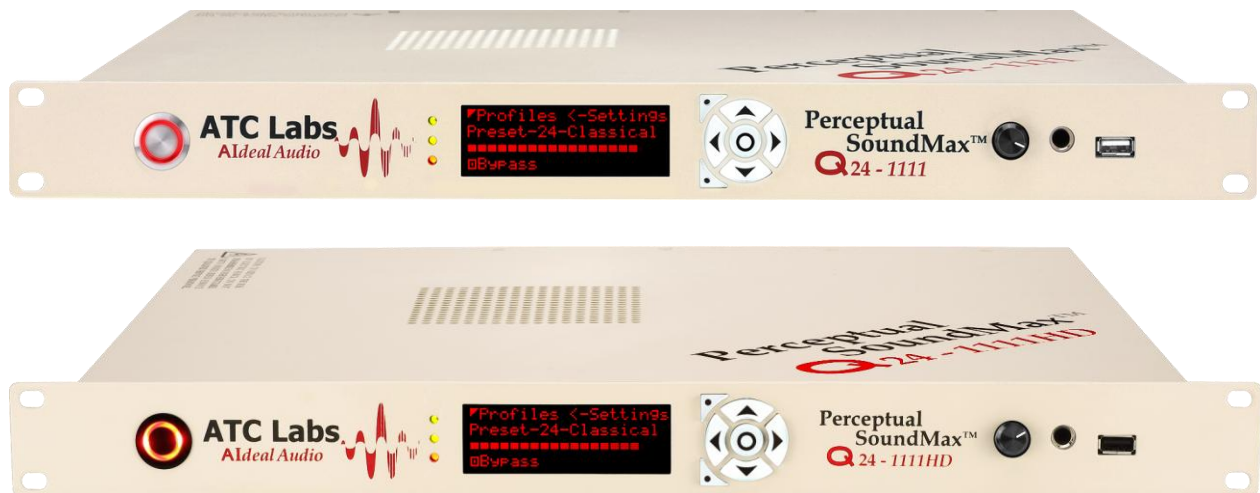


ATC Labs
AIdeal Audio



Perceptual SoundMax™

Models Q24-1111 & Q24-1111HD

24-band FM Audio Broadcast Processors

AI-Driven audio processing, Streaming and Immersive Audio

Operation Manual

Version 8.1, August 2026

IMPORTANT SAFETY PRECAUTIONS AND WARNINGS



WARNING: TO REDUCE THE RISK OF ELECTRIC SHOCK (1) THIS EQUIPMENT MUST BE EARTHED, (2) USE THE EQUIPMENT ONLY WITH A RACK AS RECOMMENDED BY THE MANUFACTURER, (3) DO NOT REMOVE THE TOP COVER, FRONT PLATE, OR BACK PLATE (CONTACT QUALIFIED SERVICE PERSONNEL).

You must read these safety instructions and warnings before using the Equipment or powering it ON.

GENERAL GUIDELINES

- The safety and operation instructions should be retained for future reference.
- All warnings on the equipment and in the operating instructions must be adhered to.
- The equipment should be situated so that its location or position does not interfere with its proper ventilation
- Read **Section 2.2.1: Environmental Considerations** and **Section 2.2.2: Use Precautions** very carefully before proceeding with installation or powering it ON.
- Care should be taken so that objects do not fall and liquids are not spilled into the enclosure through openings.
- The equipment should be connected to a **power supply** only of the type described in the operating instructions or as marked on the equipment.

DAMAGE REQUIRING SERVICE

The equipment should be serviced by qualified service personnel when:

- The power supply cord or the plug has been damaged;
- Objects have fallen, or liquid has been spilled into the equipment;
- The equipment has been exposed to rain;
- The equipment does not appear to operate normally or exhibits a marked change in performance; or
- The equipment has been dropped, or the enclosure damaged.

SERVICING

The user should not attempt to service the equipment beyond that described in the operating instructions. All other servicing should be referred to qualified service personnel. Service contact information is listed on the enclosed Warranty Card.

This Product Adheres to Expected Industrial Safety Standards for the Equipment of its Class.

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Section 1 Product Overview

Perceptual SoundMax™ Q24-1111 series is a **flagship FM & AM DSP-based audio processor** featuring AI-driven audio analysis (for HD series only), engineered for **FM broadcasts, AM broadcasts, and streaming simulcast** applications, with the Q24-1111HD offering broadcasts with Digital and HD Radio and single channel Streaming Simulcast. Built on powerful, state-of-the-art signal-processing technologies, it delivers advanced 24-band processing using an extremely detailed and accurate time-frequency representation of the audio signal. Leveraging cutting-edge innovations—such as filter banks, psychoacoustic modelling, harmonic analysis, advanced AI algorithms-based sound-quality enhancements, and ATC Labs’ deep technical expertise—we have developed a suite of novel & intelligent signal-conditioning tools that maximize the performance of the 24-band dynamics engine and elevate it with new enhancement technologies. The result is a unique, pleasing, and highly optimized broadcast sound that emphasizes presence and perceived loudness while improving long-term listener comfort. In simple terms, this enables broadcasters to create breathtakingly detailed, consistent, and balanced audio that is loud and punchy without introducing listener fatigue.

Three-in-One Integration:

The Perceptual SoundMax™ Q24-1111 series consolidates three essential broadcast processing and encoding technologies into a single hardware platform, reducing cost, rack space, and power consumption:

1. **24-Band Multiband Audio Processing:** Optimized for FM, AM, and Digital Radio channel (digital radio channel available in Model Q24-1111HD)
2. **Scalable Encoding with Streaming Simulcast**– High quality stream encoder along with scalable multi-rate HLS chunk encoding and integrated distribution server and CDN push/pull functionalities.
3. **Audio Sweetening Suite, Broadcast-Grade Noise Reduction, and Spatial Audio:** A comprehensive toolset that includes a unique dynamic listening fatigue reduction processor, along with broadcast grade noise reduction algorithm, and SpatialMax™ spatial audio enhancement.

Key Features & Capabilities

- **Aideal Audio™ Processing:** Patented 24-band AI-enhanced signal conditioning & processing platform for unmatched clarity. Multiple ML deep networks work in tandem to analyze changing characteristics of audio and gently adjust the processing while remaining true to the selected processing profile.
- **Streaming Excellence:** Integrated HE/LC-AAC (V1/V2) encoding, optimized with latest ATC Labs Psychoacoustics, and HLS encapsulation for seamless internet distribution.
- **Advanced Control:** Intuitive on-device interface paired with robust PC-based SoundMaxManager™ remote control software for network-wide tuning. It features an intuitive

screen interface for direct on-device control and is complemented by advanced PC-based remote-control software that provides comprehensive monitoring, adjustment, and tuning capabilities over a network.

- **Operational Reliability:** Engineered with premium components for 24/7 continuous operation and minimal maintenance.
- **Preloaded Versatility:** Includes a wide range of genre-specific presets with support for unlimited custom user profiles.

Technical Specifications Highlight

Feature	Details
Analog/Digital I/O	Main Path AES Digital and Balanced Analog (XLR)
FM Outputs & Inputs	2 x FM-MPX Analog Outputs (BNC)
Digital Radio (Q24-1111HD Only)	Digital Output (XLR/AES) – HD1 Diversity Delay between Main and HD1 outputs
Encoding	HE/LC-AAC V1/V2: 32-256kbps, MP3, Scalable HLS
Streaming, AOIP, and Networking	Configurable Audio-Over-IP (AOIP) input/output – multicast/unicast Streaming Simulcast: 1 channel. Distribution server & CDN connectivity Dual Ethernet (RJ-45): Control/Stream and AoIP (for Model Q24-1111HD only)
Compliance	Internal Digital RDS Encoder & Nielsen™ Watermarking
Additional Tools	Dynamic Listening Fatigue Reduction (DLFR), Wide-band broadcast noise reduction, and much more

The ATC Labs Pedigree

Founded by former **Bell Labs** researchers, ATC Labs brings decades of expertise in signal processing and acoustics to the broadcast industry. Our core technology is trusted by major FM Broadcasters, Satellite Radio, and Internet Radio providers worldwide, serving over 20 million listeners with unmatched audio consistency and detail.

Note: Below all features apply to both Q24-1111 and Q24-1111HD, unless noted otherwise.

Section 2 Installation And Quick Start Guide

2.1 Supplied Accessories

Package Contents

Your Perceptual SoundMax™ Model Q24-1111 shipment includes the following items:

1. Perceptual SoundMax™ Audio Processor for Digital and FM (Model Q24-1111) hardware unit
2. Power cord
3. Rack-mounting kit with 10-32 x ½" screws and washers
4. Ethernet cable
5. SoundMax™ Manager software installation CD-ROM
6. Operation manual
7. Warranty certificate
8. Test report

Important:

- **Inspect Upon Delivery.**
- **If any items are missing or damaged, contact your sales representative immediately.**
- **Retain Packing Materials.**

Save all original packing materials. Should you need to ship the Q24-1111 for servicing or relocation, the original carton and packing materials provide optimal protection, as they have been specifically designed to safeguard the unit during transport.

Product Registration

Complete and return the registration card to ATC Labs (address provided on the card). Registration enables us to:

- Inform you of new applications and performance improvements
- Notify you of software updates and service aids
- Respond promptly to warranty claims without requiring additional proof of purchase

Please complete and return your registration card today. ATC Labs does not sell or share customer information with third parties.

2.2 Rack Mounting the Unit

Use the supplied rack-mounting kit to install the unit in a standard 19-inch equipment rack. Model Q24-1111 or Model Q24-1111HD requires 1U (1.75 inches) of standard rack space. Ensure a solid ground connection between the rack and the chassis for proper safety and performance. For optimal airflow and heat dissipation, it is recommended to leave 1U of space above and below the unit, and at least 10 cm (4 inches) of clearance at the rear.

2.2.1 Environmental Considerations

- **Ventilation:** The equipment must be positioned to allow proper airflow. Do not place it on soft surfaces such as padded tables or any location that may block ventilation openings. Avoid installing it in enclosed spaces that restrict airflow around the unit.
- **Moisture & Water:** Do not use the equipment near source of moisture. Exposure to moisture increases the risk of damage and electrical hazards.
- **Heat:** Keep the equipment away from heat sources such as radiators, heat registers, and other devices (including amplifiers & high power transmitters) that generate heat. Excess heat can impair performance and reduce equipment life.
- **Power source:** Connect the equipment only to a power supply type that matches the specifications defined in the operating instructions or indicated on the unit's back panel.
- **Grounding or Polarization:** Ensure that the unit is properly grounded & grounding and polarization features are not tampered with or defeated. Proper grounding is essential for safe operation and to prevent electrical hazards and malfunctions due to electrostatics.

2.2.2 Use Precautions

- **Power-Cord Protection:** Power supply cords should be routed so they are not walked on or pinched by objects placed on or near them. Pay special attention to the cord at plugs, and the point where it exits the equipment.
- **Cleaning:** Clean the equipment only as recommended by the manufacturer.
- **Non-Use Periods:** Unplug the equipment from the power outlet when it will remain unused for a long period.
- **Object and Liquid Entry:** Ensure that objects do not fall onto/into the equipment enclosure.
- **General:** The audio processor is designed for durability and can operate reliably in most environments. However, to ensure long operating life, the following conditions are recommended:
 - Avoid placing the unit on unstable surfaces subject to vibration or resonance.
 - Ambient temperature should not exceed 50°C (122°F) during operation. Operating the device in cooler environments with proper air-conditioning will extend its lifespan.
- **Power Ground:** Ground the processor chassis using the third wire in the power cord. The equipment must always be connected to a proper earth ground. Floating the chassis from ground creates a potential safety hazard and may result in malfunction.

- **Circuit Ground:** To maintain equal potential across all connected equipment, ensure that circuit (audio) grounds are properly linked.
- **High RF Field Considerations:** In high-RF environments, grounding is typically achieved through the equipment rack. The rack should be connected to a solid earth ground using a wide copper strap—standard wire is ineffective at VHF frequencies due to its self-inductance.

2.3 Rear/Back Panel I/O and Power Connections

Figures 2.3.1 and 2.3.2 below show the Rear/Back Panel for both Models Q24-1111/1111HD.

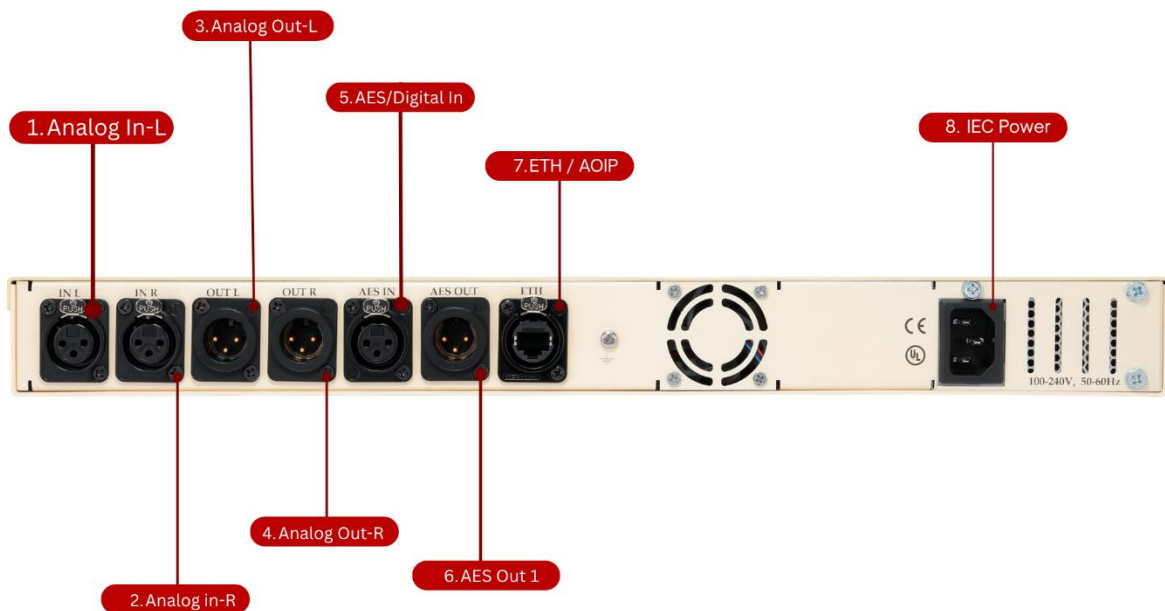


Figure 2.3.1: Model Q24-1111 REAR/BACK Panel

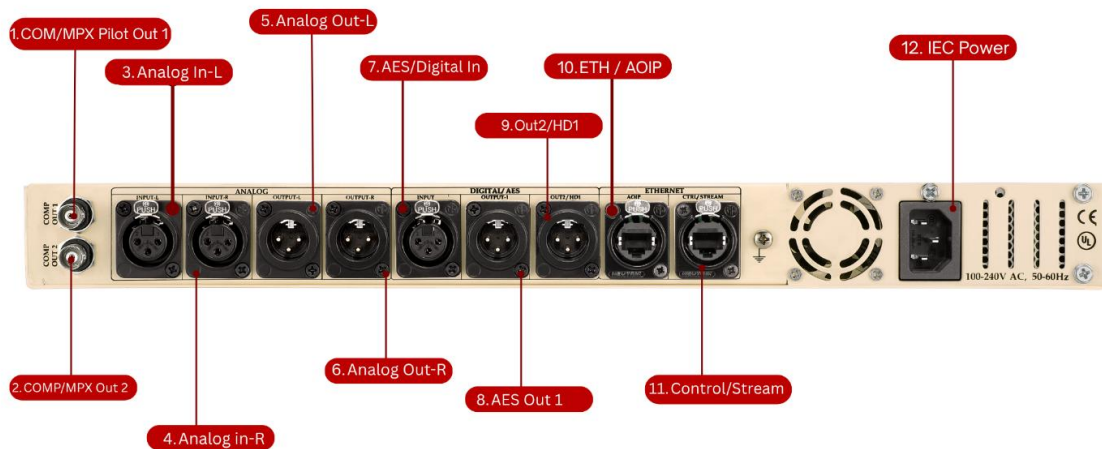


Figure 2.3.2: Model Q24-1111HD REAR/BACK Panel

All the I/O and power connectors for are available at the rear panel (except headphone monitoring output which is available on the front panel as described below). Make the power and I/O connections as described below.

Connecting the Power Supply: The audio processor can operate at 110-250V, 50Hz/60Hz voltage rating. Verify your line settings before connecting to the unit. Use the supplied power cable or another cable of equivalent rating.

2.3.1 Back/Rear Panel details and Input and Output Connections:

Composite / Baseband Connectors (BNC) (for Q24-1111HD only)

- **Composite Baseband Output 1 (COMP/MPX/PILOT)** – BNC female output providing the primary composite multiplex (MPX) signal including stereo pilot tone. Connect this output to the FM exciter's composite input.
- **Composite Baseband Output 2 (COMP/MPX)** – BNC female output providing a secondary composite MPX output. This can be used for a backup exciter, monitoring equipment, or a secondary transmission path.

Analog Audio Connectors (XLR)

- **Balanced Analog Input (Right)** - XLR-3F (female) balanced Analog audio input for the right channel. Accepts line-level audio signals.
- **Balanced Analog Input (Left)** – XLR-3F (female) balanced Analog audio input for the left channel. Accepts line-level audio signals.

- **Balanced Analog Out (Right)** – XLR-3M (male) balanced Analog audio output for the right channel. Output level is software-configurable -6 to +24 dBu.
- **Balanced Analog Out (Left)** – XLR-3M (male) balanced Analog audio output for the left channel. Output level is software-configurable -6 to +24 dBu.

Digital AES Connectors (XLR) — Main Path

- **AES/EBU Digital Input** – XLR-F (female) AES3 (AES/EBU) digital audio input. Accepts a stereo AES3 digital audio stream. Supported sample rates: 44.1, 48, 88.2, and 96 kHz.
- **AES/EBU Digital Output** – XLR-M (male) AES3 (AES/EBU) digital audio output. Provides the processed stereo digital audio output. Output sample rate is software-selectable 44.1 kHz, or 48 kHz 88.2, 96kHz for stereo mode.

Digital AES Connectors (XLR) — HD Radio Paths (Q24-1111HD only)

The Q24-1111HD model adds additional AES digital output for independent HD Radio broadcasting:

- **AES / HD1 OUT** (XLR-3M male) — AES3 digital output carrying the processed HD1 channel audio. Can also serve as a secondary main path AES output when HD Radio is not in use.

Note: HD output (HD1) carries an independently processed audio stream with its own profile and tuning.

Network and Data Connectors

- **Ethernet — CONTROL / STREAM (RJ-45):** Primary Ethernet port for remote control access via the SoundMax™ Manager PC software, firmware updates, configuration management, and HLS internet streaming output. Connect to the facility's control / management network using Cat5e or Cat6 cable.
- **Ethernet — AoIP (RJ-45):** Dedicated Ethernet port for AES67 Audio-over-IP streaming (input and output). Should be connected to a PTP-synchronized AoIP network kept isolated from general control traffic for reliable low-latency audio transport. Cat5e or Cat6 cable required.

Note: On the Q24-1111 (non-HD model), a single RJ-45 Ethernet connector handles both control and AoIP functions on the same physical port.

- **USB Connector:** USB Type-A port for firmware updates, configuration backup and restore, and diagnostic log retrieval via USB flash drive.

Power Supply:

- **Power Switch** — Rocker-type main power switch.
- **AC Power Inlet** — IEC C14 connector. Operating voltage: 100–240V AC, 50–60 Hz (auto-ranging universal power supply). Use only the supplied power cord. 3-pin Power Cord Connector (with earthing).

2.3.2 Connecting Analog Audio Equipment:

The left and right analog inputs are on EMI-suppressed XLR-type female connectors on the rear panel. Input impedance is greater than 10k Ω ; balanced and floating. Inputs can accommodate up to +24 dBu (0 dBu = 0.775Vrms). The left and right analog outputs are on EMI-suppressed XLR-type male connectors on the rear panel. Output impedance is 50 Ω ; balanced and floating. They can drive 600 Ω or higher impedances, balanced or unbalanced. The peak output level is adjustable -6 dBu to +24 dBu. Level control of the analog inputs and outputs is accomplished via front-panel control through System Setup as described in section 3.

2.3.3 Connecting Digital Audio Equipment:

The digital input and outputs conform to the professional AES/EBU standard. They have sample rate converters supporting 44.1, 48, 88.2, and 96 kHz. Input sample rate is auto sensed. Digital input is via one XLR-type female connector; digital outputs via one XLR-type male connectors. Levels are front-panel configurable. Inputs reference level can be tuned in the range of -25 to 0dBFS.

2.3.4 Connecting Composite/Pilot Outputs:

The stereo encoder provides two unbalanced Analog baseband outputs on BNC connectors. Each output can be strapped for 0 or 75 Ω source impedance and can drive up to 8V peak-to-peak into 75 Ω . Independent level control is available via the front panel in the COMPOSITE settings screen.

2.4 Front Panel Controls

Figures 2.4.1 below shows the Front Panel Controls for Q24-1111/1111HD

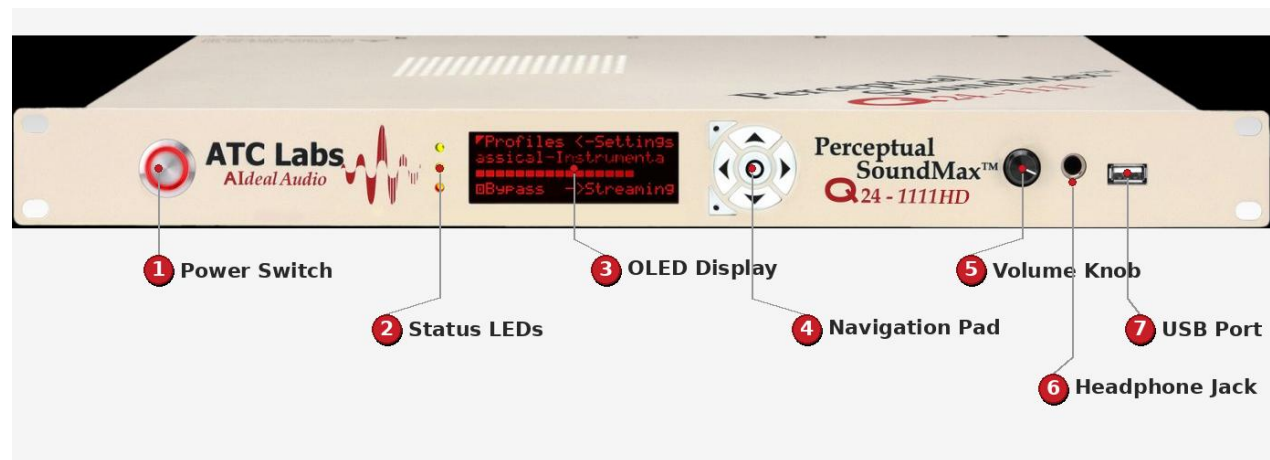


Figure 2.4.1: Model Q24-1111/1111HD FRONT PANEL

1. **Power Switch:** Turns the unit ON or OFF.
2. **Status LEDs:** LEDs display the status.

3. **Alphanumeric LCD Screen:** A front-panel LCD that displays system status and menu options, allowing the user to monitor and control the unit.
4. **Switch:** Directional navigation keypad used to browse menus, adjust settings, and monitor the unit.
5. **Headphone Volume Control:** Rotary knob that adjusts the output level of the front-panel headphone jack for quick, on-the-spot monitoring.
6. **Headphone Audio Output (¼" Jack):** Quarter-inch headphone jack for real-time monitoring of the processor's output, letting the user audition the processed audio directly from the front panel.
7. **USB Port:** External USB port for connecting media and supporting future firmware upgrades.

