFT is more "polite."

✶ Because the RSAP software is not licensed as a processor for FM radio, HARD mode is locked out when the pre-emphasis is set to 50µs or 75µs.

Audio Meters in DashBoard

This chapter provides a brief overview of the audio meter panel in the DashBoard window.

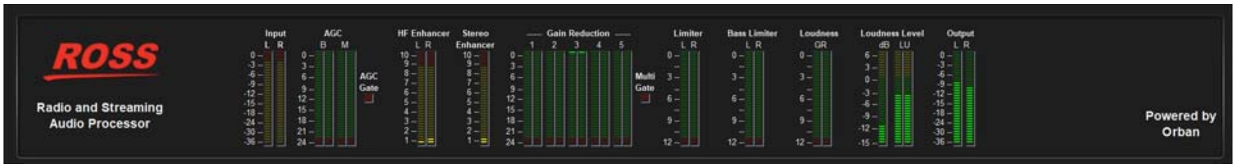


Figure 22.1 Example of the Audio Meters in DashBoard

Input

This meter reports the peak input level applied to the audio processing, in units of dBFS. Higher levels call for higher settings of the Input Reference Level setting. If the meter indicates "0," this indicates that the input peak level has reached 0dBFS. This is common if the program material applied to the RSAP was aggressively mastered and there is unity gain between the playout system and the input of RSAP.

AGC

This meter reports shows the gain reductions of the Master (M) (above-200 Hz) and Bass (B) (below-200 Hz) bands in the slow AGC processing that precedes the multi-band compressor. Full-scale is 24dB gain reduction.

Gate Indicators

Gate indicators report gate activity. They light when the input audio falls below the threshold set by the gate threshold settings. (There are two gating circuits — one for the AGC and one for the multi-band compressor/limiter — each with its own gate threshold setting.) When gating occurs, the AGC and compressor's recovery times slow drastically to prevent noise rush-up during low-level passages.

HF Enhancer

This meter reports the amount of dynamic 4kHz first-order high Frequency shelving equalization, in dB, being applied to the signal.

Stereo Enhancer

This meter reports the amount of extra gain, in dB, being applied to the stereo difference signal (L-R) to increase the apparent width of the stereo soundstage.

Gain Reduction

This meter reports the gain reduction in the multi-band compressor. Full-scale is 25 dB gain reduction. In a stereo instance, each meter is split vertically to show the gain reduction in the left and right channels. In a surround instance, each meter is split vertically to show the gain reduction in the main (all channels but center) and center channels.

Limiter

This meter reports the amount of peak limiting (dB) in the various audio channels, which we chose not to couple because the fast release time of this circuit would otherwise cause elements in one channel to modulate the opposite channel objectionably. Full-scale is 12dB gain reduction. If the output of a stereo Processor is configured for 5.1-channel, five limiter meters will appear.

Bass Limiter

This meter reports the amount of peak limiting applied to the bass energy in the various audio channels. Full-scale is 12dB gain reduction. If the output of a stereo Processor is configured for 5.1-channel, five limiter meters will appear.

Loudness

This meter reports the amount of gain reduction that the Loudness Controller and BS.1770 Safety Limiter are producing. The CBS setting gain reduction appears in blue, while the BS.1770 Safety Limiter gain reduction appears in cyan. There are two separate meters: the left meter is CBS and the right is BS.1770.

These meters will only show gain reduction if the Loudness Threshold and BS.1770 SAFETY LIMIT Threshold are not off (Multiband tab in the active processing preset), and both loudness settings are enabled in I/O Mixer > Utility.

Loudness Level

This meter reports the subjective loudness of the output, measured using the 1981 Jones & Torick CBS Technology Center algorithm and by the ITU. BS.1770 algorithm.

There is only one meter for each algorithm regardless of the number of channels and loudspeakers in the listening room because a given listener has only one perception of loudness.

The meter is calibrated with reference to the Target Loudness / dialnorm value that you specify in the RSAP's active Setup for that processing chain. If you adjust the processing so that the Loudness Level meter peaks at 0 dB on dialog and you set up your Dolby Digital encoder so that you are transmitting this same dialnorm metadata value to consumer receivers, your transmission will have the correct loudness compared to other correctly set up transmission channels. If the CBS loudness setting is disabled in I/O Mixer > Utility, the CBS Loudness Meter (dB) will show no activity.