2.5 Remote Computer Monitoring Software Installation (may be skipped during initial setup)

Model Q24-1111 hardware is supplied with SoundMaxManager™ PC software for remote computer interface. This Network Management System (NMS) software is provided with CD-ROM or USB pen drive and can also be downloaded from [ATC Labs weblink](#).

Features

SoundMaxManager™ allows for elaborate configuration, control, monitoring, and tuning of multiple Q24-1111 processors connected to a network. The presence of one or more Perceptual SoundMax™ Model Q24-1111 on the network is automatically sensed by the SoundMaxManager™ running on any computer connected to that network.

Functionality

By using this software, users have complete flexibility to configure and monitor the Q24-1111 processors. The supported functionality includes:

- Configure parameters specific to processors.
- Preset selection.
- Detailed parameter tuning.
- Custom processing profile creation.
- Streaming Configuration.
- And much more.

Monitoring Capabilities

Monitoring capabilities include:

- Global view of processor health.
- VU meters tied to the input/output of each processor.

2.5.1 Installing SoundMaxManager™

Minimum system requirements

- **Operating System:** SoundMaxManager™ is currently available for Windows 10, 11 (32/64-bit) operating systems.
- **LAN Connection:** The Manager communicates with the Station through a network connection.
- **PC:** The SoundMaxManager™ can run on any lightweight PC. We recommend at least a single core Pentium with a clock speed of 2.2 GHz (or above).

Instructions for installation

Setting up the SoundMaxManager™

- Run the SoundMaxManager™ setup on a PC and follow the instructions
- After the SoundMaxManager™ installation, the user interface screen (UI) shown in Figure 2.5.1 below will appear.
- SoundMaxManager™ will auto-detect the “Model Q24-1111 rack mount encoder/decoder” as Live Station. By clicking on the station from this list, you can get a detailed view (as shown in Figures 2.5.2 and 2.5.3 below) and options to configure that particular station.

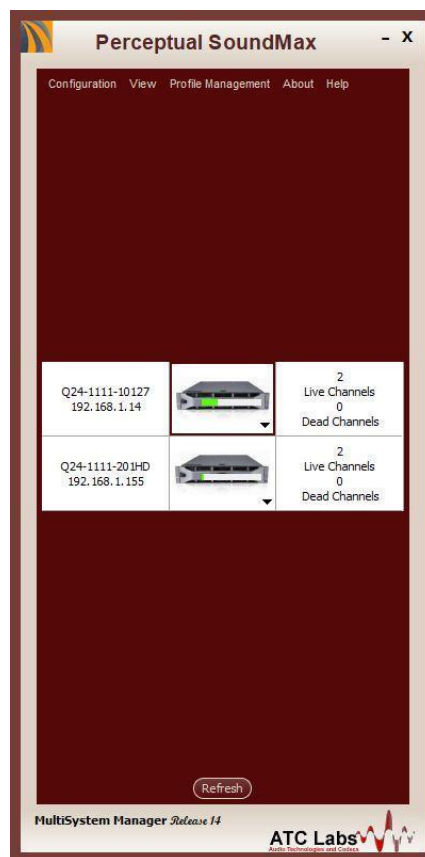


Figure 2.5.1: SoundMaxManager™ UI Screen

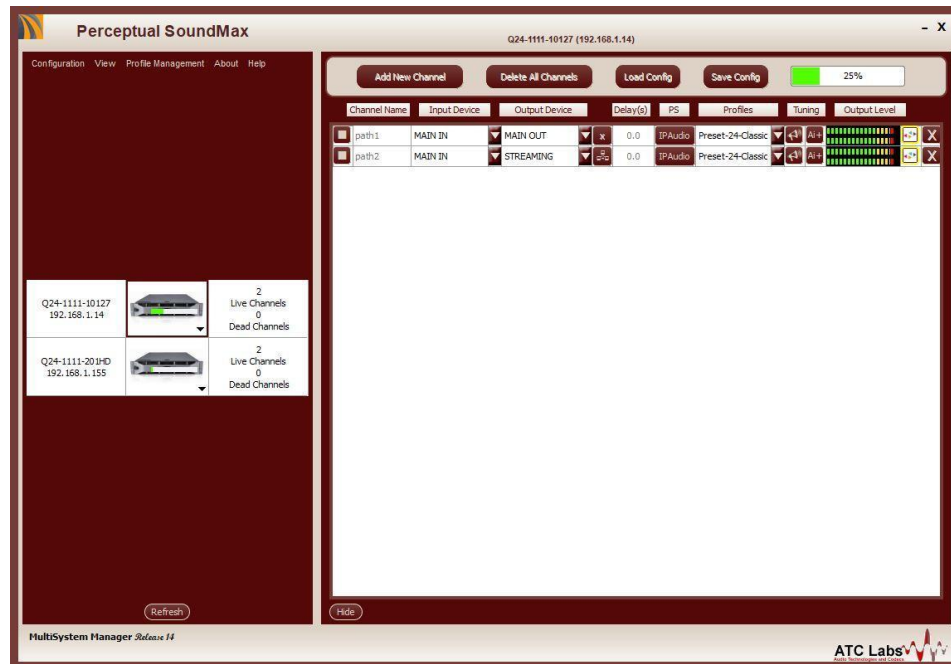


Figure 2.5.2: SoundMaxManager™ Detailed View (1111)

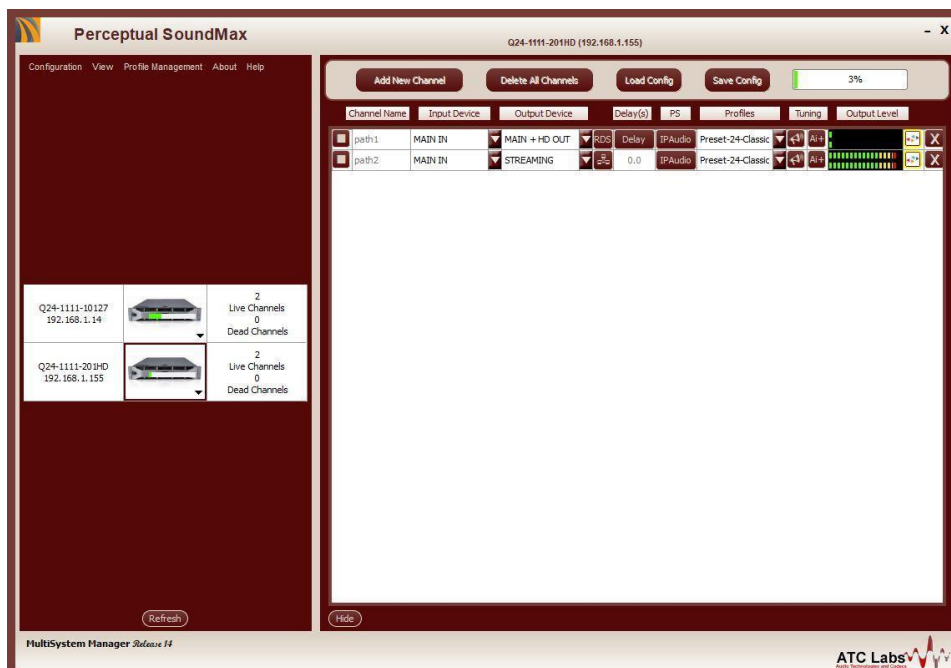


Figure 2.5.3: SoundMaxManager™ Detailed View (1111HD)

This view provides the ability to configure the following:

- It allows detailed configuration of encoding / decoding such as bitrates selection, codec selection, and streaming parameters. Fine-tuning of encoding and processing parameters can also be performed independently for each channel along with the configuration for distribution.

- It provides an option to tune the audio quality of any channel based on the requirement. Clicking on the tune button will change your view to a tuning screen showing all the tuneable modules and an option to save custom profiles and load saved profiles.

For a detailed description on SoundMaxManager™ controls refer to Section 4

Note: If SoundMaxManager™ is controlling the encoder, then the front-panel interface will be disabled.

2.6 Quick Start Guide

⚠ WARNING: Do not skip to this section without first familiarizing yourself with the material in Sections 2.1 through 2.5. Improper installation may result in equipment damage or unsafe operation.

Follow the steps below to power up and begin using the Model Q24-1111 and Model Q24-1111HD audio processors for the first time.

Step 1 — Verify Installation

Confirm that all installation steps in **Sections 2.1 through 2.5** have been completed. This includes unpacking, rack mounting, audio I/O connections, power connection, and front panel familiarization.

Step 2 — Power On the Unit

- Press the front-panel power switch to turn the unit ON.
- The LED ring around the ON/OFF switch will illuminate, confirming power is applied.
- Allow several seconds (up to 2-3 minutes) for the unit to complete its boot sequence. During this time, various start-up messages will be displayed on the front-panel screen.
- Once fully powered up, the front panel will display the Selection Screen, which should appear similar to Figure 2.6.1 below.



Figure 2.6.1: HOME Screen

The main screen displays the following items:

- **Current Profile:** Shows the currently active processing profile. In Bypass mode, the display reads: “OnAIR: Path1 – Processing Bypassed”, In PROCESSED mode, the active preset or custom profile is shown (e.g. “OnAIR: Path1 – Preset-24 BASS-TREBLE”).
- **Output VU Meter:** Displays the real-time output level of the processed audio.
- **Indications:** The labels indicate the navigation keys used to open screens or toggle functions: Profiles (, Settings (, Bypass (, and Streaming ().

Step 3 — Navigate the Selection Screen

Controls on the Selection Screen (and all other screens) can be operated navigation keys:

- **Select Key** (): On the main screen, toggles the BYPASS function to switch between the processed and unprocessed output. On all other screens, it selects or applies the highlighted option from the list.
- **Back Key** (): Returns to the main menu from any sub-screen.
- **Left Key** ()/**Right Key** (): On the main screen, opens the Settings menu (which includes Network Settings, Device Settings, and Pre-emphasis Settings). On other screens, it decreases/increases the value of the selected control.
- **Up Key** ()/**Down Key** (): Moves the cursor up/down on the front-panel screen.
- **Profile key** (): To get the list of all factory PRESETS and CUSTOM profiles.

Step 4 — Processing Profiles

Press the **Profile key** () to access available Processing Profiles (presets and custom profiles). Profiles are categorized by audio type (genre) and processing bands.

- Use **Up/Down keys** to browse profiles.
- Press **Select/OK key** to apply a profile.

Step 5 — Bypass/UnBypass

Press the **Select/OK** () key to toggle Bypass mode. When enabled, audio processing is disabled and the input signal passes directly to the output. Press again to resume normal processing.

Step 6 — General Input Output Settings:

Press the **Left key** () to access the Settings menu, which includes:

- Network Settings
- Device Settings
 - Input Selection.
 - Input Level.
 - Output Level.
 - Output Trim.
- Pre-emphasis Settings

Note: Use the **Up/Down keys** to navigate to the desired option, press the **Select/OK key** to select it, and use the **Left/Right keys** to adjust the levels

Input Selection:

Allows selection between Analog and Digital (AES/EBU) inputs. Ensure the appropriate cable is connected (XLR for Analog, AES/EBU for digital) before changing the selection.

Input Level (Analog Only):

Defines the maximum input level the unit can process without distortion.

Example: If set to 10 dBu, a 10 dBu input corresponds to 0 dBFS internally.

Supports up to 24 dBu input with proper configuration.

Output Level (Analog Only):

Sets the Analog output level. Adjustable range: -6 dBu to +24 dBu.

Output Trim:

Acts as a global output attenuator for both Analog and digital outputs. Adjustable range: 0 dBFS to -90 dBFS.

Important Notes:

- In FM broadcasting, this controls the peak modulation index and should be adjusted after selecting processing profiles.
- For Nielsen™ Watermark Encoding, set Output Trim to -3 dBFS or lower to prevent clipping and ensure proper watermark detection.

Output pre-emphasis:

Used in FM broadcast applications to apply pre-emphasis before transmission. Set according to the regional broadcasting standard.

Step 7 — Default I/O Configuration

The Model Q24-1111 ships with the following default I/O configuration:

- **Input:** Analog (XLR balanced)
- **Output:** Analog (XLR balanced) and Digital (AES/EBU) — both active simultaneously

For the initial quick start, audio connected to the Analog input will be processed and routed to both the Analog and digital outputs. No configuration changes are required to verify basic operation.

Note: If digital input is required instead of Analog, navigate to **Settings → Device Settings** and change the input selection accordingly. Refer to **Section 3.2 — Input/Output Configuration** to change the input source. The system will retain the last user settings on subsequent power-ups.

Step 8 — Verify Audio

With audio connected to the Analog input, confirm that:

- The input VU meters on the Main Screen show signal activity.
- Audio is present at both the Analog and digital outputs.

The unit is now operational and ready for on-air use with the default factory profile. To customize processing settings, profiles, or I/O routing, proceed to **Section 3 — Screen Controls**.

Note: If a user has changed any of the settings and shut down the unit, then Unit will start with last configuration.

Section 3 **Model Q24-1111/Q24-1111HD**

Screen Controls

Perceptual SoundMax™ Model Q24-1111 and Q24-1111HD audio processors feature an intuitive front-panel interface built around a high-contrast alphanumeric LCD display and a dedicated navigation keypad, designed for easy and efficient user control. Through this interface, users can adjust I/O levels, switch between digital and analog inputs, configure composite settings, change processing profiles, and access various operational parameters.

This section provides a detailed overview of the key navigation controls and their functionalities.

3.1 Home Screen

The main screen (Figure 3.1.1) has the following items:

- **Current Profile:** Shows the currently active processing profile, In Bypass mode, the display reads: “OnAIR: Path1 – Processing Bypassed”, In PROCESSED mode, the active preset or custom profile is shown (e.g. “OnAIR: Path1 – Preset-24 BASS-TREBLE”).
- **Output VU Meter:** Displays the real-time output level of the processed audio.
- **Indications:** The labels indicate the navigation keys used to open screens or toggle functions: Profiles (), Settings (), Bypass (), and Streaming ().



Figure 3.1.1: Main Screen

3.2 Settings

From the **Home Screen**, press the **Left soft key** to access the **Settings** menu (Figure 3.2.1).

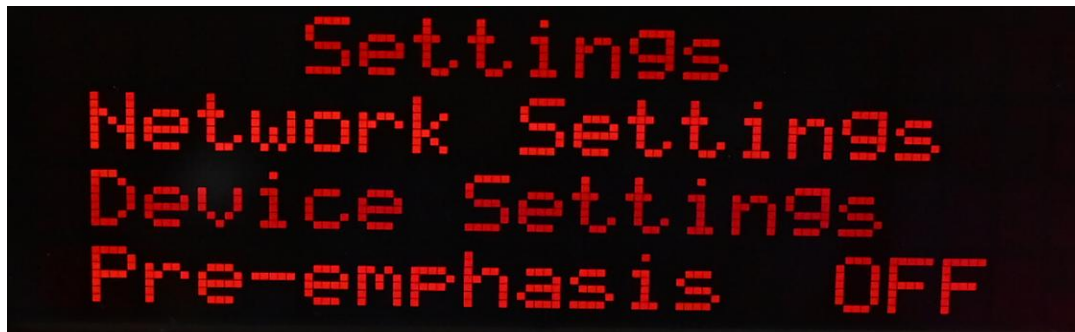


Figure 3.2.1: Settings Menu Screen

The Settings menu provides access to the following configuration options:

- **Network Settings** — Configure network connectivity parameters for the device.
- **Device Settings** — Manage input and output signal levels, including the following sub-options:
 - Input Selection
 - Input Nominal Level
 - Input Headroom
 - Output Level
 - Prescale
 - Output Sample Rate
- **Pre-Emphasis Settings** — Configure the pre-emphasis curve applied to Analog and digital outputs.
- **Composite Settings (Q24-1111HD Only)** — Configure the Composite Settings with following sub options:
 - Pilot Level,
 - Output1 Level
 - Output2 Level
 - Out1 Type
 - Out2 Type
 - Pre-Emphasis
 - RDS

Use the **Up/Down arrow keys** to navigate between menu items and press the **circular (Enter) key** to access the desired setting.

3.2.1 Network Settings

The **Network Settings** menu provides access to network interface configuration options for the device. Upon entering this menu, the screen displays the heading "**Network Interfaces**" with the following option available:



Figure 3.2.2: Network Interfaces Screen

- **eth0:**

Press the **circular (Enter) key** while the cursor is positioned on **eth0** to access the Ethernet configuration screen. The following two IP assignment options will be presented:



Figure 3.2.3: Ethernet Configuration Screen

- **DHCP** Select this option if a DHCP server is available on your network (typically provided by a router, modem, or dedicated server). When selected, the device will automatically obtain an IP address from the DHCP server, requiring no manual network configuration.
- **Assign Fixed IP** Select this option if your network uses static IP addressing or if no DHCP server is available. Assigning a fixed IP address requires manual entry of the following four network parameters:
 - IP Address
 - Subnet Mask
 - Default Gateway
 - DNS Server



Figure 3.2.4: Assign Fixed IP Screen

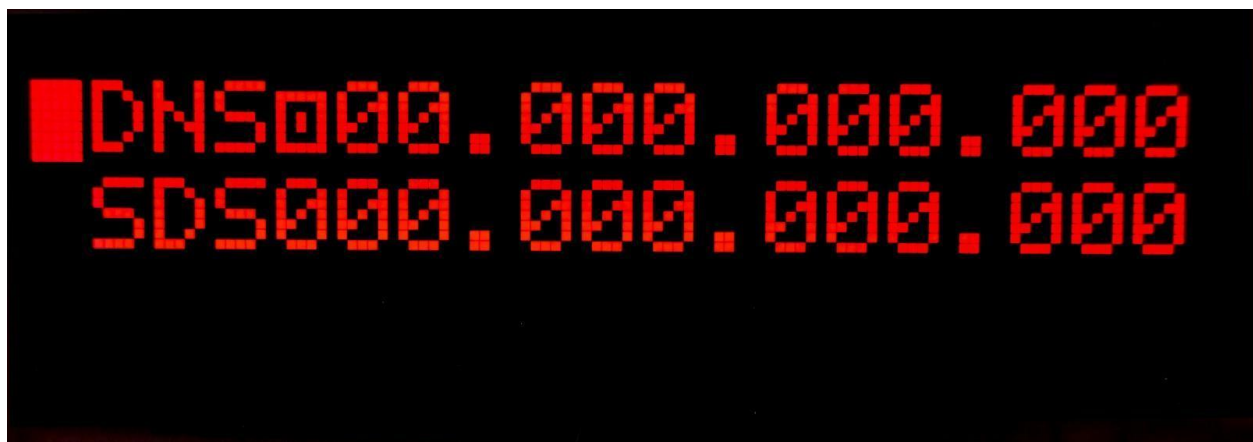


Figure 3.2.5: Assign Fixed IP Screen

Configuring a Static IP Address

Upon selecting **Assign Fixed IP**, the following sub-options will be available. For each entry, use the **circular (Enter) key** to access the input screen and the **on-screen keyboard** to enter the required value. Press the **Down arrow key (↓ Save)** to confirm and save each entry, or press the **Up arrow key (↑ Back)** to return to the previous screen without saving.

- **Set IP Address:** Enter the static IP address to be assigned to the device.
- **Set Subnet Mask:** Enter the subnet mask corresponding to your network configuration.
- **Set Gateway:** Enter the default gateway address, typically the IP address of your router or network gateway device.
- **Set DNS:** Enter the primary DNS server address for domain name resolution.

3.2.2 Device Settings

The **Device Settings** menu provides access to input and output level configuration options. Use the **Up/Down arrow keys** to navigate between settings and press the **circular (Enter) key** to access each configuration screen.

The following options are available under this menu:

- Input Selection
- Input Nominal Level
- Input Headroom
- Output Level
- Prescale
- Output Sample Rate

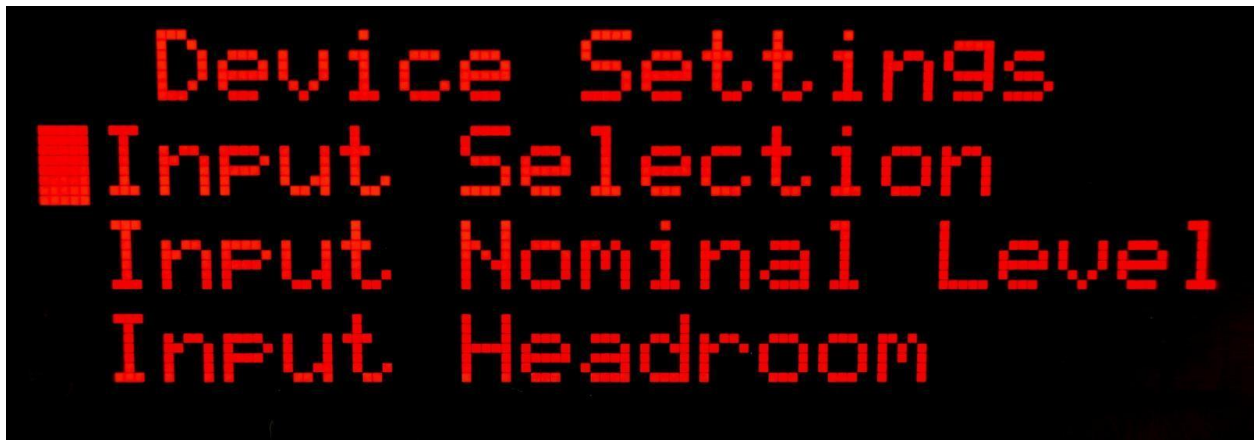


Figure 3.2.6: Device Settings Menu Screen



Figure 3.2.7: Device Settings Menu Second Screen

INPUT SELECTION: This setting allows the user to select between **Analog** and **Digital** input sources. Before changing the input selection, ensure the appropriate cable is connected:

- **Analog input:** XLR cable
- **Digital input:** AES/EBU cable



Figure 3.2.8: Input Selection Screen

Once the cable is connected, navigate to this menu and select the corresponding input type.

Input Level Configuration (Analog Input Only):

The audio processor provides two independent input level controls — Input Nominal Level and Input Headroom — that together determine how the analog input signal is mapped to the internal digital domain. Configuring these settings correctly ensures optimum signal-to-noise performance while preserving sufficient margin for transient peaks, preventing both under-utilisation of the ADC range and clipping at the input stage.

Both parameters are accessible from the front panel via:

Device Settings → Input Selection → Input Nominal Level / Input Headroom

The current value of each parameter is shown on the alphanumeric LCD screen along with a graphical bar indicator, and is adjusted using the navigation pad (▲ / ▼ to change value, ◀ / ▶ to navigate, and the centre button to confirm).

Input Nominal Level

The Input Nominal Level sets the reference operating point of the analog input — that is, the level at which the incoming audio signal is expected to sit on average during normal programme content. This reference defines the 0 dBFS-equivalent operating line within the Q24-1111 signal chain and is the anchor against which the Input Headroom is added.

This setting should be matched to the nominal output level of the upstream studio equipment (mixing console, audio router, STL receiver, etc.) so that the processor receives signal at its intended operating point. A correctly set nominal level ensures that programme audio occupies the optimum portion of the ADC dynamic range, maximising signal-to-noise ratio without unnecessarily compressing the available headroom.



Figure 3.2.9: Input Nominal Level



Figure 3.2.10: Input Headroom Level

Range: -4 dBu to +13 dBu, adjustable in 1 dBu steps

Default: +4 dBu (suitable for most professional broadcast environments)

If the Input Nominal Level is set lower than the actual programme level supplied by the source, the input will be at risk of clipping during normal operation, since less of the ADC range is reserved above the reference. If set higher than the actual source level, the processor will be under-driven, reducing signal-to-noise ratio and limiting the effectiveness of downstream gain staging.

Input Headroom

The Input Headroom defines the additional margin reserved above the Nominal Level to accommodate transient peaks — short-duration signal excursions that exceed the nominal operating level (drum hits, vocal plosives, percussive instrumental content, etc.). The headroom value sets the distance, in dB, between the Nominal Level and the ADC clipping point.

Together with the Nominal Level, the Headroom setting determines the absolute maximum input level (peak ceiling) that the ADC will accept before clipping:

Maximum Input Peak Level (dBu) = Input Nominal Level + Input Headroom

For example: with a Nominal Level of +4 dBu and Headroom of +12 dB, the input will accept peaks up to +16 dBu before the ADC clips.

Range: 0 dB to +14 dB, adjustable in 1 dB steps

Default: +12 dB (standard broadcast headroom)

Insufficient headroom causes input clipping during peaks, introducing audible distortion that cannot be removed by any subsequent processing. Excessive headroom unnecessarily reduces the average level presented to the ADC, lowering the effective signal-to-noise ratio. The objective is to set headroom just slightly greater than the largest expected peak-to-nominal ratio of the incoming programme material.

Output Level (Analog Output Only)

Upon entering the Output menu, the user is presented with the top-level output configuration screen. This screen lists the available output sub-settings. Use the **Up/Down arrow keys** to navigate between the options and press the **circular (Enter) key** to enter the desired setting.



Figure 3.2.11: Output Screen

Analog Output Screen

Selecting the Analog Output option from the Output screen opens the Analog Output screen. This screen displays the current analog output configuration and provides access to the level and mono control settings specific to the analog output path. Use the **Up/Down arrow keys** to navigate and press the **circular (Enter) key** to enter a sub-setting.



Figure 3.2.12: Analog Output Screen

Output Level Screen (Stereo Control)

The Output Level screen displays the current analog output level value along with a bar graph meter indicating the set level within the adjustable range. The output level can be set within the range of **-76 dBu to +24 dBu**.

The dBu value is defined relative to 0.7746 V RMS (equivalent to a 1 V peak-to-peak sine wave) and is calculated as:

$$\text{dBu} = 20 \times \log_{10} \left(\frac{V_{\text{rms}}}{0.7746} \right)$$

This allows the analog output to be precisely matched to the input sensitivity of connected equipment such as FM exciters, STL transmitters, or monitoring devices.

- Press the **Left arrow key** to decrease the output level.
- Press the **Right arrow key** to increase the output level.

Note: It is recommended to set the output level after the processing profile has been selected and tuned, to ensure the final output correctly matches the input requirements of the connected transmission equipment.

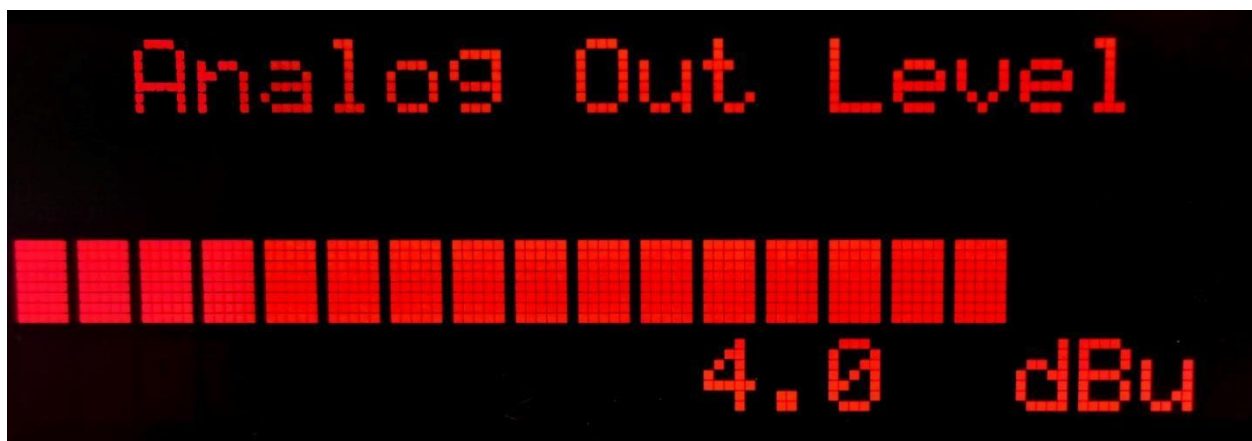


Figure 3.2.13: Output Level Screen

Output Mono Control Screen

The Output Mono Control screen provides options for configuring the mono output behaviour of the analog outputs. This screen is used when operating in mono broadcast mode and allows the user to select which channel or combination of channels is routed to the mono output. Use the **Up/Down arrow keys** to navigate between options and press the **circular (Enter) key** to select.



Figure 3.2.14: Output Mono Control Screen

Output Mono Left-Right Level Control Screen

The Output Mono Level Control screen allows independent level adjustment for the left and right channels of the analog mono output. The current level values and a corresponding bar graph are displayed on screen. Adjusting this setting sets the output level for the left and right channels independent to each other, enabling precise balance control in mono operation.



Figure 3.2.15: Output Mono Left Level Control Screen



Figure 3.2.16: Output Mono Right Level Control Screen

Output Trim

The Output Trim screen displays the current digital trim value along with a bar graph indicating the trim level within the full adjustment range. The Output Trim is a digital attenuator that affects **both analog and digital outputs** simultaneously, applying a global attenuation across all output paths — including analog XLR, digital AES/EBU, and composite baseband — before the final output stage.

The trim level can be adjusted from **0 dBFS** (full scale) down to **–90 dBFS**.

- Press the **Left arrow key** to decrease the peak output level.
- Press the **Right arrow key** to increase the peak output level.

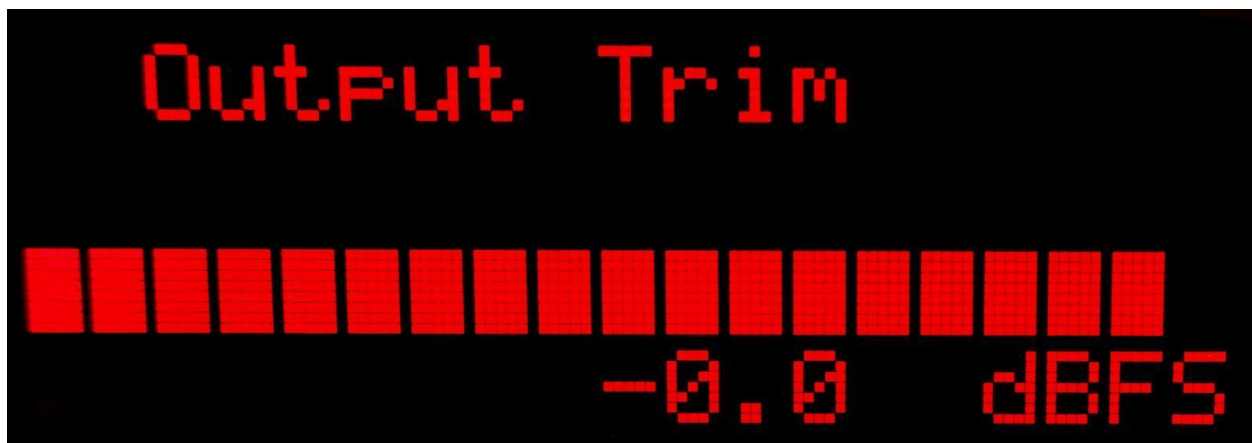


Figure 3.2.17: Output Trim Screen

⚠ Important Notice — Critical Setting

FM Broadcast Applications: The Output Trim directly controls the peak modulation index of the FM transmission. An incorrectly set Output Trim can result in either under-modulation, reducing coverage and audio quality, or over-modulation, which violates regulatory limits and causes distortion. It is strongly recommended to adjust this value only after the appropriate processing profile has been selected and fully tuned.

Nielsen™ Watermark Encoding: When Nielsen™ watermark encoding is active, the Output Trim **must** be set to **–3 dBFS or lower**. Sufficient headroom is required to accommodate the additional signal energy introduced by the watermark embedding process. Failure to provide this headroom will cause clipping in the watermark-embedded output signal, which may result in watermark detection failure at the receiver end and non-compliance with Nielsen™ encoding standards.

HD1 Output Trim (Q24-1111HD Only)

The HD1 Output Trim screen is available exclusively on the **Q24-1111HD** model and displays the current trim value for the HD1 digital output path. A bar graph indicates the current level within the adjustable range. This setting provides an independent digital trim control specifically for the HD1 AES/EBU digital

output, allowing the HD Radio digital audio level to be configured separately from the main analog and composite output levels.

This is necessary because HD Radio transmission systems typically require different input signal levels compared to the analog FM transmission chain. The HD1 Output Trim operates identically to the main Output Trim but applies exclusively to the HD1 digital output. The trim level can be adjusted from **0 dBFS** down to **-90 dBFS**.

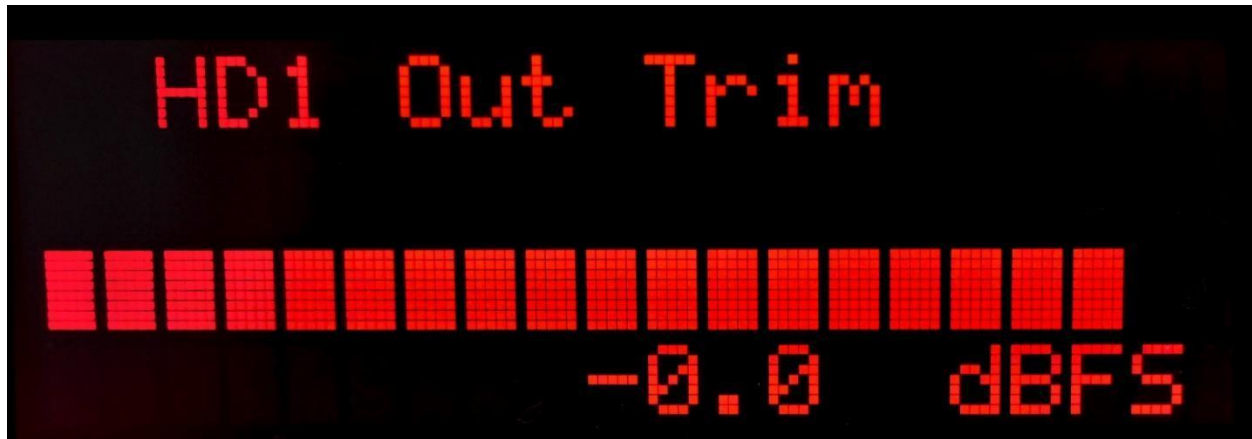


Figure 3.2.18: HD1 Output Trim Screen

- Press the **Left arrow key** to decrease the HD1 output trim level.
- Press the **Right arrow key** to increase the HD1 output trim level.

Note: The HD1 Output Trim does not affect the analog outputs or the main composite baseband output. Configure this setting in accordance with the HD Radio exciter manufacturer's input level specifications to ensure correct loudness alignment between the analog FM and HD Radio programme streams.

Output Pre-Emphasis

The Pre-Emphasis screen displays the currently selected pre-emphasis setting and presents the available options for selection. In FM broadcast applications where the Q24-1111 or Q24-1111HD is responsible for pre-emphasizing the final output signal prior to transmission, this screen allows the user to configure the appropriate pre-emphasis time constant for the target geographical region.

Pre-emphasis applies a high-frequency boost to the audio signal before transmission. The complementary de-emphasis network within the FM receiver restores a flat frequency response, resulting in an improved signal-to-noise ratio for the listener. The correct pre-emphasis standard must be selected to match the broadcast regulations applicable at the transmission site.

The following options are displayed on screen and can be selected using the Up/Down arrow keys:

- **OFF:** Pre-emphasis is disabled. Select this option when the connected FM exciter or transmitter applies pre-emphasis internally, or when pre-emphasis is handled by a downstream device in the signal chain. Enabling this setting alongside exciter-side pre-emphasis will result in double pre-emphasis and audible high-frequency distortion.
- **50 μ s:** Selects a 50 microsecond pre-emphasis time constant. This is the broadcast standard for most regions including Europe, Australia, Asia, and Africa.
- **75 μ s:** Selects a 75 microsecond pre-emphasis time constant. This is the broadcast standard for North America (USA and Canada) and South Korea.

Press the circular (Enter) key to confirm and apply the selected pre-emphasis setting.

⚠ Warning: Selecting an incorrect pre-emphasis standard will result in audible tonal imbalance at the receiver — either excessive brightness or a dull, muffled sound. Always verify the applicable regional broadcast standard before configuring this setting and confirm the result using a calibrated FM receiver or spectrum analyser before going on-air.



Figure 3.2.19: Pre-Emphasis Screen

Composite FM Configuration (HD Only)

This Model is equipped with two composite baseband outputs, each featuring independent user-controlled levels and dedicated connectors. This design provides flexibility for system integration, allowing precise adjustment of each output to meet specific installation or operational requirements.

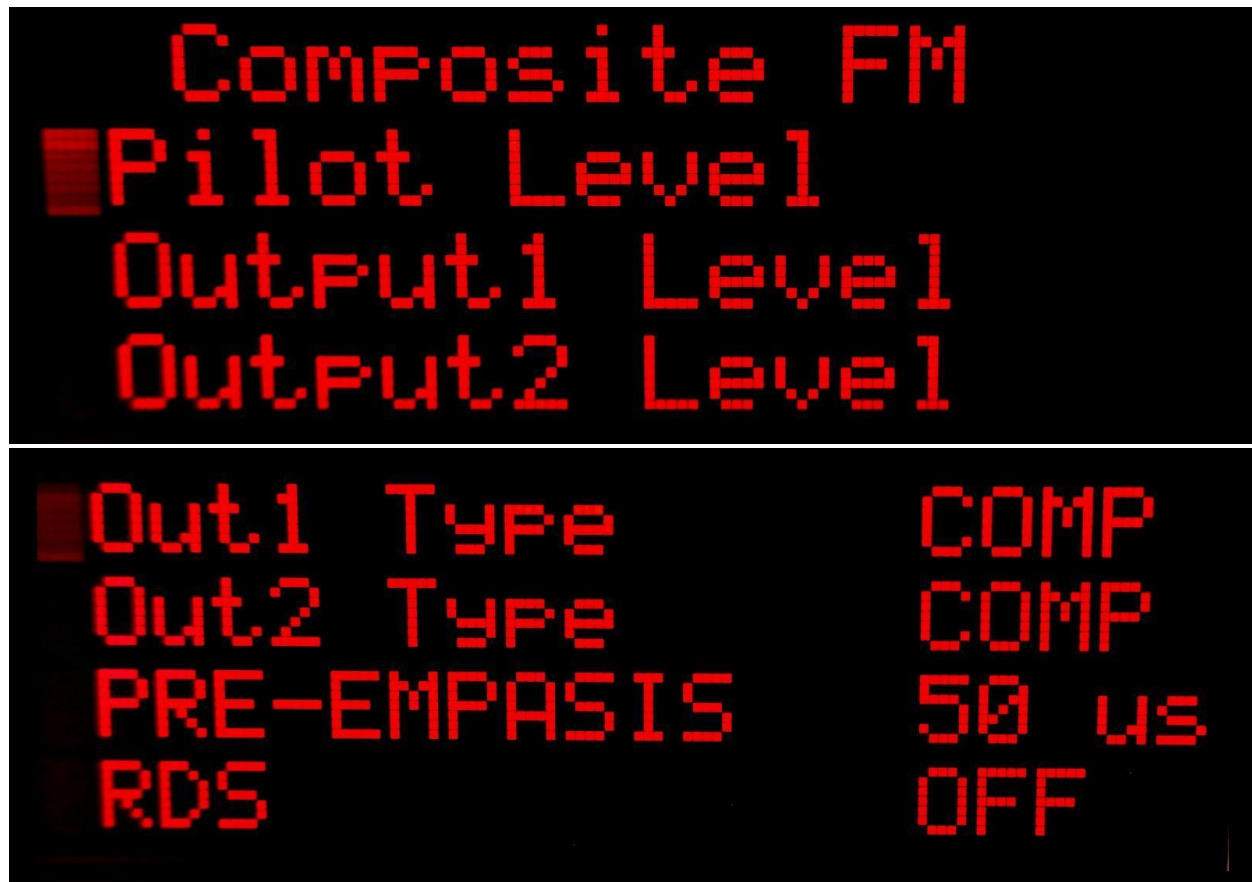


Figure 3.2.20: Composite FM Configuration Screen

All Composite FM parameters are accessed through the front-panel alphanumeric LCD by navigating to:

Device Settings → Composite FM

The Composite FM menu is split across two screens. The first screen contains the level adjustments — **Pilot Level**, **Output1 Level**, and **Output2 Level** — and the second contains the routing and processing options — **Out1 Type**, **Out2 Type**, **Pre-Emphasis**, and **RDS**. Items are selected and modified using the navigation pad: ▲ / ▼ to move between menu entries, ◀ / ▶ to scroll between screens or change parameter values, and the centre button to enter and confirm.

Pilot Level

The 19 kHz stereo pilot tone level is adjustable as a percentage of total composite modulation, between **6% and 12%**, with a typical broadcast value of 8% – 10%. The Pilot Level setting is located on the first screen of the Composite FM menu and is adjusted in 1% increments.

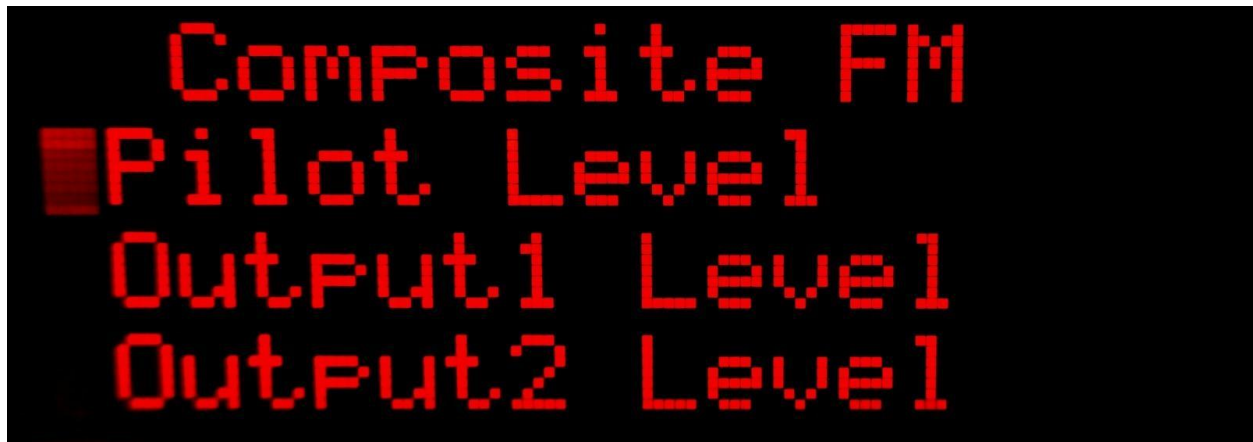


Figure 3.2.21: Pilot Level Control Screen

The value applies to whichever output(s) carry the pilot tone — the encoded MPX/COMP signal and, where configured, the standalone PILOT output on OUT2.

Output Configuration

- **OUT1:** Coaxial composite baseband output, software selectable as **COMP**, **MPX**, or **RDS**.
- **OUT2:** Coaxial composite baseband output, software selectable as **COMP**, **MPX**, or **PILOT** tone at 19 kHz.

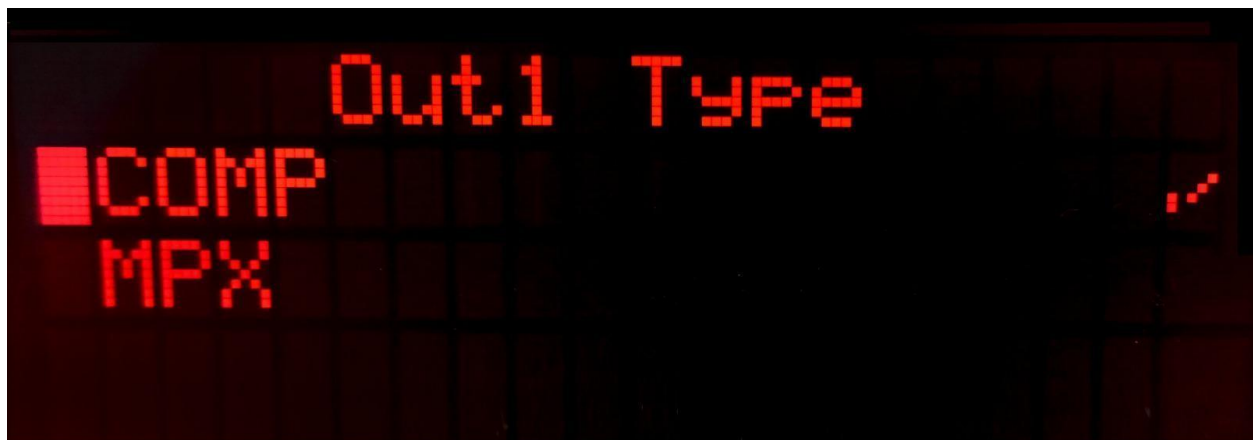


Figure 3.2.22: Out1 Type Screen

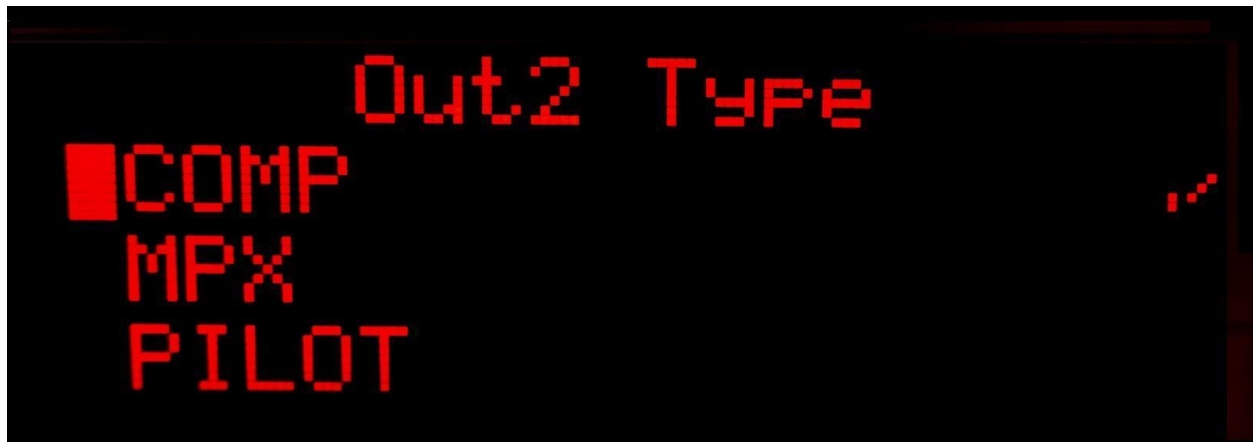


Figure 3.2.23: Out2 Type Screen

The output type for each connector is set independently from the second screen of the Composite FM menu using the **Out1 Type** and **Out2 Type** items. This allows the two physical outputs to be assigned to different signal types simultaneously — for example, a fully encoded composite (MPX) signal on OUT1 driving the FM exciter while OUT2 carries a 19 kHz pilot reference for synchronisation of an external RDS encoder or auxiliary equipment.