Output

This meter reports the peak level of the processed samples at the output, in dBFS. They will show the effect of the 100% Output Level setting. (For example, setting this setting to -2.0dBFS will cause the meters to peak at -2.)

✶ The LFE channel is not displayed because Optimix places energy below 80Hz in the Lf and Rf channels instead of in an LFE channel. This makes it maximally compatible with home theater receivers' bass management functionality— if Lf and Rf speakers are "small," the receiver will route bass to the LFE channel if present.

Upgrading the Software

The Radio and Streaming Audio Processor (RSAP) can be upgraded in the field via DashBoard.

To upgrade the software on the RSAP

1. Contact Ross Technical Support for the latest software version file.
2. Ensure the Ethernet cable is connected to the **Gb1** port on the RSAP.
3. From the **Tree View** in DashBoard, expand the node for the RSAP you want to access.
4. Double-click the Global sub-node to display the interface in the right-half of DashBoard.
5. Select **Upload**, located near the bottom of the interface, to display the **Select file Upload** dialog.
6. Navigate to the ***.bin** file you want to upload.
7. Click **Open**.
8. Click **Next >** to display the **Select Destination** menu. This menu provides a list of the compatible units.
9. Select the check box for the RSAP you want to upload the file to.
10. Verify the RSAP you want to upload the file to.

    The **Error/Warning** fields indicate any errors, such as incompatible software or product type mismatch.
11. Click **Finish**.
12. Monitor the upgrade.
    - An **Upload Status** dialog enables you to monitor the upgrade process.
    - Notice that each RSAP is listed in the dialog with a button. This button is replaced with a **Reboot** button once the software file is loaded to that RSAP.

✶ Avoid clicking the individual Reboot buttons until all units have successfully completed the file upload process and the OK button, located in the bottom right corner of the dialog, is enabled.

13. Click **OK** to close the **Upload Status** dialog.
14. Click **Reboot** to continue the upgrade process.

✶ This **Reboot** button is located at the bottom of the DashBoard window.

- The RSAP is temporarily taken off-line during the reboot process.
- The process is complete once the status indicators for the **Card State** and **Connection** return to their previous status.

DashBoard Interface Overview

This chapter summarizes the interfaces, and tabs available from DashBoard for the Radio and Streaming Audio Processor (RSAP).

Global Interface

The Global interface enables you to define the Ethernet settings for each port on the rear panel, and define the timing requirements for your RSAP. The options are organized into three tabs: Ethernet, Timing, and About.

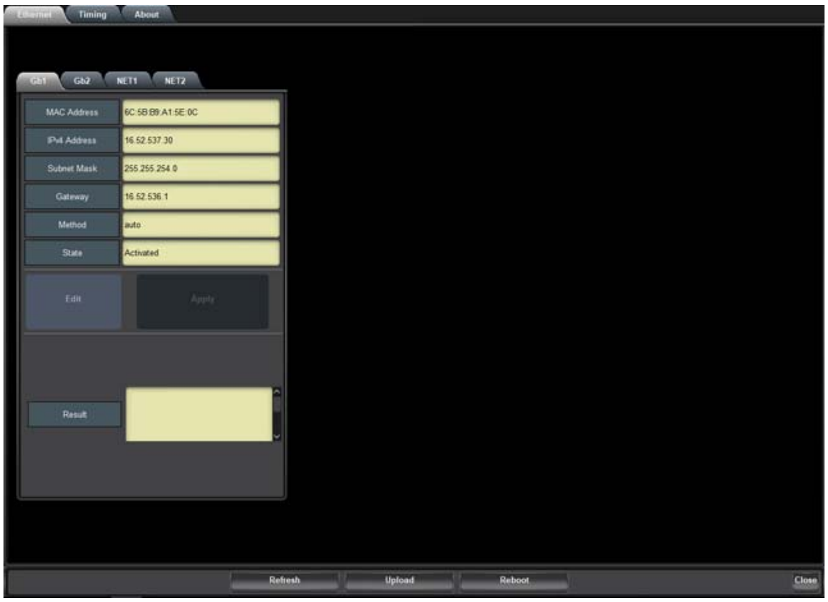


Figure 24.1 Example of the Global Interface in DashBoard

Ethernet Tab

Table 24.1 summarizes the read-only information displayed in the **Ethernet** tab.