Output Level Adjustment

- **OUT1 Level:** Adjustable from **-12 dBu to +12 dBu**, configured from the first screen of the Composite FM menu.
- **OUT2 Level:** Adjustable from **-12 dBu to +12 dBu**, configured from the first screen of the Composite FM menu.



Figure 3.2.24: Output Level1 Control Screen



Figure 3.2.25: Output Level2 Control Screen

Each output level is set independently, allowing the operator to match the input sensitivity of the connected exciter, modulation monitor, or downstream device without altering the internal processing chain. After selecting the level item with the centre button, the value is adjusted in 0.1 dBu steps using ▲ / ▼ and confirmed with the centre button.

Pre-Emphasis

The **Pre-Emphasis** setting selects the pre-emphasis curve applied to the composite path:

- 50 μ s
- 75 μ s



Figure 3.2.26: Pre-Emphasis Selection Screen

If pre-emphasis has already been applied earlier in the main signal path (for example, on the HD or analog output stage), this option is disabled and the same pre-emphasis value is automatically inherited by the composite path to prevent double emphasis. If pre-emphasis has not been applied on the main path, the operator may manually configure the pre-emphasis type for the composite path independently from this menu.

RDS

The **RDS** item enables or disables the Radio Data System subcarrier on the composite output and selects its source:

- **OFF:** RDS subcarrier disabled.
- **INT:** RDS subcarrier sourced from the internal RDS encoder of the Q24-1111HD. Station identification, programme service name, RadioText, and other RDS fields are configured from the dedicated RDS configuration menu of the device.
- **EXT:** RDS subcarrier sourced from an external RDS encoder connected to the rear-panel **RDS/RBDS IN** connector. In this mode, the internal encoder is bypassed and the externally supplied RDS signal is summed into the composite baseband.

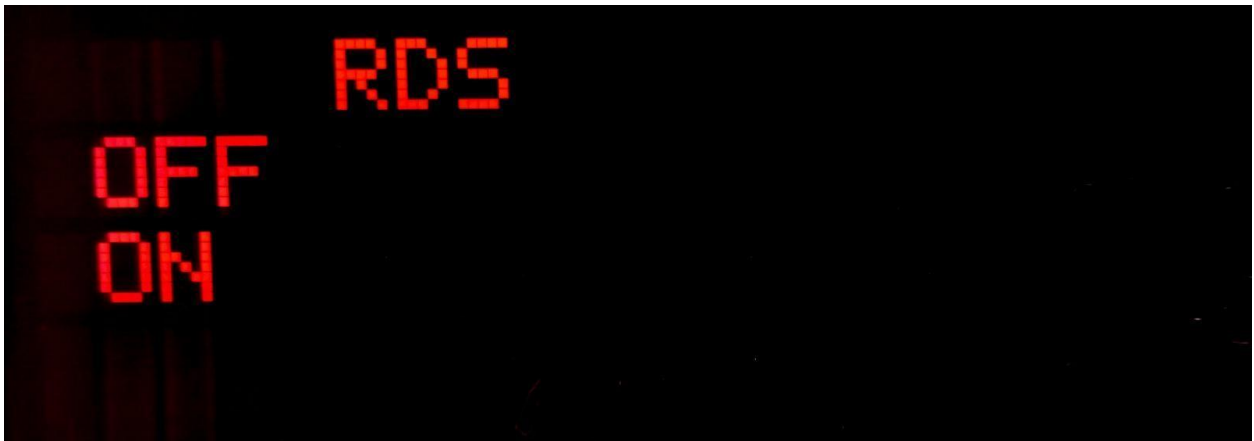


Figure 3.2.27: RDS Screen

When RDS is set to **EXT**, ensure that the external encoder is correctly synchronised to the 19 kHz pilot tone provided on OUT2 (configured as PILOT) to maintain the correct phase relationship of the 57 kHz subcarrier.

Input Configuration

The Q24-1111 provides two auxiliary baseband inputs that can be summed into the composite output:

- **IN1:** Selectable as **OFF**, **SCA**, or **PILOT**. When set to SCA, the input accepts an external SCA subcarrier (typically 67 kHz or 92 kHz) for injection into the composite baseband. When set to PILOT, the input accepts an external 19 kHz pilot reference, allowing the Q24-1111 to lock its internal stereo encoder to an external reference.
- **IN2:** Dedicated to RDS. The RDS source is selected via the **RDS** menu item described above — either the internal RDS encoder or the external signal applied to this input.

3.3 Profiles

Press the **Profile** key to display the complete list of available factory presets and custom profiles.

1. Factory Presets

The device includes a range of factory presets optimized for specific audio genres. Each preset name reflects its intended listening environment and processing configuration, indicating the number of active processing bands. Available presets include:

- Preset-24 Classical Opera
- Preset-24 Jazz Club
- Preset-24 Jazz Blue
- Preset-24 Soft Rock
- Preset-24 Hip Hop
- Preset-24 Hits-Pop



Fig 3.3.1: List of Profile

The preset name serves as a guide to its optimal use case, helping the user select the most suitable audio profile for their content.

2. Custom User Profiles

Users may create personalized audio profiles by adjusting the Equalizer settings through the Manager interface. Once saved, custom profiles are assigned a prefix in the following format:

CUSTOM-24-[Profile Name]

Custom profiles appear alongside factory presets in the selection list and can be applied in the same manner.

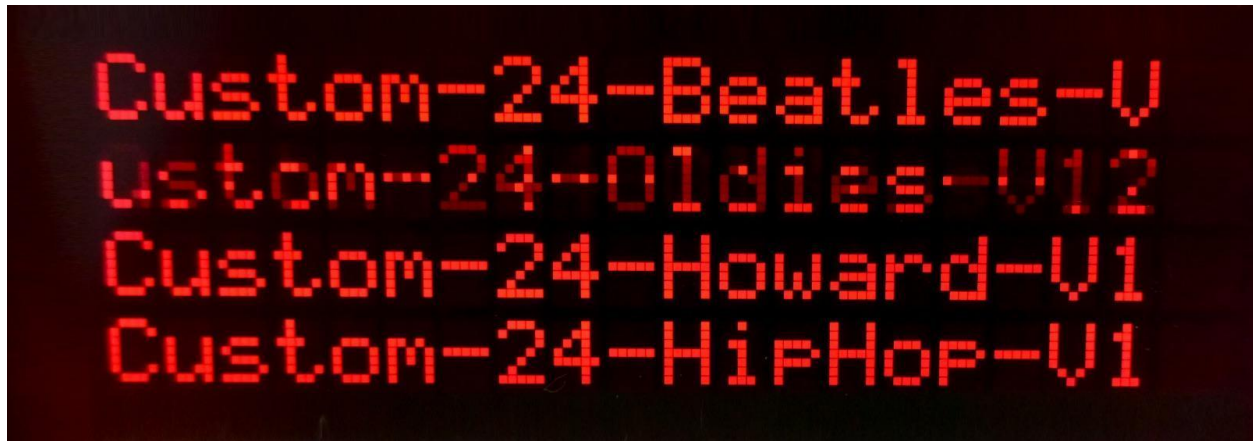


Fig 3.3.2: List of Profile

3. Selecting & Applying a Profile

- From the **Home Screen**, press the **Profile** key to open the preset and custom profile list.
- Use the **Up (▲) / Down (▼)** navigation keys to scroll through the available profiles.
- While traversing the list, the audio output updates in real time, allowing the user to audition each profile before confirming selection.
- Once the desired profile is identified, press the **circular soft key** to apply it.
- Upon applying a new preset or custom profile, the status LEDs will change color to orange, confirming the profile change (see figure).

Note: Press the **Back/Home** key to return to the Home Screen.

Section 4 SoundMaxManager™ Controls

SoundMaxManager™ controls can be adjusted from the detailed view in Figure 4.1 below.

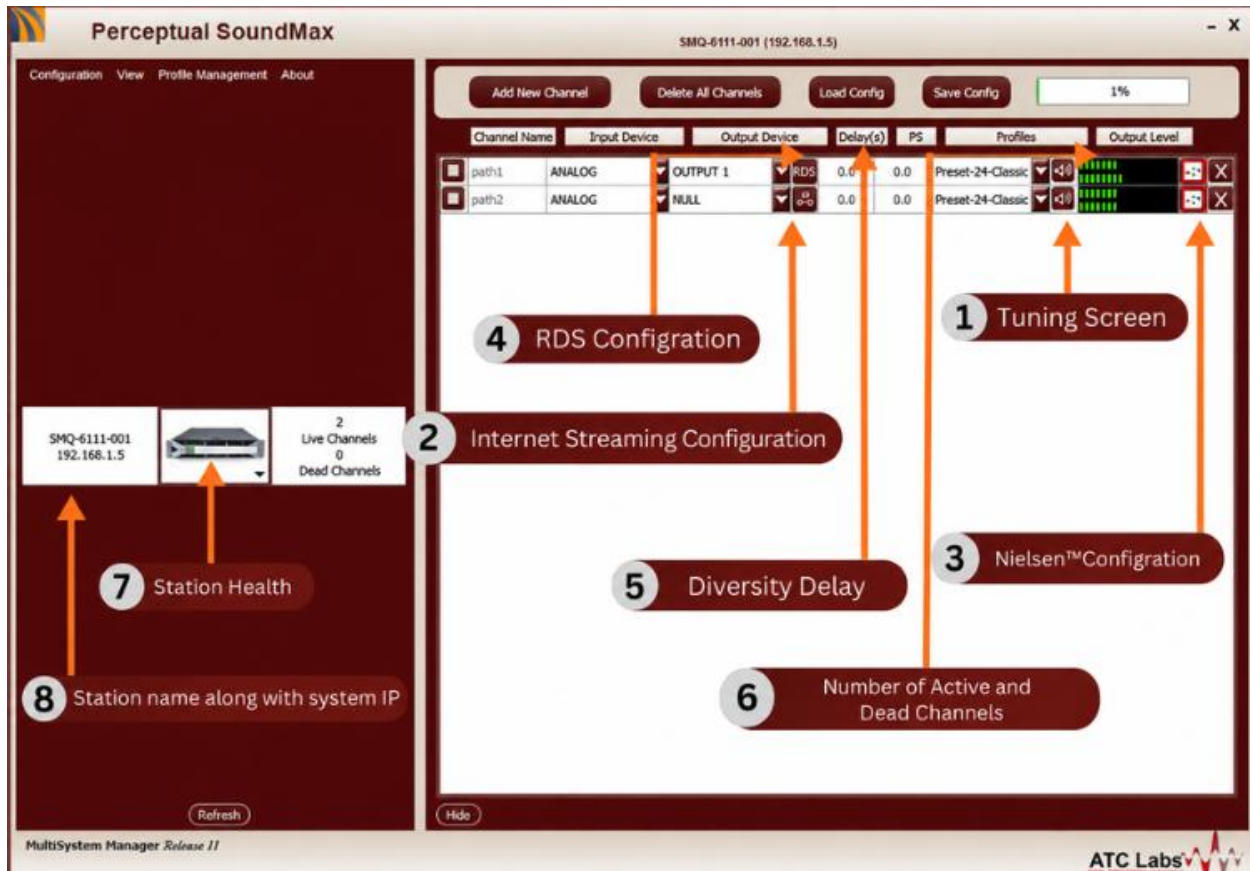


Figure 4.1: SoundMaxManager™ Detailed View (Q24-1111HD) - adjustments & controls

4.1 Tuning Screen for Creating & Fine-Tuning Processing Profiles

Detailed description of the effect of each of these modules is in Section 5.2 You may start from one of the available preset profiles and then create a custom signature sound for your station by adjusting these parameters. After tune ups, profiles are saved as a Custom profile.

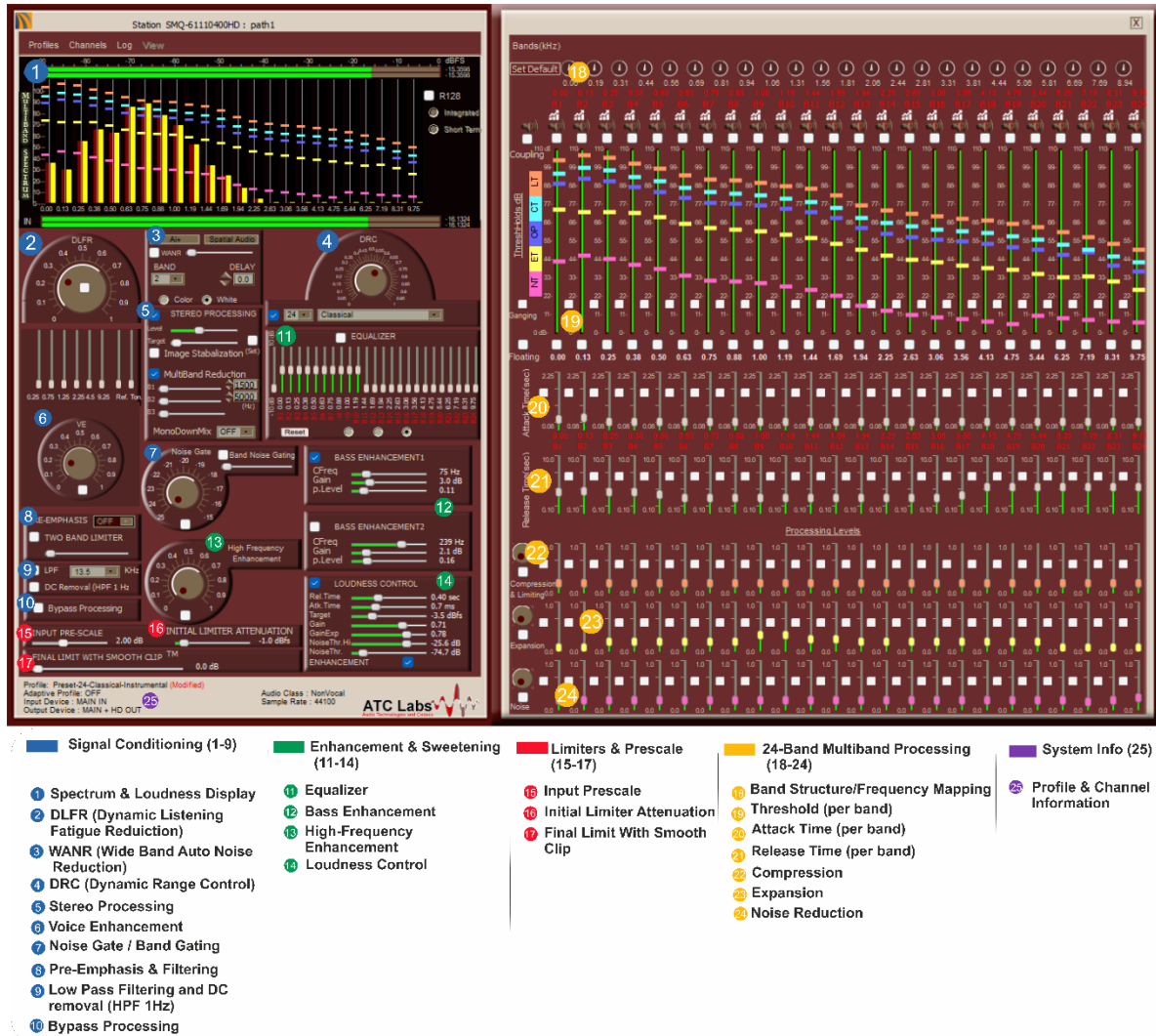


Figure 4.1.1: Tuning screen to control/set various parameters

4.1.1 Factory Preset Profiles

Available Factory Programming Presets are our “factory recommended settings” for various audio formats or classes. These are designed as starting points to help get on air quickly without having much

to understand and adjust the various controls. Each of these can be further tuned based on your broadcast's preferences and saved as a custom profile or user defined profile, so that next time the user can directly access and use one of these user-defined Custom Profiles along with the Preset Profiles. The available Factory Preset Profiles broadly fall in these categories:

- **CLASSICAL:** Softer compression to ensure that frequency details do not become dense, and sound open and softer on the ears.
- **CLASSICAL-INSTRUMENTAL:** Optimized for purely instrumental classical performances (orchestral, chamber, solo recitals). Uses extremely gentle multiband compression to preserve the natural dynamic range and tonal balance of acoustic instruments, with minimal high-frequency processing to avoid coloring strings, woodwinds, and brass. Ideal for stations broadcasting symphonies, concertos, and chamber music.
- **CLASSICAL-OPERA:** Tailored for operatic and vocal classical content. Provides slightly more midrange presence than the standard Classical preset to keep vocal lines forward and intelligible against the orchestra, while preserving the wide dynamic range expected in opera broadcasts. High-frequency processing remains gentle to avoid sibilance on dramatic soprano passages.
- **JAZZ-CLUB:** Tuned for live, small-ensemble jazz recordings — combos, trios, and quartets typical of club performances. Provides intimate presence with light re-equalization that enhances the warmth of upright bass and brushed drums while keeping cymbals and horns natural and unfatiguing. Low-noise floor management preserves the ambience of live recordings.
- **JAZZ-BLUES:** Designed for jazz-blues and blues-influenced jazz programming. Adds slightly more low-mid warmth and a touch of midrange grit for vocal presence, while maintaining the clean, mellow high-end characteristic of jazz processing. Suitable for stations playing electric blues, blues-rock fusion, and vocal jazz with blues roots.
- **SOFT ROCK:** A balanced preset for adult contemporary, soft rock, and lite-rock formats. Provides moderate compression with smooth high-frequency presence and gentle bass enhancement. Vocals are kept forward and warm without harshness, making it ideal for relaxed daytime and evening programming.
- **SOFT ROCK-OLDIES:** Optimized for classic soft rock, oldies, and gold-format stations. Compensates for the varying production quality of older recordings by providing consistent loudness and warmth, with mild high-frequency restoration to bring out detail in vintage masters without exposing tape hiss or analog artifacts.
- **ROCK-HARD ROCK:** A more aggressive preset designed for rock and hard rock formats. Delivers punchy low end, tight midrange, and bright, articulate highs to cut through in mobile and noisy listening environments. Uses faster multiband release times to maintain density and energy on heavily produced rock recordings.
- **HITS-POP:** Designed for mainstream contemporary hit radio (CHR) and pop formats. Provides bright highs, punchy lows, and a forward midrange that keeps vocals prominent. Loudness and density are optimized for competitive on-air presence in crowded urban dial positions.
- **HITS-POP HEAVY:** A more aggressive variant of HITS-POP with denser multiband compression and stronger limiting. Delivers maximum perceived loudness and consistency for stations

targeting younger demographics and competitive market positioning. Best suited for high-energy dayparts and pop/dance crossover programming.

- **HITS-POP HEAVY-VOCALPLUS-LOWERBASS:** A specialized variant of HITS-POP HEAVY with enhanced vocal presence and reduced low-bass energy. Designed for broadcasters whose target listeners primarily use small speakers, smartphones, or earbuds where excessive sub-bass is wasted or causes distortion. Vocals remain crystal clear while overall loudness stays competitive.
- **HITS-POP-ENHANCED-VOCALS:** A pop preset with increased vocal clarity and presence processing. Uses targeted midrange enhancement and dedicated vocal-band tuning to push lead vocals forward in the mix without sacrificing instrumental balance. Ideal for vocal-driven pop, R&B, and adult contemporary formats.
- **HITS-POP-ENHANCED-VOCALS-BASS:** Combines the enhanced vocal processing of HITS-POP-ENHANCED-VOCALS with stronger bass enhancement for full-range listening environments. Designed for stations whose audience primarily listens on car audio systems, home stereos, or full-range speakers where both vocal clarity and powerful low end are desired.
- **HIP HOP:** A specialized preset tailored for hip hop, rap, trap, and urban music programming. Delivers powerful sub-bass and bass response with controlled, punchy kick drums and 808s. Midrange is tuned to keep rap vocals articulate and intelligible against dense low-end production. Stronger limiting maintains competitive loudness without losing low-frequency impact.
- **HITS-HEAVY/LIGHT:** The HITS-HEAVY/LIGHT presets are designed for bright high and punchy low end. A midrange boost provides enough presence energy to ensure that vocals stand out.
- **JAZZ:** Specifically tailored toward broadcasters that play mostly instrumental music, particularly classic jazz (Coltrane, Mingus, Monk, etc.). It is a quiet preset with a very clean, mellow high end to prevent stridency on saxes and other horns. It preserves much of the qualities of the original recordings, doing light re-equalization. The preset produces very low listening fatigue, so it is a good choice for broadcasters that want listeners to stay all day.
- **NEWS SPORTS BROADCAST:** Optimized for news, talk, and live sports broadcasting. Uses fast multiband release times and aggressive automatic equalization to handle widely varying source material — studio microphones, telephone calls, field reports, stadium audio, and pre-recorded segments. Speech intelligibility is prioritized over musical fidelity, ensuring announcers and commentators remain clear and consistent across changing conditions. Particularly effective for live sports where crowd noise, on-field audio, and announcer voices must coexist clearly.
- **NEWS-TALK:** This preset is quite different from the others. It is based on the fast multiband 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 uncontrolled quality.

4.1.2 Custom Profile Creation

As mentioned above a user may start from a factory preset and then tune parameters associated with one or more processing blocks to create a signature on-air sound. Alternatively, user may want to invoke a processing block which was previously unused. These tasks can be accomplished using either the LCD screen interface in Section 3 or the SoundMaxManager™ Software as described in Section 5; with the SoundMaxManager™ providing a much higher level of control in terms of tuning the processing parameters as described above. Once a user is satisfied with the sound the resulting profile may then be saved as a custom profile either from the LCD or Controller and available as an option in future.

4.2 Internet Streaming Configuration

Customized internet radio from Perceptual SoundMax™ gives you a totally new experience in terms of simplicity and quality. Internet streamlining can be controlled from both SoundMaxManager™ and the Q24-1111. Following are the steps of configuration.

4.2.1 Stream's Common Settings Panel

The Common Settings Panel (Figure 4.2.1) is used to define the identity of your stream and view network connection details.

- **Stream Name:** Enter a unique stream name for the selected channel. Each channel must have a distinct stream name to avoid conflicts.
- **Streaming Interface:** (Read-only) Displays the public IP address associated with the streaming interface.
- **HTTP Pull Port (AAC):** (Read-only) Displays the port used for standard AAC HTTP Pull requests if native AAC HTTP server is enabled as described in Section 4.2.2.
- **HTTP Pull Port (m3u8):** (Read-only) Displays the port dedicated to HLS HTTP Pull requests if native MP3 HTTP server is enabled as described in Section 4.2.2.