Table 24.1 Ethernet Tab

Item	Parameters	Description
Gb#, NET#		
MAC Address	#	Specifies the unique Media Access Control (MAC) Address assigned to the RSAP
IPv4 Address	#	Specifies the static IP Address that the users wants to manually assign to the specified RSAP port
	Unknown	The specified port is not configured
Subnet Mask	#	Specifies the subnet mask value for the specified port on the RSAP
	Unknown	The specified port is not configured
Gateway	#	Specifies the gateway for communications outside of the local area network (LAN) the specified port on the RSAP will use
	Unknown	The specified port is not configured

Table 24.1 Ethernet Tab

Item	Parameters	Description
Method	Auto	The settings will be assigned by a DHCP server in your facility
	Manual	The user manually supplies the Ethernet settings
	Link-local	
	Unknown	The specified port is not used
	DHCP	Automates the assignment of the Ethernet settings
State (read-only)	Activated	The specified port is active and is communicating with valid settings
	Disconnected	Communication to the specified port is interrupted. Verify that the settings are correct and the port is physically cabled correctly.
	Unavailable	The specified port is not active and is not configured for use.
Edit	Click this button to edit the Address, Subnet Mask, gateway, and Method fields for the specified port	
Apply	Click this button to apply the new settings to the specified port	
Result (read-only)	Success	The settings are valid and applied. The specified port is now active.
	Problem: x, y	Indicates that the specified port is experiencing an error where: • **x** states the type of problem • **y** indicates the possible cause

Timing Tab

Table 24.2 summarizes the fields displayed in the **Timing** tab.

Table 24.2 Timing Tab

Item	Parameters	Description
PTP Status (read-only)	OK (Green)	Communication between the RSAP and the Grandmaster or Master device is valid
	GM Not Found (Red)	Connection to the Grandmaster or Master device is invalid. Verify that the Domain field on the Timing tab is set to the correct value.
	Apply Changes	There are unsaved changes made to the Timing tab settings
Current Time (read-only)	#	Reports the current time as reported by your system clock
Clock Selection Mode	Automatic	
	Manual	
Clock Source Selected (read-only)	NET #	Reports the physical port on the RSAP that will communicate with the Grandmaster or Master device

Table 24.2 Timing Tab

Item	Parameters	Description
Profile	AES	Specifies that the RSAP timing uses the AES67 Media standard
	SMPTE	Specifies that the RSAP timing uses the SMPTE ST 2059-2 standards. This is the recommended setting.
	Default	Specifies that the RSAP timing uses the IEEE1588 Default profile standards.
	Free-running	The RSAP automatically selects the timing profile based on the detected stream
Slave Only (read-only)	Selected	Defines the RSAP as a slave only device in the system; the RSAP cannot be used as a Grandmaster or Master device
Domain	#	Specifies that the RSAP is within the group of clocks in your network
Priority 1	#	Assigns the first priority level to the RSAP during a Grandmaster election where a value of: • 1 is the highest priority • 255 is the lowest priority This menu is applicable when the Slave Only box is not selected.
Priority 2	#	Assigns the secondary priority level to the RSAP during a Grandmaster election where a value of: • 1 is the highest priority • 255 is the lowest priority This menu is applicable when the Slave Only box is not selected.
Sync Interval	#	Specifies how often the RSAP sends Sync messages
Announce Interval	#	Specifies how often the RSAP sends Announce messages
Announce Receipt Timeout	#	Controls how long the RSAP will wait before declaring the Grandmaster absent and initiating a new election
Steps Removed (read-only)	#	Specifies the number of communication paths to the Grandmaster clock
Offset from Master (read-only)	#	Specifies the last measured time offset from the master in number of nanoseconds
Mean Path Delay (read-only)	#	Average time in nanoseconds it takes a packet to traverse end to end from the PTP master
Master Offset (read-only)	#	The last measured offset of the clock from the Master in nanoseconds.

Table 24.2 Timing Tab

Item	Parameters	Description
GM Present (read-only)	True	Indicates a Grandmaster is available
	False	Indicates a Grandmaster is unavailable
GM Identity (read-only)	#	Reports the ID number assigned to the Grandmaster within the system

About Tab

Table 24.3 summarizes the read-only information displayed in the About tab.

Table 24.3 About Tab

Item	Parameters	Description
Serial Number	#	The serial number assigned to this RSAP
Assembly Revision	#	The version of the RSAP hardware
Software Version	#	The software build the RSAP is currently running
Frame Name	<text>	Assigns a unique identifier to the RSAP. This name also displays in the node for the RSAP in the Tree View of DashBoard.
Factory Default	Click this button to reset all RSAP editable fields to the factory default values. This impacts all tabs except the Ethernet, Timing, and About tabs.	

AES67 Receiver Interface

The options for configuring the AES67 Receiver are organized into four tabs: Network Streams, Connections, Status, and Device Setup.

Network Streams Tab

The options in the Network Streams tab enable you to create and manage the IP sessions in your system for the AES67 Receiver. A session is defined using the Transport IP, Port, and Link Offset fields for the audio signals.

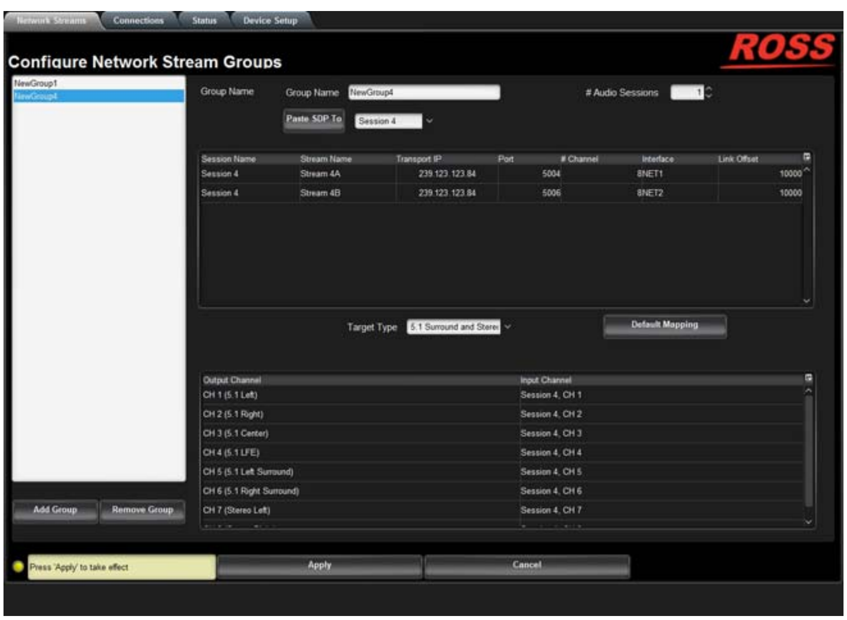


Figure 24.2 Network Streams Tab

Table 24.4 outlines the options displayed in the Network Streams tab starting from the left-most area of the tab.

Table 24.4 Network Streams Tab

Item	Parameters	Description
Configure Network Stream Groups		
List	<name>	Lists the configured groups for the RSAP
Add Group		Enables you to configure a new stream group
Remove Group		Deletes the selected group
Group Name	<text>	Specifies a unique identifier for the group
# Audio Sessions	#	Specifies the number of audio streams available in the stream group. The default value is 2.
Paste SDP To	<session name>	Applies the settings, copied from a device that supports the Session Description Protocol (SDP), to the specified session of the selected RSAP group
Interface	#	Indicates the specified port the RSAP is using for the AES67 Receiver streams
Stream Name	<name>	Assigns a name for the specific audio stream
Session Name	<name>	Assigns a name for the session
Transport IP	#	Specifies the network socket for the audio data for the session
Port	#	Specifies the source port to connect to the stream. This must match the source you are attempting to connect to
# Channel	#	Specifies the maximum number of audio channels in the specified stream
Link Offset	#	Specifies the offset from the input stream to add to the output
Target Type	Stereo	Specifies that the audio stream transports stereo data
	5.1 Surround	Specifies that the audio stream transports Dolby® 5.1 Surround data
	5.1 Surround and Stereo	Specifies that the audio stream transports the six channels of 5.1 Surround and two channels of stereo
Audio Channel Map		
Default Mapping		Automatically maps all incoming streams to the output streams using a 1:1 map
Output Channel	#	The options depend on the Target Type.
Input Channel	#	

Connections Tab

The Connections tab is a patch-panel style interface that enables the RSAP to connect to the configured AES67 network stream groups to output channels.