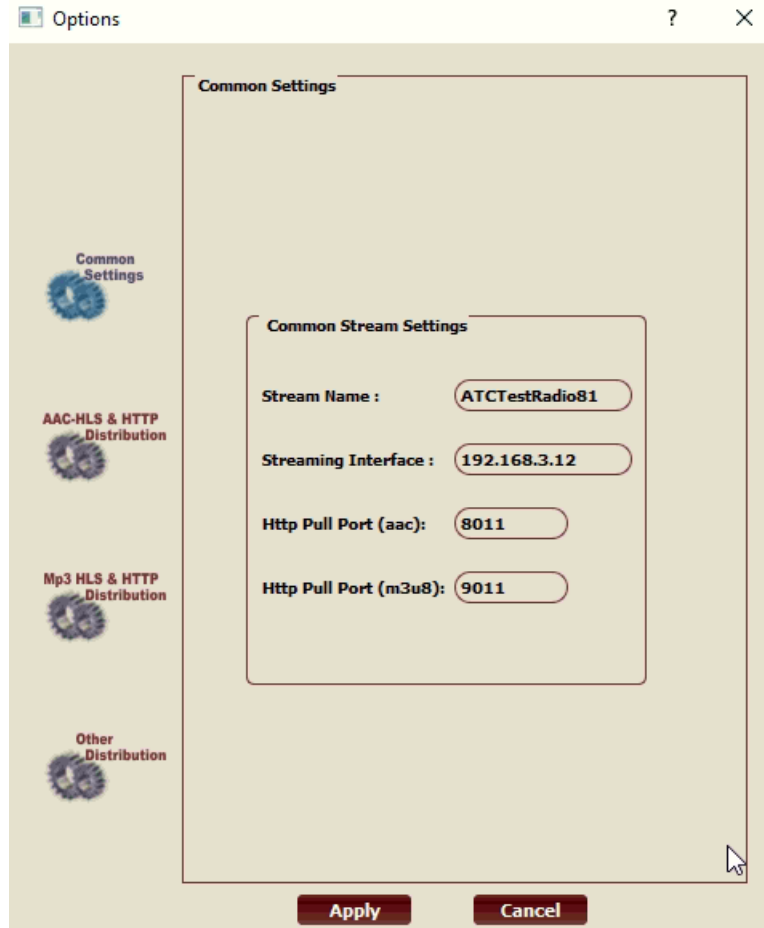


Figure 4.2.1: Common Settings Panel

4.2.2 AAC HLS & HTTP Distribution Settings

The **AAC HLS & HTTP Distribution** panel (Figure 4.2.2) provides comprehensive control over audio encoding, segmenting, and delivery methods.

AAC Codec Profiles & Bitrates

SoundMax™ Serverware offers industry-standard AAC encoding with flexible profile support.

- **HLS Configuration**

Enable the HLS Configuration checkbox to prepare your stream for adaptive bitrate playback

- **Chunk Duration (in secs):** Sets the length of individual audio segments (Default is 10s; adjustable from 3s to 10s).
- **Codec Profile:** Select the desired AAC profile for the adaptive stream.
- **LC (Low Complexity):** Ideal for high-fidelity streaming at higher bitrates.
- **HE (High Efficiency):** Optimized for superior audio quality at lower bitrates (includes HEv2 automatically for 32 kbps selections).

- **Note:** Only one profile type (either LC or HE) can be active for the adaptive stream at a time.
- **Encoding Configuration:** Select any two bitrates to be encoded simultaneously. This allows the player to switch between qualities based on the listener's network speed.
- **TS (Encapsulation):** Enable this toggle to encapsulate audio segments in the MPEG-TS format.
- **HTTP Pull:** Enables the local server to host the stream for direct client requests.
- **HTTP Primary/Backup Push:** Enables the application to send the stream to remote destinations. Click HTTP Push Settings (Figure 4.2.3) to configure URLs and credentials.

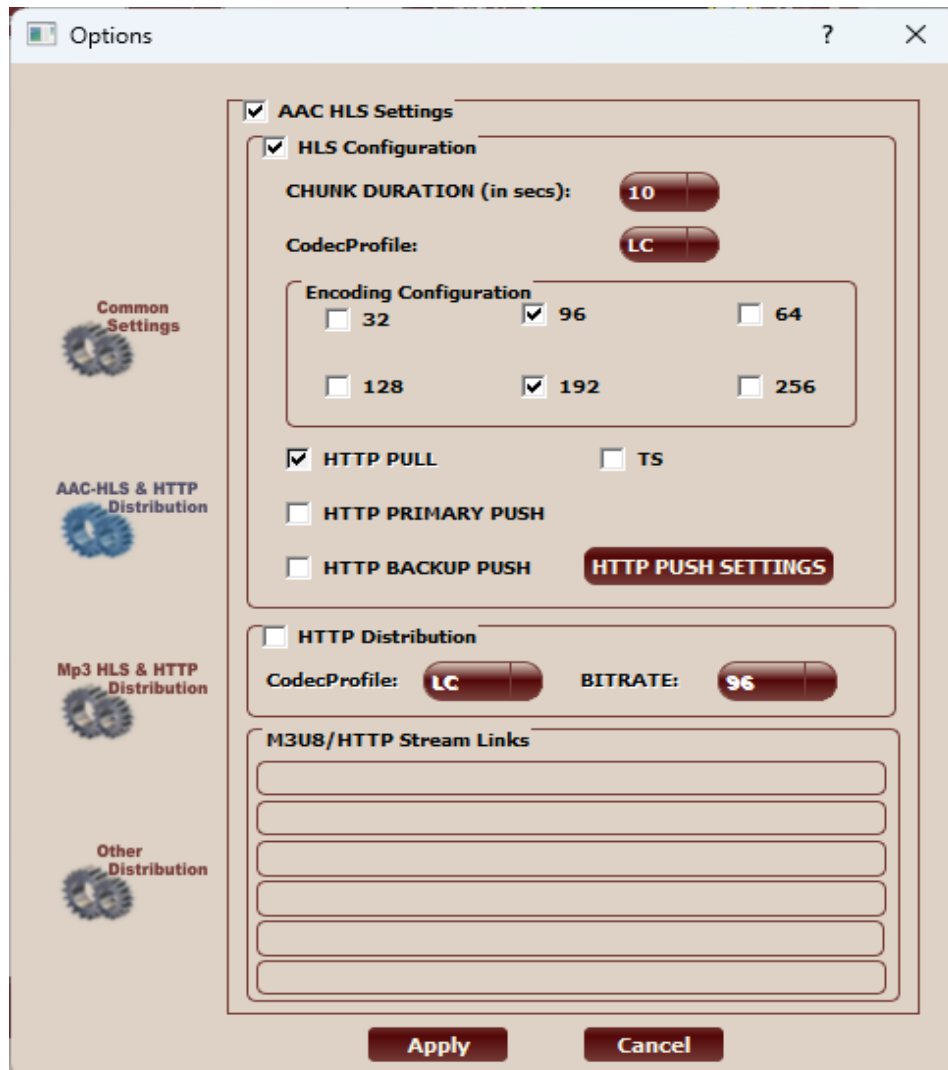


Figure 4.2.2: AAC HLS & HTTP Distribution Settings Panel

- **HTTP Distribution (Single Bitrate)**

This section is designed specifically for local distribution. When enabled, the Q24-1111 hardware functions as an internal IceCast-compatible server, hosting a standalone, non-adaptive HTTP stream for your local network.

- **Local Hosting:** In this mode, the hardware acts as the primary media origin. It accepts incoming **HTTP Pull** requests directly from local listeners or players without requiring an external CDN.
- **Configuration:**
 - ❖ **Codec Profile:** Select the desired profile (LC, HE, or HE-V2).
 - ❖ **Bitrate:** Choose a single, fixed bitrate (e.g., 64 kbps or 128 kbps) for the local broadcast.
 - ❖ **Use Case:** This is ideal for in-house monitoring, office-wide broadcasts, or any scenario where low-latency distribution within a private network is required.

Technical Note:

Accessing the Local Stream: Once configured and enabled, the stream is accessible via the local IP address of the Q24-1111 unit. The specific URL and port information will be displayed in the Common Settings Panel or the Stream Links area.

M3U8/HTTP Stream Links

Once settings are applied, this area automatically populates with the generated streaming URLs. These links can be copied and distributed to listeners or embedded into web players.

Adaptive Strategy: For the best user experience, it is recommended to select one high bitrate (e.g., 128 or 192) and one lower bitrate (e.g., 32 or 64). This ensures that listeners on mobile data stay connected even in poor signal areas.

HTTP Info Settings Panel

The Set HTTP Info panel allows users to configure the technical parameters required for pushing HLS data to remote servers or CDNs (such as Akamai, BitGravity, or AWS).

Figure 4.2.3: AAC HTTP Info Settings Panel

Primary & Backup Publishing Info

To ensure stream redundancy, the system supports two independent publishing destinations:

- **Host IP / URL:** The destination server address (e.g., ep2036929.i.akamaientrypoint.net).
- **Remote Path:** The specific directory path on the server where the .m3u8 and audio segments will be stored.

Common Info (Authentication & Protocol)

These settings apply to both the Primary and Backup connections:

- **Username / Password:** Credentials required by the remote server for secure access.
- **Port:** The network port used for the upload (typically 80 for HTTP or 443 for HTTPS).
- **HTTP Push Type:** Defines the method used to send data.
 - **POST:** Standard for pushing data segments.
 - **PUT:** Typically used for updating the manifest file.
- **HTTP Auth:** The authentication method used by the server (e.g., BASIC or DIGEST).

Note: If your destination does not require authentication, the Username field may be set to NULL.

- **Redundancy:** Remind users that the Remote Path for the Backup Info should usually be different from the Primary path (e.g., adding a -b suffix) to keep the stream sources distinct.
- **Verification:** Ensure the Port matches the security settings of the destination server (Port 80 is often blocked by strict firewalls in favour of Port 443).

4.2.3 MP3 HLS & HTTP Distribution Settings

For applications requiring legacy compatibility or specific bandwidth requirements, the **MP3 Settings panel** allows for high-quality MP3 encoding and distribution.

HLS Configuration (MP3)

Like the AAC configuration, this section enables adaptive bitrate streaming using the MP3 codec.

- **Encoding Configuration:** Select any two bitrates from the available options: **32, 64, 96, 128, 192, 256, or 320 kbps**. The system will encode these simultaneously for seamless adaptive switching.
- **Distribution Options:** Use the checkboxes to enable **HTTP Pull** (local hosting), **TS Encapsulation**, or **HTTP Primary/Backup Push** to remote servers.

MP3 HTTP Distribution

This section handles single-bitrate streaming.

- **Bitrate:** Choose a fixed bitrate for standard HTTP distribution.

M3U8/HTTP Stream Links

Once settings are applied, the generated URLs for the MP3-based HLS playlist or the raw HTTP stream will appear in this list for easy copying.

- **Bitrate Ceiling:** MP3 supports up to **320 kbps**, providing a higher fidelity option compared to the standard AAC profile limits.
- **Compatibility:** Use MP3 HLS for maximum compatibility with older web players and legacy mobile devices that may struggle with AAC/HE-AAC profiles.

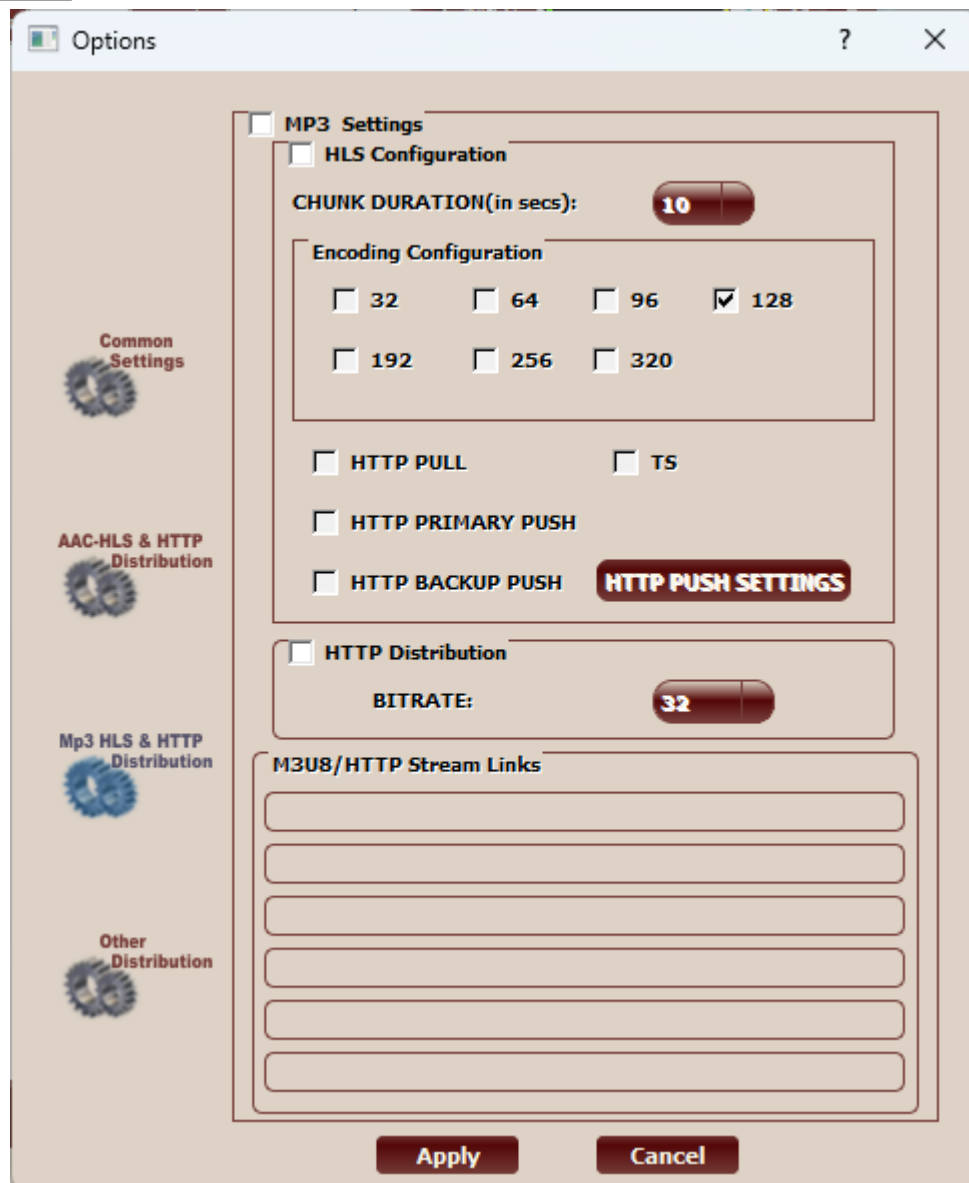


Figure 4.2.4: MP3 HLS & HTTP Distribution Settings Panel

4.2.4 Other Distribution Settings

The Other Distribution Settings panel handles legacy web broadcasting and local network streaming. This interface supports two specific transmission modes: **IceCast** and **Udp/Multicast** (Multicast/Unicast).

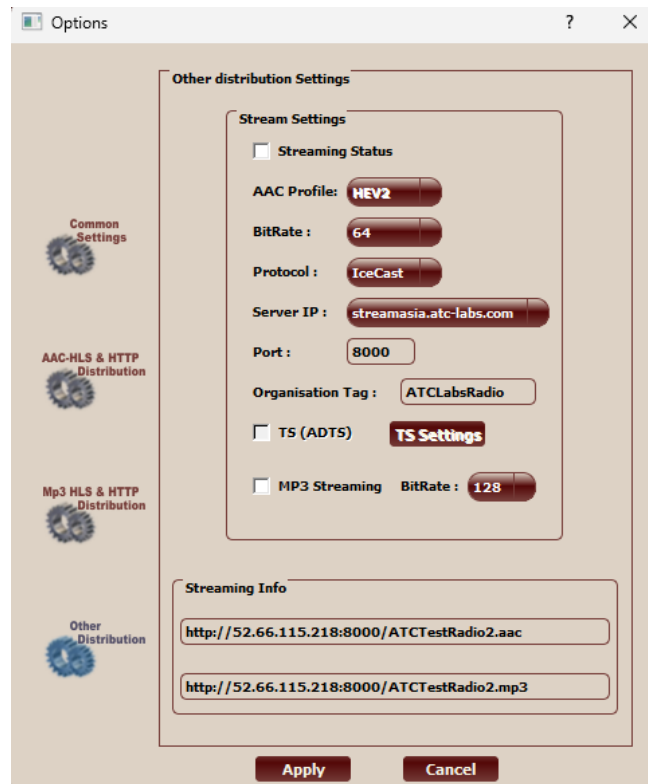


Figure 4.2.5: Streaming Panel

Server/Protocol Settings

- **Protocol:** Select your delivery method:
 - **IceCast:** Delivers the stream to one of the pre-authorized ATC Labs IceCast servers.
 - **Udp/Multicast:** Enables Multicast or Unicast transmission for distribution over a private Local Area Network (LAN).
- **Port:** The destination network port (Default: 8000).
- **Server:** Use the dropdown menu to select the authorized IceCast server (e.g., streamasia.atc-labs.com). If using Udp/Multicast, enter the target IP address.
- **Organisation Tag:** Enter the unique mount point name for your station.
- **TS (ADTS):** Note: This option is exclusively available for Udp/Multicast (Multicast/Unicast) modes. It enables Audio Data Transport Stream headers for improved synchronization across local network receivers.

Primary (AAC) Stream Settings

Streaming (Checkbox): Enables the primary high-efficiency AAC stream.

- **AAC Profile:** Select from LC, HE, or HE-V2 profiles.
- **Codec Mode:** Choose between STEREO or MONO output.
- **BitRate:** Set the target bitrate for the primary stream.

Backup (MP3) Stream Settings

- **Streaming (Checkbox):** Enables a secondary MP3 stream.
- **BitRate:** Set the bitrate for the backup MP3 stream (e.g., 128 kbps). This ensures compatibility for devices or legacy software that do not support AAC.

Streaming Info

This section displays the finalized URLs for your streams. These links are automatically updated based on the Server and Organisation Tag entered above.

- **Protocol Constraint:** The TS (ADTS) toggle and TS Settings are strictly for use with the **Udp/Multicast** protocol. When **IceCast** is selected, these settings are disabled or should be left unchecked as the IceCast protocol handles stream encapsulation natively.

4.3 Nielsen™ Configuration

The Nielsen Configuration Panel is used to manage **CBET (Critical Band Encoding Technology)** watermarking. This panel enables the insertion of industry-standard watermarks into the audio stream. These watermarks are inaudible to listeners but are detectable by Nielsen metering devices.

The screenshot shows the 'NIELSEN CONFIGURATION PANEL' with a close button (X) in the top right corner. The panel contains the following settings:

- CBET Enable:** A checkbox that is currently unchecked, followed by a text field containing 'UNCONFIGURED'.
- Process Type:** A dropdown menu currently set to 'DISABLE_WATERMARKING'.
- CBETSid(hexadecimal):** A text field containing '0x'.
- CBETCheckDigits:** A text field with a double-headed arrow icon on the right.
- TimeCode:** A dropdown menu currently set to 'INSERTER_CLOCK_ALWAYS'.
- CBETRadioMode:** A dropdown menu currently set to 'DOMESTIC (USA)'.
- PriorDetMode:** A dropdown menu currently set to 'OVERWRITE'.
- EAS Disable:** A dropdown menu currently set to 'ON'.

An 'OK' button is located at the bottom center of the panel.

Figure 4.3.1: Nielsen™ configuration Panel

Configuration Settings

- **CBET Enable:** Toggle this checkbox to activate or deactivate the Nielsen watermarking engine.
- **Process Type:** Select the watermarking operational mode (e.g., DISABLE_WATERMARKING, ENABLE_WATERMARKING).
- **CBETSid (Hexadecimal):** Enter the unique Source Identifier (SID) provided by Nielsen for your station. This must be entered in hexadecimal format (starting with 0x).

- **CBETCheckDigits:** Used to input the specific check digits assigned to your SID to ensure data integrity.
- **TimeCode:** Configures how the watermark is timestamped. Options like INSERTER_CLOCK_ALWAYS ensure the watermark stays synchronized with the real-time broadcast.
- **CBETRadioMode:** Defines the geographic/regulatory region for the watermark (e.g., DOMESTIC (USA)).
- **PriorDetMode:** Controls how the system handles pre-existing watermarks in the source audio. Selecting OVERWRITE ensures the current station's watermark takes precedence.
- **EAS Disable:** Allows you to toggle the Emergency Alert System (EAS) detection. When set to ON, it prevents watermarking from interfering with or being applied during emergency broadcast signals.
- **Important:** Improper configuration of the CBETSid or CheckDigits will result in inaccurate audience measurement. Ensure you use the official credentials provided by your Nielsen representative. If you are not participating in Nielsen ratings, keep Process Type set to DISABLE_WATERMARKING.

4.4 RDS Configuration

The Internal RDS Encoder Panel (Figure 4.4.1) is used to configure the Radio Data System (RDS) information. This allows you to transmit digital data such as station names and song titles alongside your audio, which is displayed on compatible FM receivers and digital players.

The screenshot shows a window titled "Internal RDS Encoder Panel" with a close button (X) in the top right corner. The window is divided into three main sections:

- Programme Service Configuration:**
 - PS Type:** A dropdown menu set to "STREET_NAME".
 - Program Service (PS):** A text input field containing "ATCRadio".
 - Send Rate (secs):** A dropdown menu set to ".5".
- Radio Text Configuration:**
 - Static Radio Text:** A text input field containing "TCRadio: FM Stereo and RDS encoder".
 - Dynamic Radio Text:** Three radio buttons: "Overwrite" (selected), "Append", and "Disabled".
- RDS Data Input Configuration:** (This section is currently unchecked)
 - IP Address:** A text input field containing "192.168.1.7".
 - Port:** A text input field containing "0".
 - Protocol:** A dropdown menu set to "TCP".
 - ConnectionType:** A dropdown menu set to "SERVER".

An "OK" button is located at the bottom center of the panel.

Figure 4.4.1: Internal RDS Encoder Panel

This panel manages the metadata that identifies your station and provides real-time information to your listeners.

Programme Service (PS) Configuration

The PS is the static or dynamic 8-character name that appears on the listener's radio display.

- **PS Type:** Select the display behavior (e.g., STREET_NAME or STATION_ID).
- **Program Service (PS):** Enter the station name to be displayed (e.g., ATCRadio).
- **Send Rate (secs):** Defines how frequently the PS data is refreshed or cycled (e.g., 0.5 seconds).

Radio Text (RT) Configuration

Radio Text provides a longer string of information (up to 64 characters), typically used for artist names, song titles, or station slogans.

- **Static Radio Text:** Enter a permanent message (e.g., ATCRadio: FM Stereo and RDS encoder).
- **Dynamic Radio Text:** Configures how external metadata (like "Now Playing" info) interacts with your static text:
 - **Overwrite:** New metadata replaces the current text.
 - **Append:** New metadata is added to the end of the current text.

- **Disabled:** Disables real-time metadata updates.

RDS Data Input Configuration

This section is used to ingest RDS metadata from an external source (such as an automation system) over the network.

- **IP Address & Port:** Enter the network details of the source sending the metadata.
- **Protocol:** Select between TCP or UDP for data transmission.
- **Connection Type:** Defines if the encoder acts as a SERVER (waiting for data) or a CLIENT (reaching out to a data source).

Configuration Tip: For the best listener experience, ensure your Program Service (PS) name is clear and recognizable, as this is the primary identifier users see on their device displays. If you are using external automation software to send song titles, ensure the RDS Data Input Configuration matches your software's output settings.

4.5 AES67 IPAUDIO Configuration

The AES67 Configuration Panel (Figure 4.5.1) allows the system to interface with high-performance Audio-over-IP (AoIP) networks. This standard ensures interoperability between different hardware vendors for low-latency, uncompressed audio transmission over a local network.