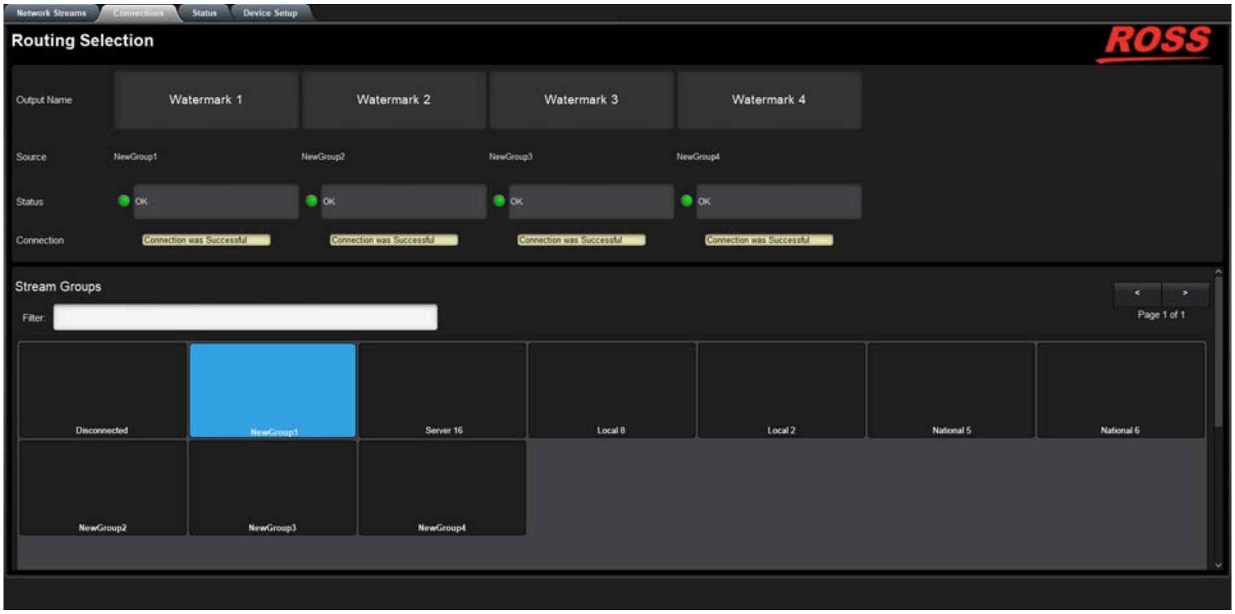


Figure 24.3 Connections Tab

Outputs Area

The Outputs area is located at the top of the Connections tab and provides options for routing signals to the outputs on the RSAP. From this area you can quickly select outputs and monitor the status of the output signals.

Table 24.5 summarizes the buttons, menus, and fields available in the Output area of the Connections tab.

Table 24.5 Connections Tab — Outputs Area

Item	Parameters	Description
Output Name	#	Each button represents an output that is configured and available for switching
Source (read-only)	#	Indicates the network stream currently used by the specified output.
Status (read-only)	OK (Green)	No errors are detected on this output
	Off (Green)	This SDI output is disabled
	Alarm Suppressed (Green)	An alarm condition is present, but the alarm is disabled on the Alarm Enable tab
	Network Delay Too Big (Yellow)	The link offset selected by the user is smaller than the propagation delay of the network
	No packets received (Yellow)	The configured destination IP stream(s) is not receiving any packets; stream might not be on the network or experiencing other issues

Table 24.5 Connections Tab — Outputs Area

Item	Parameters	Description
Status (read-only)	System clock is in failure (Red)	The RSAP is unable to re-obtain a stable clock source. Sessions cannot be created until this condition is fixed. It is recommended to navigate to the Advanced > Timing > PTP tab to check the status of the PTP and update the Configuration settings. Once PTP is locked again, the Network Groups will need to be disconnected and then re-connected to clear the alarm.
	Param Out of Range (Red)	One of the following is occurring: • A Destination was configured with an invalid setting • Two receivers with the same network stream were created. RSAP can only subscribe to a stream once.
Connection (read-only)	Connection was successful	Indicates the connection status between the selected input and output
	Not in Use	
	<blank>	

Stream Groups Area

The Stream Groups area is located on the bottom half of the Connections tab. From this area you can route any source signal to an output. Each button displayed here represents a configured network stream.

✶ Once an Output is selected, clicking a **Stream Group** button performs an immediate switch (a hot-punch). In the example below, **Server10** was selected.

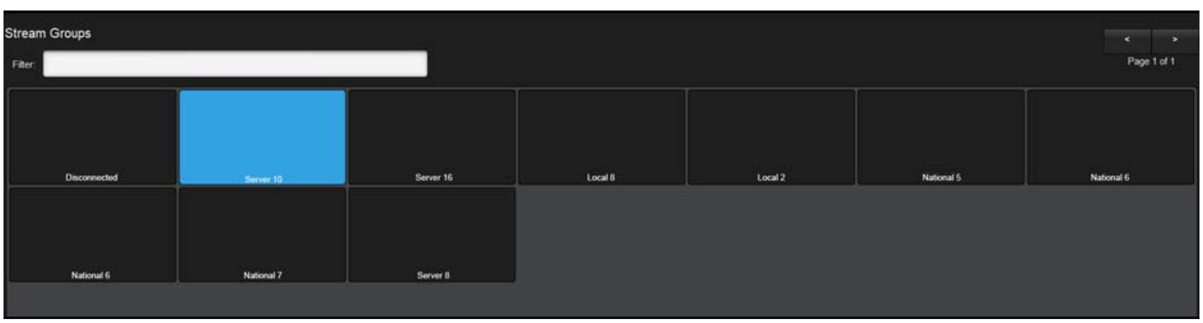


Figure 24.4 Connections — Stream Groups Area

Status Tab

Table 24.6 summarizes the read-only information displayed in the Status tab.

Table 24.6 Status Tab — Device Status

Item	Parameters	Description
Stream Status		
Stream Name	<text>	Displays the unique identifier assigned to the stream (as defined on the Network Streams tab)

Table 24.6 Status Tab — Device Status

Item	Parameters	Description
Session Name	<text>	Displays the unique identifier assigned to the current session
Status	OK (Green)	The incoming stream is valid and is not experiencing errors
	Exceed Link Offset (Yellow)	The link offset selected by the user is smaller than the propagation delay of the network
	No Packet Received (Yellow)	The configured destination IP stream(s) is not receiving any packets; stream might not be on the network or experiencing other issues
	Packets Missing (Yellow)	The configured receiver IP stream(s) is not receiving any packets; stream might not be on the network or experiencing other issues
Packets Received	#	The number of packets received by the RSAP
Buffer Level	#	The number of packets stored in the RSAP buffer
Packets Lost	#	The number of packets lost by the receiver
Packets Dropped	#	The number of packets dropped by the receiver
Packets Out of Order	#	The number of packets that were received but were not in the correct order
Net Delay	#	Specifies the detected network delay in microseconds

Device Setup Tab

Table 24.7 summarizes the read-only information displayed in the Device Setup tab.

Table 24.7 Device Setup Tab

Item	Parameters	Description
Sample Rate	#	Indicates the sample rate of the audio signal
Packet Time	#	Indicates the offset to the audio streams
Default # Channels	#	Specifies the default number of audio channel available for configuration
CODEC	#	Indicates the communication protocol used for the AES67 Receiver stream(s)
Link Offset	#	Reports the Audio Offset and/or Audio Delay values set for the output
Clock Reference	NET #	Assigns the specified port on the RSAP as the timing source
Streams Per Session	#	Indicates the total number of network streams that are active in the current session. The default is 2.
Version	#	The software build the Receiver component of the RSAP is currently running

RSAP Interfaces

The options for configuring the audio processing features are organized into nine separate tabs in the RSAP interface. Refer to the appropriate chapter earlier in this user guide for information on a specific tab.

AES67 Sender Interface

The AES67 Sender interface provides options for configuring the network streams for the AES67 senders.