AES67 Configuration Panel

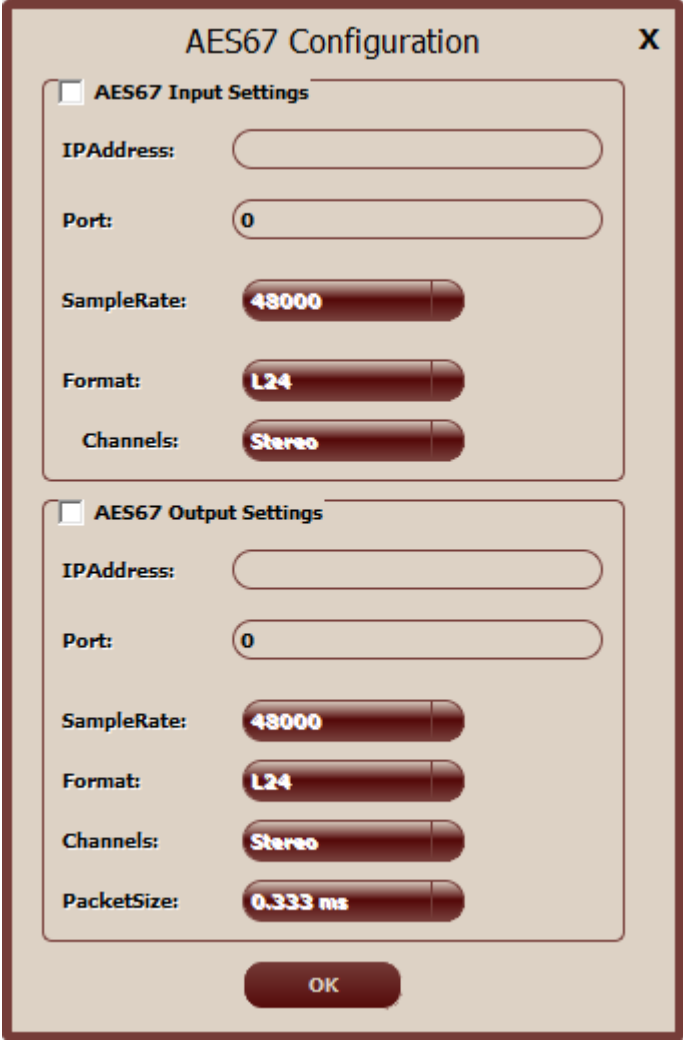
The **AES67 Configuration Panel** (Figure 4.5.1) enables high-performance Audio-over-IP (AoIP) interoperability. This allows the system to send and receive uncompressed, low-latency audio across a local network.

Important Note: The Q24-1111 / 1111HD series does not support auto-discovery. Manual configuration is required for both Input and Output streams. You must manually enter the exact IP addresses and ports assigned to your AOIP network devices.

AES67 Input & Output Settings

Configure the following parameters to match your specific network environment:

- **IP Address & Port:** Manually enter the Unicast or Multicast IP address and the corresponding network port for the stream.
- **Sample Rate:** Supports high-fidelity audio at either 44.1 kHz (44100 Hz) or 48 kHz (48000 Hz).
- **Format (Bit Depth):** Select between L16 (16-bit) or L24 (24-bit) linear PCM audio.
- **Channels:** Configure the stream for Stereo or Mono operation.
- **Packet Size (Latency):** (Available for Output) Defines the transmission interval. Smaller values provide lower latency but require a more robust network.
 - 0.125 ms / 0.333 ms / 1 ms / 4 ms
- **Clock Synchronization:** AES67 requires precise timing. Ensure your network switch is configured for PTPv2 (IEEE 1588-2008) to prevent audio clock drift.



The image shows a software window titled "AES67 Configuration" with a close button (X) in the top right corner. The window is divided into two main sections: "AES67 Input Settings" and "AES67 Output Settings", each with a checkbox at the top left. The "Input Settings" section includes fields for "IPAddress:" (empty), "Port:" (0), "SampleRate:" (48000), "Format:" (L24), and "Channels:" (Stereo). The "Output Settings" section includes fields for "IPAddress:" (empty), "Port:" (0), "SampleRate:" (48000), "Format:" (L24), "Channels:" (Stereo), and "PacketSize:" (0.333 ms). An "OK" button is located at the bottom center of the window.

Figure 4.5.1: AES67 Configuration Panel

Important: Double-check your IP and Port entries; because there is no auto-discovery, the system will not "see" the stream unless these details are perfectly matched.

4.6 Diversity Delay (Model Q24-1111HD Only)

The Diversity Delay dialog (Figure 4.6.1) provides precise control over the audio delay applied to the output signal. This is essential for HD Radio (IBOC) installations where the analog FM signal must be time-aligned with the digital HD Radio signal to ensure seamless blending during receiver transitions between analog and digital modes.

The dialog is opened from the corresponding Diversity Delay control on the parent processing screen, and is presented as a modal window over the Manager interface. Click the **X** button in the top-right corner to close the dialog without applying changes.

Fig 4.6.1: Diversity Delay

Sample Rate

At the top of the dialog, the **Samplerate** field displays the current operating sample rate of the audio path (e.g. **44100** Hz). This value is informational and is used internally by the Manager to convert the configured delay between time-based units (seconds / milliseconds / microseconds) and the equivalent number of audio samples shown in the **Total Delay (samples)** field. The sample rate is determined by the system configuration and is not editable from this dialog.

Delay Configuration

The total delay is specified using three numeric fields, each with up/down spinner controls for precise adjustment. The configured delay is the sum of the three values:

- **Seconds (2.0 – 9.0)** — Sets the integer-seconds component of the delay. Adjustable from 2 to 9 seconds in 1-second steps using the up/down spinners. The value may also be typed directly into the field.
- **Milliseconds (0 – 99)** — Sets the millisecond component, adjustable from 0 to 99 ms.
- **Microseconds (0.0 – 999.0)** — Sets the microsecond component, adjustable from 0 to 999 μ s.

The total configured delay is therefore the sum: *Seconds + Milliseconds + Microseconds*. For example, a setting of Seconds = 8, Milliseconds = 4, Microseconds = 188 produces a total delay of 8.004188 seconds.

Display Fields (Read-only)

The two fields below the input controls are calculated automatically from the values entered above and cannot be edited directly:

- **Total Delay (secs)** — Displays the combined total delay expressed in seconds (e.g. 8.004188 seconds). This is a real-time readout that updates as the Seconds, Milliseconds, and Microseconds values are changed.
- **Total Delay (samples)** — Displays the equivalent delay expressed in audio samples at the current sample rate (e.g. 384,201 samples at 44,100 Hz). This value is provided for engineers who specify diversity delay in samples rather than time.

Samples Adjusted (Fine Calibration)

Below the read-only fields, the **Samples Adjusted (-10 to 10)** control allows fine-tuning of the delay by adjusting a small number of audio samples (range: -10 to +10) on top of the configured time-based delay. This is intended for minor calibration corrections to bring the analog and HD signals into precise sample-level alignment without modifying the primary Seconds / Milliseconds / Microseconds settings.

A positive value adds samples (lengthens the delay); a negative value removes samples (shortens the delay). Each sample corresponds to $1 / \text{sample-rate}$ seconds — at 44.1 kHz, one sample is approximately 22.7 μs .

The **RESET** button beside this field clears the Samples Adjusted value back to **0**, removing any fine-calibration offset without affecting the primary delay configuration.

Apply

The **APPLY** button at the bottom of the dialog commits the current delay configuration (Seconds + Milliseconds + Microseconds + Samples Adjusted) to the active processing path. Changes are not active until APPLY is pressed; closing the dialog with the **X** button discards any pending changes that have not been applied.

After clicking APPLY, the Total Delay readouts continue to reflect the active configuration, and the Manager confirms the change has taken effect.

Typical Workflow

A typical diversity-delay alignment proceeds as follows:

1. Confirm the **Samplerate** field shows the expected operating sample rate of your processing path.
2. Set **Seconds**, **Milliseconds**, and **Microseconds** to the coarse delay value supplied by your HD Radio import / exporter equipment (typically in the range of 8 – 8.5 seconds for IBOC installations).
3. Verify the **Total Delay (secs)** and **Total Delay (samples)** readouts match the expected target.
4. Click **APPLY** to commit the delay.
5. Listen to the on-air analog and HD signals through a representative receiver and confirm seamless blending. If a small audible discontinuity remains, use **Samples Adjusted** to nudge the alignment by a few samples, click APPLY again, and re-evaluate.
6. If you need to start over, click **RESET** to clear the Samples Adjusted offset, or re-enter the primary delay fields manually.

Section 5 Algorithmic Insights & Processing Optimizations

5.1 Audio Processing Architecture & Operation

This section discusses the audio processing architecture and operation.

5.1.1 Processing Architecture Highlights

Perceptual SoundMax™ offers several unique audio processing features.

These include:

High Resolution Audio Processing: Perceptual SoundMax™ introduces a novel *high-resolution* audio processing engine. It imparts processing and offers control over up to 24 psychoacoustically tailored bands. Complex signal envelope in each of these up to 24 bands is continuously estimated at a high time resolution of 0.5 msec and adaptively processed using sophisticated perceptually driven algorithm. User controllable parameters are available in all these 24 bands and can be independently tuned. Compared to conventional 5-6 band audio processing technologies the high-resolution processing allows for the processing to be accurately matched to application scenarios; e.g., to the genre of the audio input or in the case of digital broadcasting to the unique codec characteristics/impairments, in multiple frequency bands. Furthermore, in general, more bands allow for *lighter processing* to achieve a target impact level since each bands affects a smaller portion of audio, resulting in a more open and richer sound.

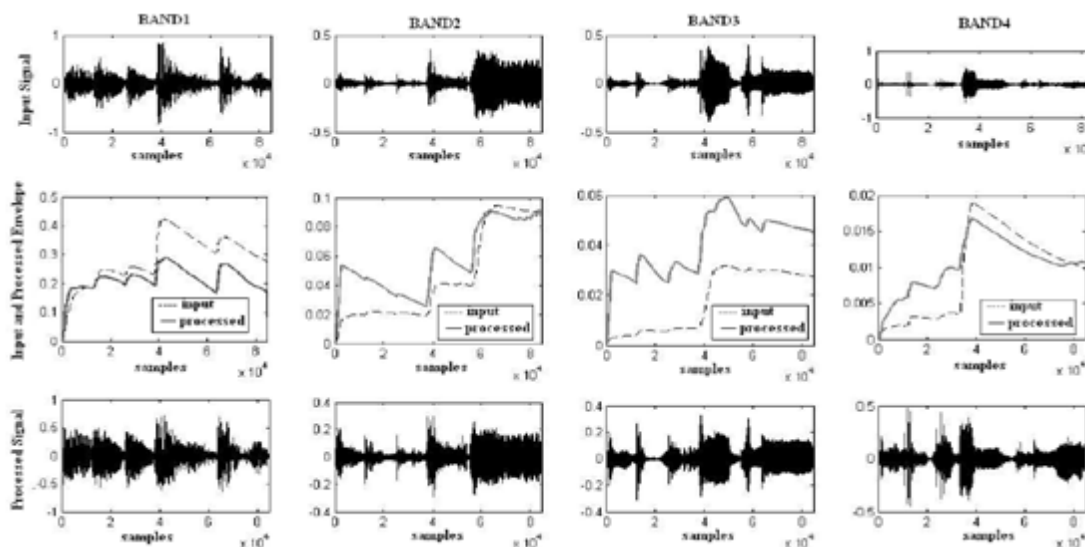


Figure 5.1.1: Perceptual SoundMax™ Envelope Processing in multiple *high resolution* frequency bands

Linear Phase Audio Processing: The multi-band processing in Perceptual SoundMax™ is totally Linear Phase. Traditionally, audio processors have preferred minimum phase processing (as opposed to linear phase) in the belief that minimum phase preserves attack sharpness. However, this results in phase distortion which needs to be corrected with a *phase compensator*. This is not an ideal situation, particularly for digital broadcasting as codecs have increased sensitivity to phase distortions. Perceptual SoundMax™ on the other hand uses a carefully optimized *linear phase* yet *low-delay* filterbank to achieve the right balance. The end effect is completely linear-phase processing with high temporal resolution that totally avoids phase distortion - common with many other audio processors - while preserving attack sharpness

Integrated Compression/Limiting/Expansion/Noise Gating: Perceptual SoundMax™ implements all envelope processing (i.e. expansion/AGC and compression/limiting) in each of the 24 bands in a single stage. This avoids a situation where different processing stages are fighting and/or negating each other's impact - a situation common with many other processing technologies, which typically implement multi-band compression and multi-band AGC as two separate modules. The *single-stage* envelope non-linearity for each band in Perceptual SoundMax™ is illustrated in Figure 5.1.2 below. In this it may also be noted that complete flexibility is offered in setting the boundaries of each region –i.e. limiting, compression, expansion, and noise thresholds in each bands are fully configurable. Moreover, the level of processing in each of the regions is also independently controllable.

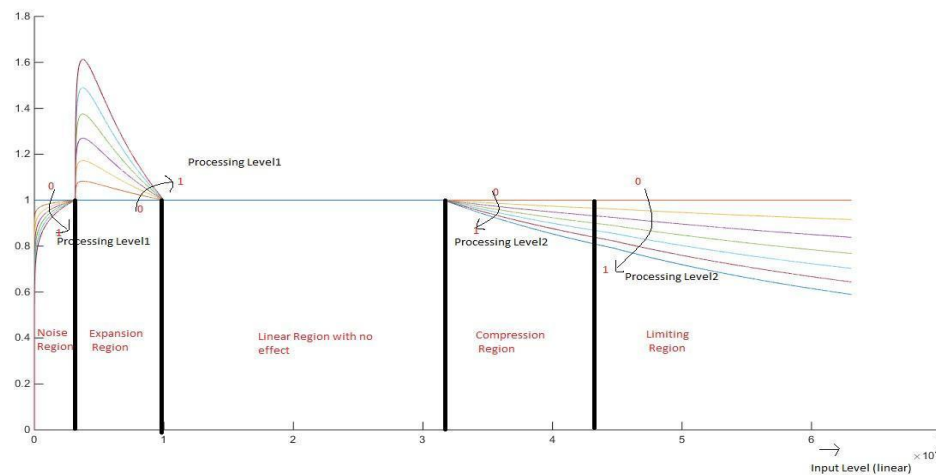


Figure 5.1.2: The configurable shape of Perceptual SoundMax™ Envelope Processing in each of the 24 bands: non-linear processing gain structure in multiple regions

Advanced Final Look Ahead Limiting with SmoothClip™: proprietary smooth windowing technology is used to minimize spectral growth (increase in high frequency energy when processing leads to saturation). This is particularly helpful for digital broadcasting. Figure 5.1.3 (a) below shows the audio

spectrogram for an AGC with conventional look ahead limiting, high frequency growth is readily obvious at the top. Figure 5.1.3 (b) shows the impact of technique used in Perceptual SoundMax™ (with much less spectral growth).

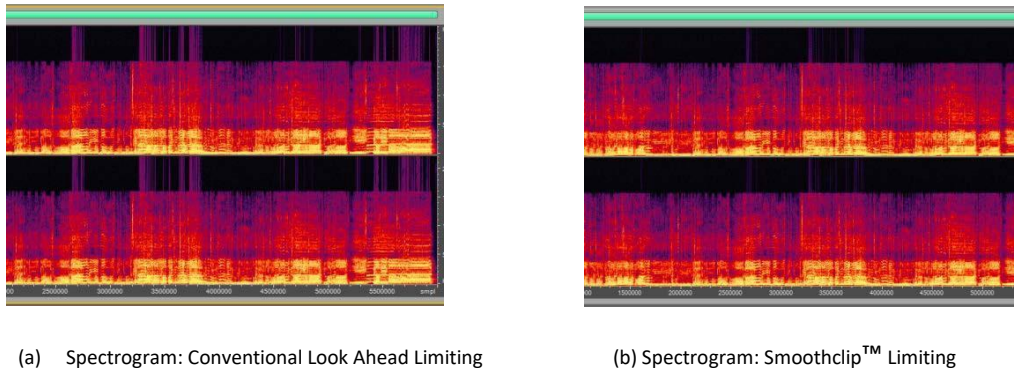


Figure 5.1.3: SmoothClip™ technology in Perceptual SoundMax™ substantially limits increase in high frequency energy during look ahead limiting

Extensive use of AI Based Audio Processing Algorithms: The Aldeal Audio™ platform embedded in Perceptual SoundMax™ utilizes AI/ML based techniques throughout its processing platform. Techniques include *sophisticated patent pending real-time low-latency ML models* for:

1. Multiband noise-profile analysis
2. Multi-Class Hierarchical Audio Classification
3. Real-time Mood analysis
4. Real-time Genre analysis

Inputs from these ML models are used to adaptively adjust processing parameters based upon changing audio characteristics.

5.1.2 Processing Architecture: *Signal Flow*

(Each module is applied or Bypassed based on profile controls)

The Processing Architecture Signal Flow is shown in Figure 5.1.4 below.

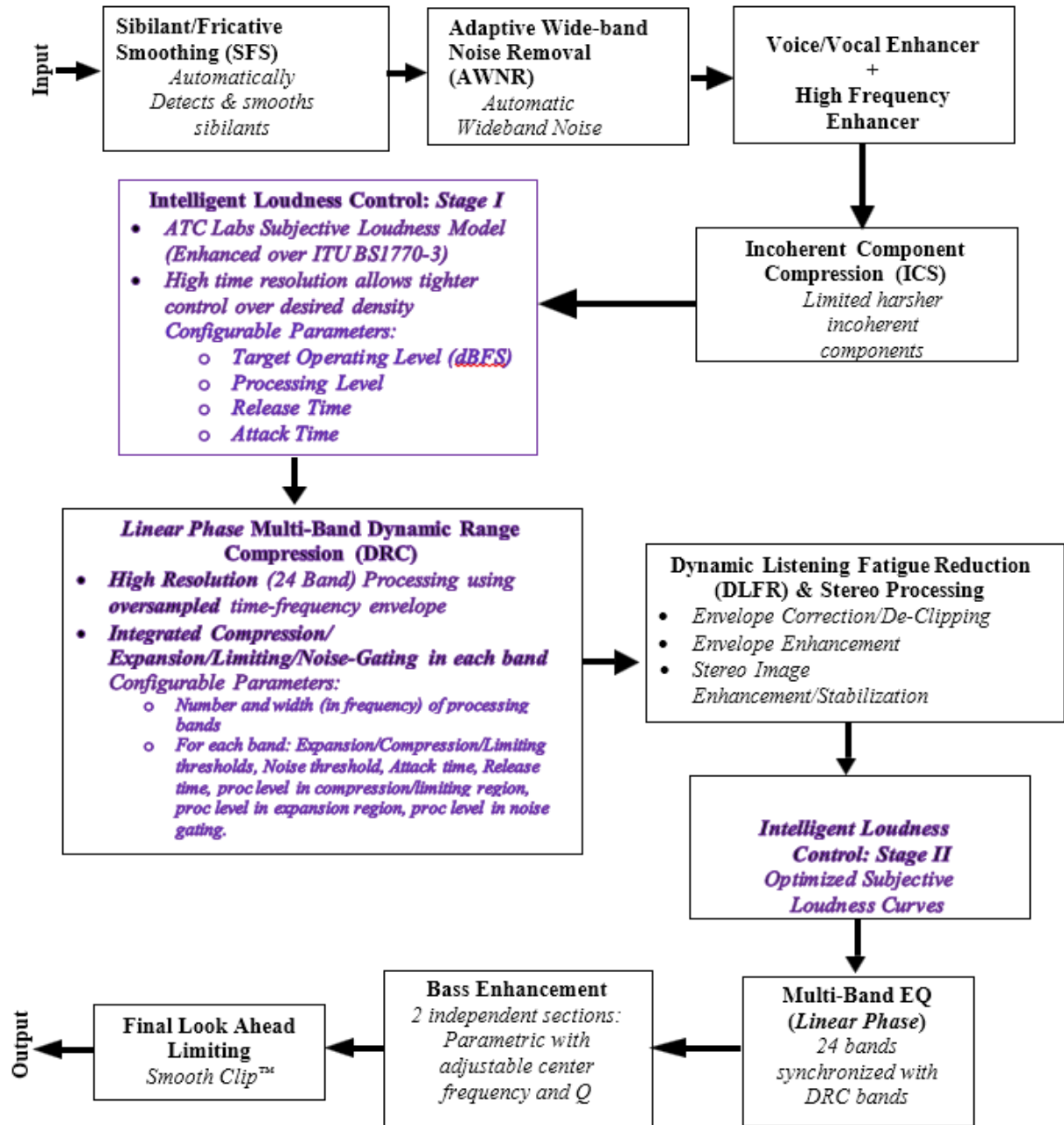


Figure 5.1.4: Processing Architecture – Signal Flow

5.2 Perceptual SoundMax™ Tuning Controls: Tuning Guide for Individual Audio Processing Modules

(Note: Profile parameter fine tunings for enhanced sound quality are best performed using SoundMaxManager™ Tuning Screen Controls; although limited tuning is also possible using the the front-panel screen as described later in Section 5.3 below)

SoundMaxManager™ offers two levels of tuning controls, namely Basic and Advanced, each is described in detail below. Each of these two control modes are available through their own GUI panel which can be accessed by clicking on the “Tuning Screen” button on the SoundMaxManager™ main GUI (Figure 4.1 -button labelled as 1); clicking on this button opens the selections panel shown in Figure 5.2.1 below:

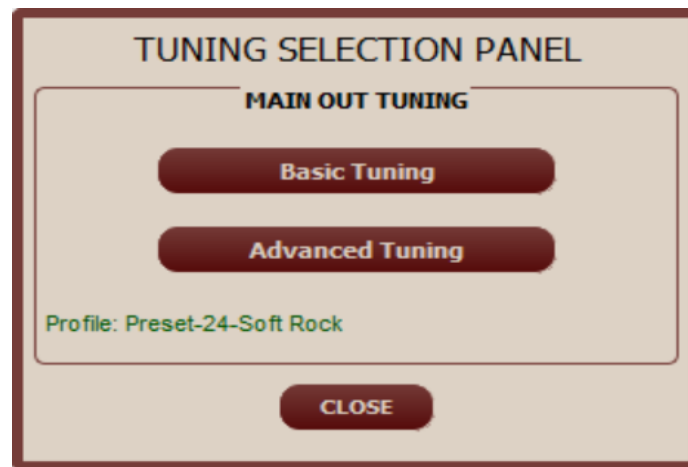


Figure 5.2.1: SoundMaxManager™ Tuning Mode Selection Screen

5.2.1 Basic Tuning Control

This section outlines the specialized controls available for shaping the characteristics and dynamic impact of audio using the Basic Tuning controls. The idea behind Basic Tuning Control option is that each control in this interface serves as a *macro-control* which is used to intelligently adjust multiple internal parameters— without burdening the user with the responsibility adjust each of these individual internal parameters. The user decides which of the macro level processing attributes to adjust upon and the corresponding internal multi-parameter adjustment achieves cohesive results with a single movement of control.



Fig 5.2.2: SoundMaxManager™ Basic Tuning screen

As seen in the Basic Tuning screen interface in Figure 5.2.2, three **macro-control** knobs are available:

(i) Full Processing

This master control knob may be used to fine tune the effect of overall processing, i.e., it impacts the entire range of audio processing parameters simultaneously. It is designed to normalize and enhance audio quality across every frequency band in a synchronized manner.

- **Increasing the knob value - Audio becomes more Energetic/Dense**
 - **Behavior:** Reduces attack and release times while increasing thresholds and gain across all bands (Note: processing parameters such as attack/release times are described in more detail below under the Advanced Tuning Control section).