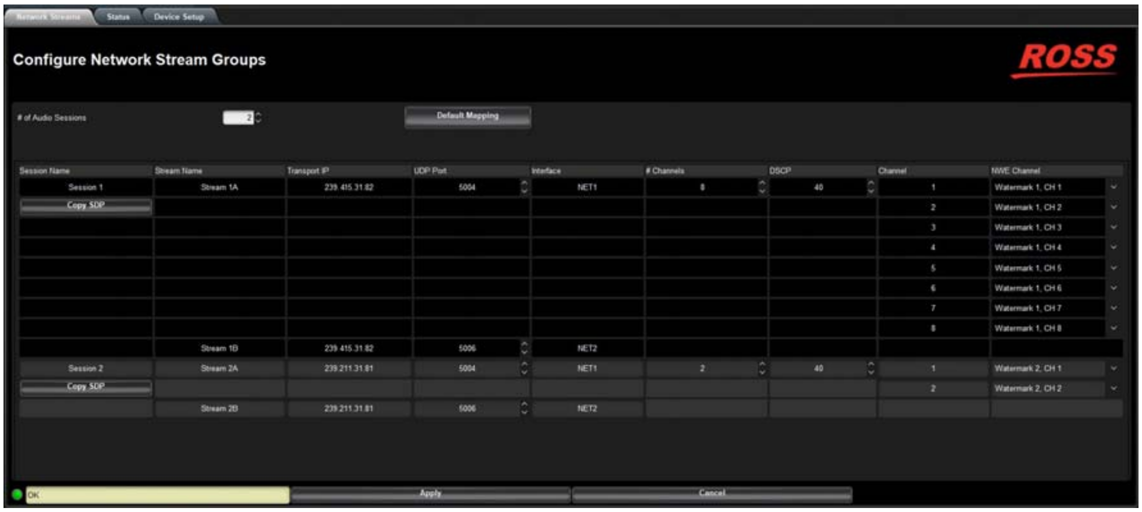


Figure 24.5 Example of the AES67 Sender Interface

Network Streams Tab

Table 24.8 summarizes the options displayed in the Network Streams tab.

Table 24.8 Network Streams Tab

Item	Parameters	Description
# of Audio Sessions	#	Specifies the number of audio sessions for the AES67 Sender. The Network Streams tab updates automatically to provide options for configuring each stream independently.
Default Mapping		Applies a default audio mapping of 1:1 where: • Target 1-8 is mapped to A1-A8 • Target 9-16 is mapped to B1-B8 etc.
Session Name	<text>	Specifies a unique name for the Sender session
Copy SDP	<session name>	Copies the Session Description Protocol (SDP) settings of the specified RSAP session. These settings can then be applied to a downstream device to establish communications via SDP.
Stream Name	<text>	Specifies a unique name for the AES67 stream
Transport IP	#.#.#.#	Specifies the network socket for the audio data for the session
UDP Port	#	Specifies the source port to connect to the advertised stream. This must match the source you are attempting to connect to.

Table 24.8 Network Streams Tab

Item	Parameters	Description
Interface	NET#	Specifies the physical port on the RSAP that is used for the stream
# Channels	#	Specifies the maximum number of audio channels in the specified stream
DSCP	#	Specifies the Differential Service Code Point value for the network layer service for this stream
Channel (read-only)	#	Assigns a channel ID to the stream
NWE Channel	<blank>	The specified stream is not assigned to an output of the RSAP
	Output #, CH #	Assigns the audio stream to an output of the RSAP

Status Tab

Table 24.9 summarizes the read-only information displayed in the Status tab.

Table 24.9 Status Tab

Item	Parameters	Description
Stream Name	<text>	Displays the unique identifier assigned to the specified AES67 stream
Packets Sent	#	Specifies the number of packets sent on the output for the specified stream
Buffer Level	#	The number of packets stored in the RSAP buffer for the specified stream
Buffer Underrun	#	Specifies the number of under runs of the sender buffer
Packets Lost	#	The number of packets lost by the specified stream
Session Name	<text>	Reports the unique name for the Sender session

Device Setup Tab

Table 24.10 summarizes the read-only information displayed in the Device Setup tab.

Table 24.10 Device Setup Tab

Item	Parameters	Description
Packet Time	#	Adds an offset to the audio streams if you suspect the audio packets may be received out of order or delayed. This impacts all connected audio stream. It is recommended to set the Packet Time before configuring your network streams.
Default # Channels	#	Defines the default number of audio channels available for configuration. You can still specify a value other than the default for each individual stream.

Table 24.10 Device Setup Tab

Item	Parameters	Description
CODEC (read-only)	L24	Specifies the Real-time Transport Protocol (RTP) the RSAP uses for delivering audio over the network
	L16	
AES67 Timestamp	Conception Timestamp	The RSAP uses the original timestamp from when the packet was received on the AES67 Receiver stream
	Processed Timestamp	The RSAP replaced the original timestamp with the current PTP timestamp
Clock Reference	NET#	Indicates the physical port on the RSAP that is used as the timing source
Streams Per Session	#	Indicates the total number of network streams that are active in the current session. The default is 2.
Version (read-only)	#	The software build the Sender component of the RSAP is currently running

Technical Specifications

This chapter provides technical information for the Radio and Streaming Audio Processor (RSAP).