- **Result:** Produces a "densely normalized" sound with controlled attacks and a significant boost in signal density and loudness consistency.
- **Recommended For:** High-energy, fast-tempo genres such as Club, Hip-Hop, Rap, and Disco.
- **Decreasing the knob value – Audio becomes more Relaxed**
 - **Behavior:** Increases attack and release times while decreasing thresholds and gain.
 - **Result:** Creates a more "relaxed" normalization, allowing for natural signal peaks and a smoother, more transparent dynamic response.
 - **Recommended For:** Acoustic or nuanced genres such as Jazz, Strings, and Soft Rock.

(ii) Vocal Processing

This control specifically targets the vocal components within the audio signal, allowing you to manage the presence and loudness of vocals independent of the background music.

- **Increase:** Boosts the loudness and clarity of the vocals. Use this setting if the singing or dialogue feels slightly recessed and needs to stand out more prominently against the music.
- **Decrease:** Softens the vocal presence, making them less prominent. Use this setting if the vocals are overpowering the instrumental elements and you prefer a more blended, "laid-back" vocal mix.

(iii) Instrument Processing

This slider focuses exclusively on the non-vocal (musical) instrumental components. It manages the normalization and impact of the instruments without affecting the vocal range.

- **Increase:** Enhances the loudness, details, and definition of the music. This makes the instrumental elements feel more "forward", and each small instrumental sound such as string pluck becomes more noticeable.
- **Decrease:** Reduces the aggression of musical attacks. This creates a softer instrumental bed, which is ideal when you want the music to be supportive and unobtrusive rather than the primary focus.

Note: For the remaining controls shown in Fig. 5.2.2, refer to Section 5.2.2 (Advanced Tuning Control). These sliders and checkboxes function in the same way as described in the Advanced Tuning section; since adjustments to these parameters are more commonly made, these are also exposed and made adjustable on the Basic Tuning Control panel. These include: Intelligent Loudness Control, Initial Limiter Attenuation, Final Limiter Value, and the Preemphasis control.

5.2.2 Advanced Tuning Control:

This section provides more detailed overview of the various audio processing tools available in the Perceptual SoundMax™ audio processors and guidance on how to use them effectively to achieve highest-quality processed audio. The SoundMaxManager™ Tuning Screen is shown in Figure 5.2.2. The audio processor offers multiple processing modules, with several tunable parameters, which

can be adjusted individually or in combination to create an even more enhanced desired signature sound for your broadcast. As noted earlier, the user should attempt to use one of the existing Preset Profiles first to achieve desired sound, further sound quality tuning may then be undertaken after a careful study of this section.

Advanced Tuning Screen of SoundMaxManager™ is accessed through the tuning mode selection screen of Figure 5.2.1 - pressing on the Advanced Tuning button on that screen bring the user to this screen as shown in Figure 5.2.3. There are about 1000 parameters exposed on this screen, and can be tuned independently for each of the signal path (such as MAIN and STREAM on all models and HD1 on Perceptual SoundMax™ Q24-1111HD). After tuning, the profile may be saved as a *custom profile* and is available for application to all the signal paths. Moreover, in setups incorporating more than one Perceptual SoundMax™ audio processors, custom profiles created on one processor may also be exported and made available on the other processors.

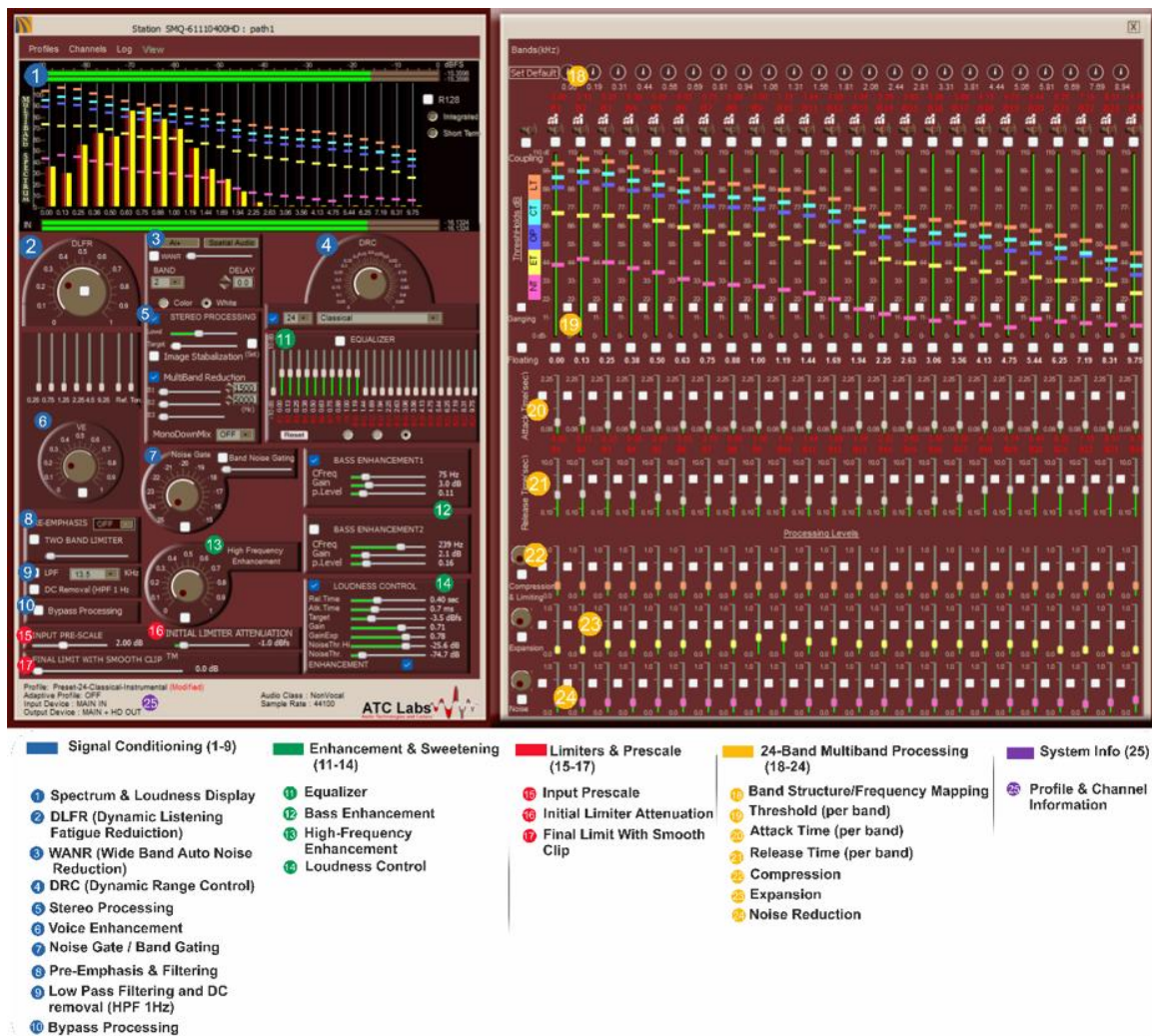


Figure 5.2.3: Perceptual SoundMaxManager™ Advanced Tuning Screen

As described above following different processing modules are available:

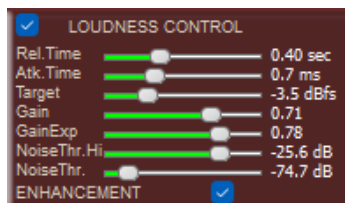
- AI based processing control.
- Intelligent Loudness Control
- Multi-Band Dynamic Range Compression (DRC) – 24 Bands
- Base Enhancement – 2 Sections
- Multi-Band Graphic Equalizer (24 bands)
- Vocal Enhancement (VE)
- Stereo Processing
- Incoherent Component Suppression (ICS)
- Sibilant/Fricative Smoothing (SFS)
- Wideband Automatic Noise Removal (WANR)
- Dynamic Listening Fatigue Reduction (DLFR)
- Pre-Emphasis
- Low Pass Filter
- Initial & Final Look Ahead Limiting
- Input Pre-scale

These different tools when tuned judiciously can provide even further sound quality enhancement such as more consistent, clear, pleasant, high impact audio, and offer the broadcast engineer flexibility for creating a compelling signature sound for each service channel including digital channels operating at different bit rates.

Tuning Guide for Individual Audio Processing Modules

This section provides a description of the impact of different parameters in each of the processing modules. Each module is invoked or bypassed by checking or unchecking the master control box in that module.

- **Intelligent Loudness Control (ILC):**



This module continuously estimates instantaneous subjective loudness of the audio signals and adjusts it to the configured target level. It is one of the key processing modules for ensuring consistent loudness for broadcast and streaming. In the case of digital broadcasts and streaming a consistent loudness envelope also helps perceptual audio codecs better mask the compression artifacts. ILC uses a proprietary enhanced subjective loudness measure optimized for stereo signals. The parameters in this module include:

- **Target:** target level in dB LKFS that ILC strives to achieve (typical range -2 to -4 dB LKFS; high value e.g. -0.5 may result in increased clipping).
- **Release Time:** A key parameter; lower values (i.e. dragging the slider to the left) results in higher density for the processed signal – i.e. a fuller or fatter sound which is more consistent to the listener and better masks compression artifacts. It controls how fast the estimated loudness, used in computing processor gain, responds to the actual audio signal envelope on a waning trend.

- **Attack Time:** Higher attack time (slider to the right) leaves more dynamics in audio signal attacks at the cost of increased modulation index or bit demand variation. A good compromise in many situations may be around the middle of the slider (about 1.2 msec). It controls how fast the estimated loudness responds to the actual audio signal envelope on a rising trend.
- **Gain:** Control the effectiveness of the ILC module on the *compressive* side. This should be used in conjunction with the GainExp parameter below. Good loudness control is achieved in the 0.7-0.8 range for both these parameters, while more aggressive control is achieved by setting the gains in the 0.85-0.95 range.
- **GainExp (Gain Exponent):** Determines how strongly the ILC responds as the input loudness deviates from the target when the estimated loudness is lower than the target, i.e., on the *expansion* side. Lower values produce gentler, more open correction; higher values produce a tighter, more assertive loudness control at the expense of audible compression. A typical operating range is **0.6 – 0.9**, with **0.75 – 0.80** providing a good balance between transparency and consistency for general broadcast material. If the user wants to achieve higher loudness but a more open attack dynamics, GainExp may be set to be higher than the Gain parameter (compressive control) described above.
- **NoiseThr.Hi (Noise Threshold High):** Defines the upper edge of the noise-threshold to expansion region transition, the processing gain is unity at noise threshold (determined by the parameter below), starts rising at a gentler pace till ILC kicks in at full strength at **Noise Threshold High**. Audio levels above this threshold are treated as most prominent program content and are processed at full ILC effectiveness. The gap between this threshold and **NoiseThr** (low) determines how gracefully the ILC fades into and out of action as program level falls toward noise or silence. A typical setting is in the range **–25 dB to –30 dB**. A lower **Noise Threshold High** value (leftward movement of slider) leads to a fuller/denser sound. The displayed value in the example is **–25.6 dB**.
- **Noise Threshold:** Loudness control processing is in a gating mode if estimated instantaneous signal loudness is below **Noise Threshold** and returns to a linear level (i.e. becomes ineffective) around the noise threshold. Typically, it may be set to -70dB or lower. But if there is persistent background noise in the source, setting the noise threshold to a higher value of -60 dB or higher (i.e. moving the slider to the right) results in lower noise pumping.
- **ENHANCEMENT:** A switch (checkbox) that enables an additional perceptual-loudness enhancement stage downstream of the main ILC processor. When enabled, this stage provides supplementary subjective-loudness lift and improved consistency on program material with a wide short-term dynamic range — e.g. dialogue-heavy news content or live sports commentary. It can be left enabled for most broadcast applications and disabled where maximum transparency to the source is desired (e.g. classical or jazz formats). In the example shown, ENHANCEMENT is **enabled**. If ENHANCEMENT is **disabled**, the **Noise Thr. Hi** slider and **GainExp** slider are disabled and: (i) single Gain value is used for both compressive and expansive controls as determined by the **Gain** slider, and (ii) ILC kicks in at full strength from **Noise Threshold** and above.

- **Dynamic Range Compression (DRC):**

Multiband Dynamic Range Compression (DRC) is an integrated 24-band Dynamics Processor allowing for compression, limiting, expansion, and noise thresholding in 24 distinct frequency bands. Driven by the



principles of Psychoacoustics, it employs completely linear phase signal processing. Intelligent control over the operating points using a desired spectral balance or a signal-adaptive characteristic is employed. To change the dynamic range in different bands individually, this module is provided with greater functionality than any other product available in the market. At the heart of the

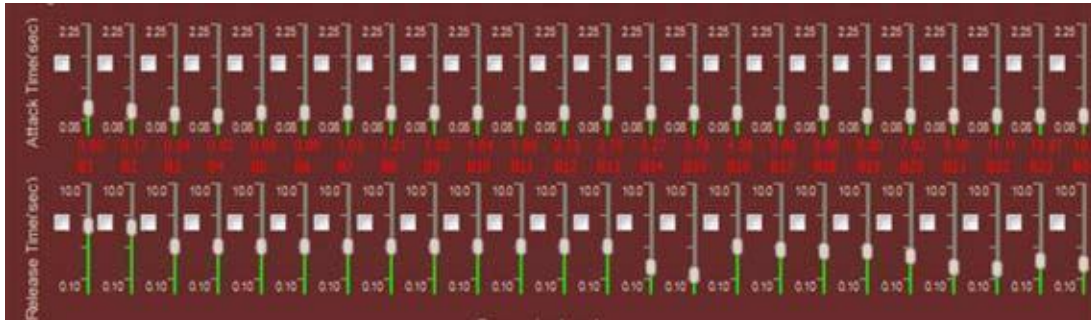
processor is a specially designed filter bank for generating an extremely detailed time-frequency representation of the signal. Available controls in DRC include:

- **Limiting/Compression/Expansion/Noise Thresholds for each of the 24 bands:** the 4 thresholds on each band are placed on a single slider with the relationship Noise Thr < Expansion Thr < Compression Thr < Limiting Thr. These thresholds can range 0 to 90+ dB (depending upon the band) with the highest frequency band limiting the range to about 80 dB. For tuning it help to keep the noise thresholds at the bottom initially and raise these judiciously in chosen bands where noise gating is desired. Furthermore, compression and limiting thresholds can be kept very high initially (i.e. almost eliminating the effect of compression and limiting) and focus on the tuning of expansion thresholds to achieve a desired spectral balance and brightness characteristics (raising the expansion threshold in a specific band, results in making that band “louder”). The compression threshold can then be lowered achieve desired level of control in each frequency region following by limiting threshold.



The interface allows several options to ease tuning: (i) bands can be ganged together such that all the thresholds for ganged bands move together, (ii) all the thresholds for a specific band may be locked to each other by checking the coupling box on top of that slider – when coupled just by moving the operating point in that band all the thresholds move up or down together while maintaining the relative difference between the thresholds, (iii) there is an additional float option for each band which at times works well for an operator not fully trained in tuning the relative values of thresholds across frequency. When a band is floated its operating point is continuously estimated over time and all the 4 thresholds values are with respect to this “floating” operating point, (iv) just below the DRC master dial there is a drop down list of a few pre-configured thresholds match to different genre of music, this may be used to get an initial configuration of thresholds.

- **Release Time and Attack Time in each of the bands:** Release Time is a key parameter for achieving density, with the difference that it is a band specific parameter in DRC (as opposed to wideband parameter in ILC). Lowering the release time in a band makes the audio more controlled and denser in that band. So, for example, to make the vocal steadier and more prominent, it helps to lower the release times in the mid-frequency range (about 800 Hz to 3 kHz). The Attack Time in contrast can be used to control the level of dynamics in a band - a higher value results in more dynamics, while a lower value reduces pumping and increases cleanliness of the sound at higher expansion processing level for the band (see below). As with the thresholds, the interface allows the ganging of multiple bands for convenient tuning.



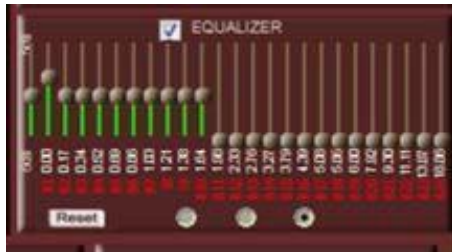
- **Processing Levels:** The interface allows for independent control of processing levels of DRC for each band, in three regions of non-linearity (Figure 5.1.2): (i) compression/limiting, (ii) expansion, (iii) noise-gating. Correspondingly there are 24 sliders for compression/limiting processing level, 24 sliders for expansion processing level, and 24 sliders for noise gating processing levels. The interface allows for ganging of bands and offers a master dial for each of the 3 sets of sliders. Also, changing the master DRC dial impacts all processing levels equally (with saturation of individual values for maximum/minimum).



- **Center Frequencies of the processing bands:** The interface allows for the center frequency of each band to be increased or decreased using a set of dial (with the constraint that center frequency of each band is upper/lower bounded by respectively the adjacent higher/lower band).



- **Multi-Band Graphic Equalizer:**



This module further equalizes the audio in 24 bands according to the 24-slider setting and offers the option for additional per band gain control post **DRC** (or as the primary multi-band gain control mechanism if DRC is not engaged, i.e., set to OFF). If a slider is set to the middle of the range (0 dB) the equalizer is flat in that band. Each slider can be moved between -5dB and 5dB.

There are three coupling options in the form of Soft keys, namely Tight Coupling, Loose Coupling and No Coupling. The center frequency of the equalizer bands is aligned with the center frequency of the DRC bands B1-B24.

- **Bass Enhancement:**



Perceptual SoundMax offer the options for using up to 2 stages of parametric bass boost. Either one or both stages may be invoked by enabled either “Bass Enhancement1” or both “Bass Enhancement1” and “Bass Enhancement2” sections on the interface. For each of the 2 bass boost stages the following parameters may be set:

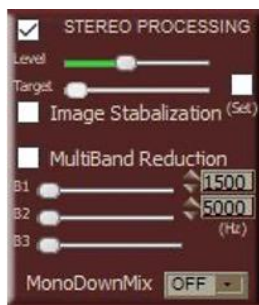
- **CFreq:** this is the center frequency of the bass boost notch or peak (in Hz)
- **Gain:** this specified in dB controls the height of the bass boost notch.
- **P. Level:** This processing level controls the width (i.e. Q-value) of the bass boost notch/peak; a higher level corresponds to a wider peak.

- **High Frequency Enhancement:**



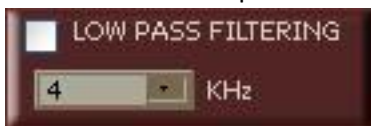
This control enhances the higher frequencies and provides enhanced treble effect.

- **Stereo Processing:**

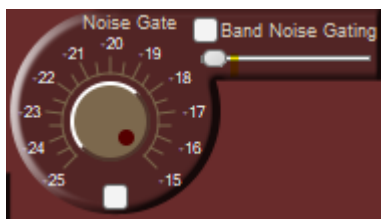


This module controls the stereo balance in the audio. It may widen or shrink the stereo image depending on the characteristics of the stereo audio and the slider setting – i.e. it serves as a dynamic range control for the stereo image. This tool helps in maintaining a consistent stereo image which would make audio sound pleasant. A new user can safely set the Level slider at 0.3. Additional controls are also available:

- **Level:** This is used to set the level of stereo image reduction - if the **Image Stabilization** control as described below is OFF; or the level of stereo dynamic range control - if the **Image Stabilization** control as described below is ON.
- **Image Stabilization:** if this box is checked, stereo image is also expanded, otherwise the stereo image processing is purely compressive. Therefore, if this feature is invoked the stereo image may at times be wider than the original audio and remains more stable, losing some of its dynamics. If **Image Stabilization** is enabled then the **Target** slider above it may also be enabled and set to a desired *fixed target*, else that control remains ineffective.
- **Multiband Reduction:** This is a 3 band stereo image reduction module. When enabled, the bandwidth of the 3 bands, B1-B3, can be set using the combo boxes and level of reduction controlled by the 3 sliders.
- **Mono Downmix:** Mono downmix from Sum/Left/Right is available by invoking this module.
- **Target/Set:** Normally, the stereo processing is with respect to an adaptive smooth estimate of stereo width. However, optionally this can be done with respect to a fixed target which is set using the Target Slider (and this functionality is invoked by checking the “Set” box on the interface).
- **Low Pass Filter:** Optional low pass filter. Set any cutoff frequency limit between 4-20 kHz.



- **Noise Gating/Band Noise Gating:**

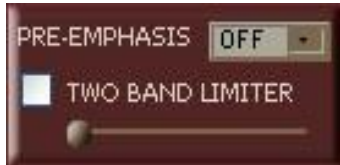


The Noise Gate module configures the noise gating functions, providing both wideband noise gating and multi-band noise gating controls. The wideband **Noise Gate** may be invoked from the checkbox below the dial, and its application strength is configured using the dial; this algorithm is used to prevent noise boost in signal *valleys/tails*, i.e., in segments where signal falls

into silence and a background noise is present. When the **Band Noise Gating** is enabled, it invokes a sophisticated 3-band AI/ML algorithm for determining noise-like behavior of signal in those bands; this determination is then used to adopt multi-band processing parameters in those bands so that signal enhancement is achieved without any increased level of noisiness. The available controls are as below:

- **Noise Gate** Checkbox (below the dial) – it is used to enable or disable the wideband noise gate.
- **Noise Gate** Dial – This sets the wideband noise gate threshold level. When the input signal falls below the threshold, the gate adjusts the processing algorithms so that noise during the silent segment is not boosted and is instead suppressed during the application of the loudness control algorithm.
- **Band Noise Gating** Checkbox – It is used to enable or disable multi-band noise gating.
- **Band Noise Gating** Slider - The slider controls the intensity of the gating algorithm used to adjust the processing parameters across the 24 frequency bands, this allows for frequency-selective noise floor management using specialized AI/ML algorithms.

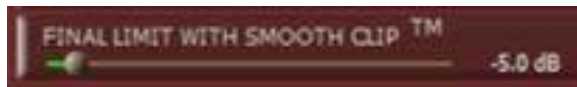
- **Pre-Emphasis:**



Pre-Emphasis is essential for FM broadcasting. It is typically set to a parametrized value of either 50 μ s or 75 μ s depending upon the geographical region. For North America a Pre-Emphasis curve of 75 μ s is specified, while India/European Union/UK and many other regions

specify a pre-emphasis curve of 50 μ s in FM broadcasting. Choose a value that matches your geographical region. For hardware products, this is set from the LCD controls as described in previous sections and this interface simply displays that chosen value. There is also an available 2-band limiter, which may be carefully enabled, with listening verification, to further optimize modulation index and loudness for FM broadcasts.

- **Final Limiting Volume & Initial Limiter Attenuation:**



This slider displays/sets the final level for the look ahead limiter. For the MAIN path in hardware processors, the final limiter value is set from front LCD controls as described above and this slider will simply follow or display the value set there. For streaming and digital broadcast signal paths a value less than peak 0 dBFS is desirable depending upon the headroom required by the codec in use. For FM broadcasting the level should be chosen to control the desired peak modulation index.

The implemented look ahead limiting incorporates a SmoothClip™ algorithm for saturation using proprietary smooth windows which minimize spectral growth due to saturation. The algorithm may be aided by the application of an **Initial Limiter Attenuation** using an available additional control. This



specifies an attenuation level that is applied to the signal as a final touch-up before it hits the SmoothClip™ look ahead limiter and it can help in preventing excessive saturation or density,

leading to a cleaner sound. A typical value of -1 to -2 dB for the initial limiter attenuation is recommended.