✶ Specifications are subject to change without notice.

Gb1 Specifications

✶ The Gb2 port is not implemented.

Table 25.1 Technical Specifications — Gb1

Item	Specifications
Standards Accommodated	1000 BASE T
Connector Type	RJ45

NET1, NET2 Specifications

Table 25.2 Technical Specifications — NET1

Item	Specifications
Standards Accommodated	1000 BASE T
Connector Type	RJ45

USB Port Specifications

Table 25.3 Technical Specifications — USB Port

Item	Specifications
Connector Type	USB3.0

Environment

Table 25.4 Technical Specifications — Environment

Item	Specifications
Maximum ambient temperature	40°C (104°F)

Power

Table 25.5 Technical Specifications — Power

Item	Specifications
Power supply	350W per power supply
Maximum Power Consumption	700W

Service Information

Routine maintenance to this Ross product is not required. In the event of problems with your product, the following basic troubleshooting checklist may help identify the source of the problem. If the Radio and Streaming Audio Processor (RSAP) still does not appear to be working properly after checking all possible causes, please contact the Technical Support department at the numbers listed under the "**Contacting Ross Video Technical Support**" on page 14.

1. **Visual Review** — Performing a quick visual check may reveal many problems, such as connectors not properly seated or loose cables. Check the RSAP, and any associated peripheral equipment for signs of trouble.
2. **Power Check —** Inspect the power indicator LED on the chassis for the presence of power. If the power LED is not illuminated, verify that the power cable is connected to a power source and that power is available at the power main. If the power LED is still not illuminated, replace the power supply with one that is verified to work.
3. **Input Signal Status** — Verify that source equipment is operating correctly and that a valid signal is being supplied.
4. **Output Signal Path** — Verify that destination equipment is operating correctly and receiving a valid signal.
5. **Unit Exchange** — Exchanging a suspect unit with a unit that is known to be working correctly is an efficient method for localizing problems to individual units.

Warranty and Repair Policy

The RSAP is warranted to be free of any defect with respect to performance, quality, reliability, and workmanship for a period of THREE (3) years from the date of shipment from our factory. In the event that your RSAP proves to be defective in any way during this warranty period, Ross Video Limited reserves the right to repair or replace this piece of equipment with a unit of equal or superior performance characteristics.

Should you find that this RSAP has failed after your warranty period has expired, we will repair your defective product should suitable replacement components be available. You, the owner, will bear any labor and/or part costs incurred in the repair or refurbishment of said equipment beyond the THREE (3) year warranty period.

In no event shall Ross Video Limited be liable for direct, indirect, special, incidental, or consequential damages (including loss of profits) incurred by the use of this product. Implied warranties are expressly limited to the duration of this warranty.

This User Manual provides all pertinent information for the safe installation and operation of your RSAP. Ross Video policy dictates that all repairs to the RSAP are to be conducted only by an authorized Ross Video Limited factory representative. Therefore, any unauthorized attempt to repair this product, by anyone other than an authorized Ross Video Limited factory representative, will automatically void the warranty. Please contact Ross Video Technical Support for more information.

In Case of Problems

Should any problem arise with your RSAP, please contact the Ross Video Technical Support Department. (Contact information is supplied at the end of this publication.)

A Return Material Authorization number (RMA) will be issued to you, as well as specific shipping instructions, should you wish our factory to repair your RSAP. If required, a temporary replacement RSAP will be made available at a nominal charge. Any shipping costs incurred will be the responsibility of you, the customer. All products shipped to you from Ross Video Limited will be shipped collect.

The Ross Video Technical Support Department will continue to provide advice on any product manufactured by Ross Video Limited, beyond the warranty period without charge, for the life of the equipment.

Glossary

The following terms are used throughout this guide:

5.1 — refers to 5.1 surround sound.

Basic Tree View — refers to the area located to the far left of the DashBoard window. This area displays devices in a tree structure. When you launch DashBoard, all openGear Connect compatible devices on the same network are auto-detected by default.

DashBoard — refers to the DashBoard Control System.

DHCP — refers to Dynamic Host Configuration Protocol.

NTP — refers to Network Time Protocol.

Operator and **User** — refer to the person who uses the RSAP.

RSAP — refers to the Radio and Streaming Audio Processor.