- **Bypass Processing:**



By checking the bypass processing button, the processing done in the processor is bypassed. This helps the user judge the quality of audio before and after processing instantaneously by switching between modes.

- **Wideband Automatic Noise Removal Toolkit (WANR):**



The audio processor comes bundled with broadcast quality real-time automatic noise control block that reduces wide-band noise without excessive gating (hence maintaining audio fidelity). The user can choose a few optional parameters and background noises, including dynamic background noises, are automatically identified on the fly and removed from the audio.

At a broad level the user can operate the algorithm with few additional controls:

- **BAND:** The algorithm may be applied as a 1 band algorithm or as a 2 band algorithm (low/high with a 5 kHz cutoff)
- **DELAY:** Allowing some delay in the algorithm (value specified in seconds) helps improve the background noise estimation but at the cost of increased latency.
- **White/Color Model:** For general noise types specify a **White** noise model, while for machine and/or instrument type noise specify a **Color** noise model.

- **Voice/Vocal Enhancement (VE):**

This module enhances the speech/vocal content of the audio. It uses an accurate harmonic analysis of the input audio and effectively enhances the vocal content. Using this information input audio sounds cleaner with improved presence. This module is also useful in sports/news broadcast to make the commentators voice more intelligible, where high background audience noise exist.



- **Dynamic Listening Fatigue Reduction (DLFR):**



This is an audio sweetening tool that conditions the temporal envelope of the audio signal in multiple frequency bands. The 6 vertical bars below the dial provide finer control over envelope conditioning in individual frequency bands (with indicated center frequency values). The higher the sliders, the more conditioned the audio in that band is. There are two associated slider controls which control respectively the tonal characteristics of overall audio and the refinement level of envelope conditioning.

- **SpatialMax™ Spatial Audio:**



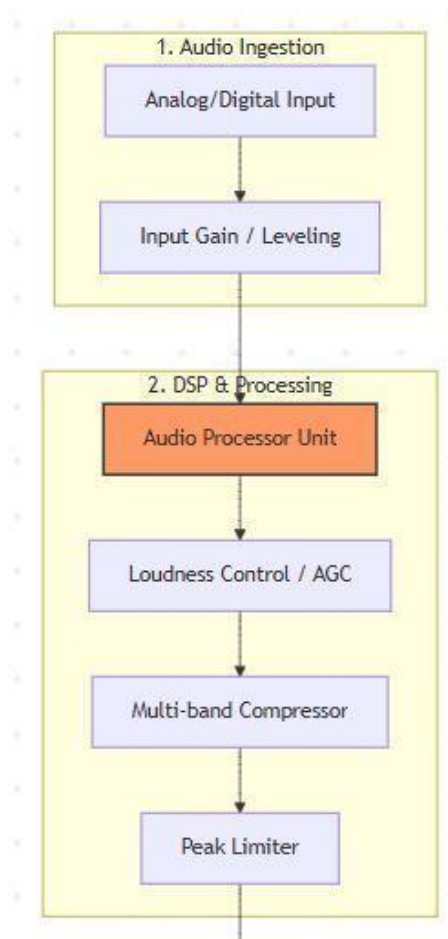
Perceptual SoundMax™ product offers an integrated & powerful *audio spatialization* tool that may be used to expand the sound stage. This panel is used to configure this algorithm. This Spatial Audio Settings panel is accessed by clicking on the Spatial Audio button next to WANR (button 3 in Figure 5.2.3).

The available controls are:

- **OFF/ON:** The
- **Model:** Currently two algorithm models are offered – invoking the **MASS3** model results in a wider if a bit more uncontrolled sound stage; while **MASS8** model leads to a bit more defined and controlled soundstage.
- **Vocal Control:** This slider may be set between 0 to 1; higher values lead to more controlled/cantered vocals.

5.3 Streaming Protocols (Coding Formats AAC/MP3, HLS Streaming, CDN with HTTP Push/Pull)

To meet the bandwidth demands of internet streaming, high compression efficiency and high perceptual quality compression technology is essential. Perceptual SoundMax™ Models Q24-1111 and Q24-1111HD integrate quality optimized encoding algorithms – such as LC/HE-AAC compressions algorithm enhanced with ATC Labs industry leading Psychoacoustic Modeling and Signal Analysis know-how. Furthermore, Perceptual SoundMax™ audio processing incorporates unique capabilities in terms of optimizing the sound quality of compressed audio. Audio processing becomes an important building block of Internet streaming architecture implemented in Perceptual SoundMax™. Figure 5.3.1 shows the various steps in this quality optimized internet streaming process. After processing, compression, and encapsulation audio is transmitted on Internet using quality standard codecs and protocols. AAC and MP3 are well established standard based audio compression technologies available in the market which are optimized and compatible with all online audio players running on target devices chipset. A whole range of bit rates and *scalable* coding is supported in Perceptual SoundMax™ which can then be tailored based on use case specifications and targets. This section describes the Codecs technology and streaming protocols used in the product.



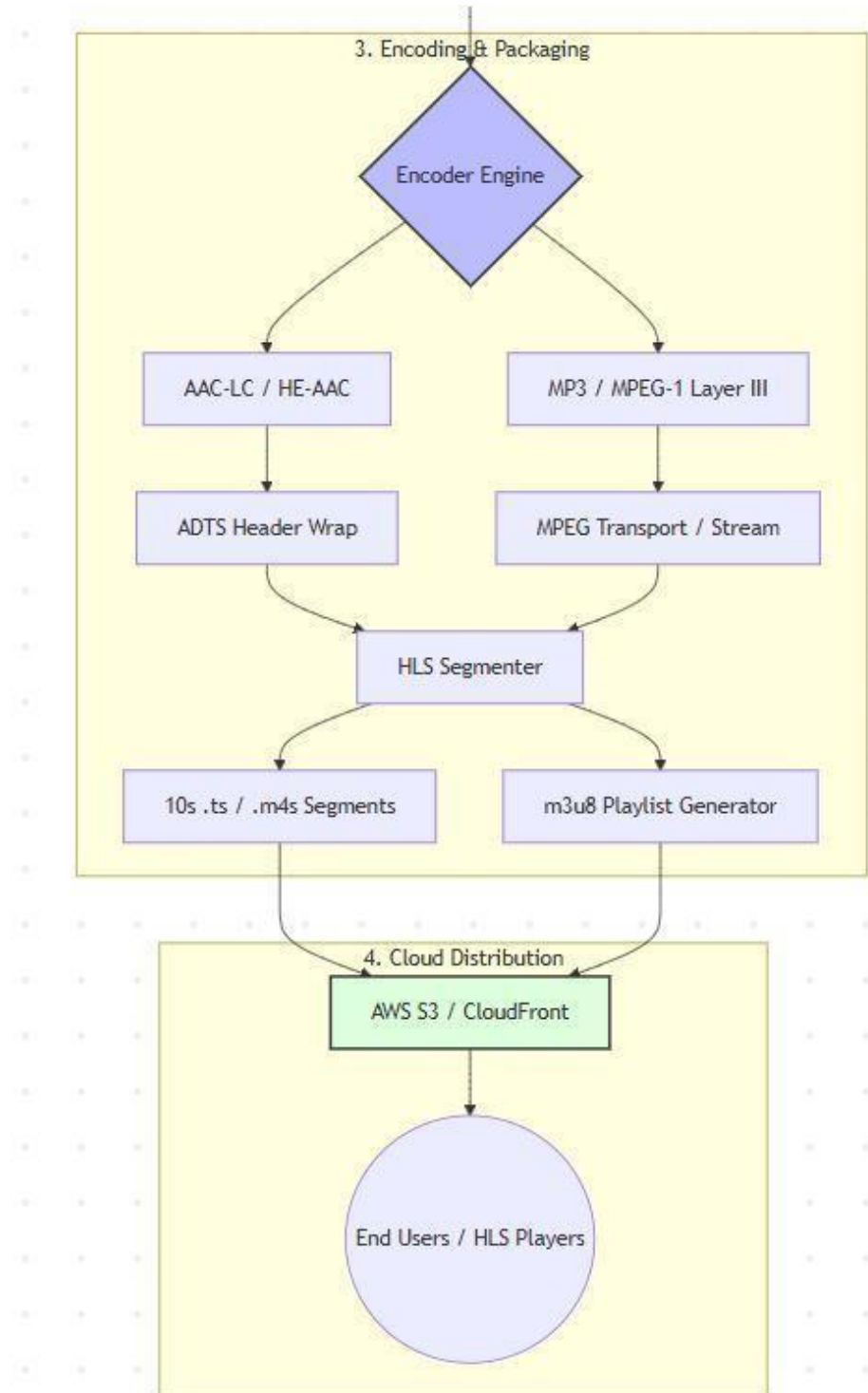


Figure 5.3.1: Internet Streaming Signal Flow in Perceptual SoundMax™

5.3.1 Quality Optimized AAC-LC and AAC-HE (V1/V2) Algorithms

AAC is the latest and widely implemented standard codec used in audio compression which provides highest perceptual quality at relatively moderate bitrates.

Algorithmic Overview of AAC-LC Encoding (ISO/IEC 14496-3) is as follows

1. Time-to-Frequency Transformation: The MDCT

Unlike a standard Fourier Transform, the **Modified Discrete Cosine Transform (MDCT)** is designed specifically for gapless audio.

- **Overlapping Windows:** The algorithm uses a "50% overlap." If a window is 2048 samples, it shares 1024 samples with the previous block and 1024 with the next.
- **Time Domain Aliasing Cancellation (TDAC):** This is the "magic" of MDCT. By overlapping, it cancels out the errors (aliasing) created at the edges of the windows. This is why HLS stream won't have "clicks" between the 10-second segments—the waveform remains mathematically continuous.
- **Block Switching:** For steady music, it uses **Long Blocks (2048 samples)** for high frequency resolution. For sudden transients (like a snare drum), it switches to **Short Blocks (256 samples)** to prevent the "smearing" of the sound in time.

2. The Psychoacoustic Model (The "Filter of Perception")

This stage determines what information is "audible" versus "irrelevant." It calculates two primary phenomena:

- **Simultaneous Masking:** If a loud bass note (masker) is present, the ear's threshold of hearing for nearby frequencies is raised. The model calculates a **Masking Threshold**; any signal below this line is assigned zero bits during quantization.
- **Temporal Masking:** A loud sound masks quieter sounds that occur immediately *before* (pre-masking) and *after* (post-masking) it. The algorithm uses this to "hide" quantization noise behind loud transients.
- **SMR (Signal-to-Mask Ratio):** The goal is not to eliminate noise, but to ensure that the quantization noise floor is always lower than the masking threshold.

3. Advanced Quantization & Bit Reservoir

AAC uses a **Non-Uniform Quantizer**, specifically a 3/4 power-law compressor.

- **The Logarithmic Advantage:** Humans are more sensitive to changes in quiet sounds than loud ones. The 3/4 power-law allocates more precision to low-amplitude signals and less to high-amplitude ones, mimicking the logarithmic nature of human hearing.

- **Bit Reservoir Logic:** HLS usually requires **CBR (Constant Bitrate)** for predictable AWS bandwidth. However, audio complexity varies.
 - *Simple frames (silence/pure tones):* Use fewer bits than the target.
 - *Complex frames (applause/orchestra):* These pull the "saved" bits from the reservoir.
 - This ensures that a 10-second HLS segment maintains a steady data rate while allowing "bursts" of quality where the music demands it.

4. TNS (Temporal Noise Shaping)

TNS is the primary reason AAC sounds better than MP3 at low bitrates.

- **The Problem:** In the frequency domain, quantization noise is spread evenly across a time window. On a sharp "click," this noise can start *before* the click occurs (Pre-echo).
- **The Solution:** TNS applies a linear prediction filter to the frequency coefficients. This effectively "pushes" the quantization noise to be exactly under the loud part of the sound in the time domain.
- **Result:** Percussive sounds stay "crisp" and don't sound "mushy" or "pre-chirped."

High-efficiency profile of audio processor, **Spectral Band Replication (SBR)** is used. This is the "magic" behind **HE-AAC (v1)** that allows for high-fidelity audio at bitrates as low as 32kbps—a common requirement for mobile-optimized HLS streams on AWS.

Algorithmic Overview of SBR (Spectral Band Replication)

SBR is not a standalone codec; it is a **bandwidth extension** technology. In a standard encoder, low bitrates force a "brick-wall" filter (usually around 4Khz to 10Khz) to save data, resulting in "muffled" audio. SBR eliminates this by splitting the encoding process into two distinct parts.

1. The Split-Band Architecture

- **The Core (Baseband):** The lower frequencies (typically 0Khz to 5Khz) are encoded using the standard **AAC-LC** algorithm described previously. This ensures that the most critical audio information (vocals, fundamental instrument frequencies) is captured with high precision.
- **The High-Band (Replication):** Instead of encoding the high frequencies (5Khz to 20Khz), the encoder analyses them and generates a small set of **side-chain metadata** (only about 2-3kbps).

2. The Reconstruction Algorithm

When the HLS player (client-side) receives the segment, the SBR decoder performs the following steps:

- **Transposition:** It takes the decoded low-band harmonics and "transposes" (copies) them up into the high-frequency range.
- **Envelope Adjustment:** It applies the side-chain metadata to the transposed high-band. This ensures the energy levels of the high frequencies match the original source.

- **Noise/Inverse Filtering:** SBR can inject "synthetic noise" or "sine addition" into the high-band to reconstruct non-harmonic sounds like atmospheric air or percussive brushes that aren't present in the low-band harmonics.

3. QMF (Quadrature Mirror Filter) Bank

SBR operates in the **QMF domain**, not the MDCT domain.

- The signal is passed through a 64-channel QMF bank.
- This allows the algorithm to have high frequency resolution for the replication process while maintaining low computational latency—critical for live HLS streaming where the processor must maintain real-time performance.

ADTS Encapsulation & Header Structure

Since product uses HLS, the AAC bitstream must be "packetized" so it can be decoded starting from any 10-second segment. We use the **ADTS (Audio Data Transport Stream)** format.

ADTS Fixed Header Structure (7 Bytes):

- **Syncword (12 bits):** Always 0xFFF. This allows the AWS-hosted stream to be "found" by the player immediately.
- **ID (1 bit):** 0 for MPEG-4, 1 for MPEG-2.
- **Layer (2 bits):** Always 00.
- **Protection Absent (1 bit):** Indicates if a CRC is present for error checking.
- **Profile (2 bits):** 1 for AAC-LC, 4 for HE-AAC (SBR).
- **Sampling Frequency Index (4 bits):** Indicates 44.1Khz, 48Khz, etc.
- **Channel Configuration (3 bits):** Mono, Stereo, or 5.1.

Algorithmic Overview of Unique ATC Labs Enhancements to Standard Compliant LC/HE-AAC Encoder

The AAC codec used in Perceptual SoundMax™ is further enhanced using ATC Labs expertise in signal processing and perceptual modelling. Specifically, our codecs incorporate:

- **Accurate Harmonic Analysis Models:** accurate frequency estimation, and identification of signal harmonics is required in many application scenarios including in optimally setting the SBR parameters in the HE-AAC codec. Our research developed new results improving by a factor of 10 the accuracy of a previously known frequency estimation algorithm, thus improving the accuracy of complex signal harmonic estimation. This was then successfully deployed to improve the performance of bandwidth extension algorithms such as SBR.
- **Sophisticated Wide-Band Psychoacoustic Models:** As noted above quantization in AAC encoding is based on a psychoacoustic model which is typically based the principle that for complex signals masking pattern due to narrow band maskers is combined using the principle of superposition in one form or other. However, research suggests the Masking Threshold for a wide band masker (>critical band) can be substantially lower than a narrow band masker. This

led ATC Labs to develop a high accuracy wide-band psychoacoustic model that has led to increased efficiency and quantization noise control in our codec.

5.3.2 MP3

While AAC is the modern choice for HLS, the Perceptual SoundMax™ provides robust **MP3 support** for legacy hardware and broad compatibility. The MP3 algorithm is a popular compression mainstay incorporating some of the best early advances in digital signal processing and balancing computational limits with acoustic transparency. Key components of MP3 encoders are as below:

1. The Hybrid Filter Bank (The Analysis Stage)

MP3 is unique because it does not perform a single mathematical transform. It uses a two-stage hybrid approach:

- Sub-band Filtering (PQMF): The input signal is first split into 32 sub-bands of equal width using a Polyphase Quadrature Filter. This mimics the "critical bands" of human hearing but is computationally cheaper than a full high-resolution transform.
- Secondary Transform (MDCT): Each of the 32 bands is further subdivided using the Modified Discrete Cosine Transform. This increases frequency resolution to 576 frequency lines.
- The Trade-off: By using this hybrid structure, the encoder can handle "Steady State" audio (like a violin note) with high frequency detail but switch to "Short Windows" during "Transients" (like a drum hit) to prevent Pre-echo artifacts.

2. Psychoacoustic Modelling & Huffman Coding

This stage determines which data is "expendable":

- Masking Thresholds: The model identifies loud frequencies that "hide" quieter, nearby frequencies. The encoder then allocates zero bits to those hidden signals.
- Huffman Coding (The Final Crunch): After quantization, the data is compressed without loss. It analyses the bit patterns; a pattern that appears 1,000 times gets a very short code (e.g., 01), while a rare pattern gets a long code. This is why MP3 files have varying levels of internal efficiency.

3. Joint Stereo: MS (Mid-Side) Processing

In standard stereo, Left and Right channels are encoded independently, often wasting bits on identical information.

- Mid (M): $S_L + R_S$ (The mono center information).
- Side (S): $S_L - R_S$ (The directional/spatial information).
- Efficiency: In most broadcasts, the "Side" signal is very quiet. The Q241111 allocates the majority of the bit reservoir to the "Mid" channel, ensuring the core audio remains crystal clear while using minimal data for the stereo width.

4. MPEG-TS: The "Engine" for HLS Delivery

HLS (HTTP Live Streaming) requires that every 10-second segment be a "complete" story that a player can understand instantly. Since a raw MP3 file is just a continuous stream of bits, it lacks the structure needed for web-based "chopping."

A. PES (Packetized Elementary Stream)

Raw MP3 data is "Elementary." To make it "Transportable," it is wrapped in PES headers:

- **PTS (Presentation Time Stamp):** This is a 33-bit value that tells the player exactly when to play that specific frame relative to the start of the stream. Without PTS, 10-second segments would drift out of sync, causing "jitter" or "lip-sync" issues in long-duration streams.
- **DTS (Decoding Time Stamp):** Tells the hardware decoder when to begin processing the data.

B. MPEG-TS Muxing (The Container)

The PES packets are then broken into small 180-byte packets (Transport Stream).

- **PID (Packet Identifier):** Every packet in your 10-second .ts segment has a label. The Audio PID (often 0x11 or 0x101) tells the AWS-hosted player, "This packet contains the music."
- **Continuity Counter:** A 4-bit field that helps the player detect if a packet was lost during the upload to the S3 bucket or during the download by the user.

5.3.3 RDS Overview

The Radio Data System (RDS) is a standardized communication protocol (IEC 62106) for embedding small amounts of digital metadata into conventional FM radio broadcasts. It operates on a 57 kHz subcarrier, allowing receivers to display station names, song titles, and automatically switch to stronger signals for the same network. Figure 5.3.2 shows the RDS Insertion Overview.

Sender Architecture (Encoder)

The sender's primary role is to format digital data into groups and modulate it onto the FM baseband signal.

Functional Blocks

- **Data Formatting:** Raw input (Station Name, PI Code, Traffic Flags) is organized into 104-bit Groups. Each group is split into four 26-bit blocks.
- **Error Protection:** For each 16-bit data word, a 10-bit Checkword is calculated to allow for error detection and synchronization.
- **Differential Encoding:** The bitstream is encoded so that the data is represented by changes in state, making the signal more robust against phase inversions.
- **Biphase Modulation:** To prevent interference with the 19 kHz stereo pilot, the signal is shaped into a biphase symbol.

- **Subcarrier Injection:** The signal is modulated using Suppressed Carrier Amplitude Modulation at 57 kHz and mixed with the mono and stereo audio signals into the FM Multiplex (MPX) chain.

Key Transmission Parameters

- **Data Rate:** 1187.5 bits/s (Fixed).
- **Modulation Type:** BPSK (Binary Phase Shift Keying).
- **Subcarrier:** 57 kHz (Phase-locked to the 19 kHz pilot for stereo broadcasts).

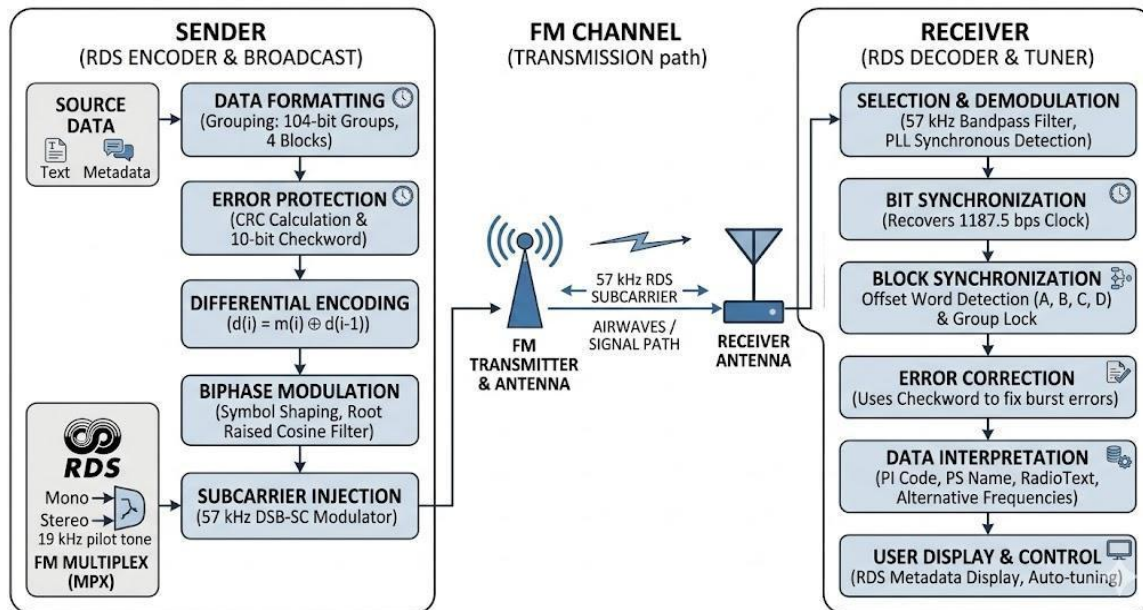


Figure 5.3.2: Integrated Signal Flow Diagram

Receiver Architecture (Decoder)

The receiver identifies, corrects, and displays the transmitted data.

Functional Blocks

- **Selection & Demodulation:** A bandpass filter isolates the 57 kHz component from the FM discriminator output. A synchronous demodulator recovers the raw biphasic bitstream.
- **Synchronization (The "Flywheel"):** The receiver continuously checks the incoming bits against the Offset Words (A, B, C, D). Once it identifies the offset pattern, it "locks" to the start of the 104-bit groups.
- **Error Correction:** The decoder uses the 10-bit checksum to detect and, if possible, correct single-bit or burst errors caused by signal fading.

Data Interpretation:

- **PI (Program Identification):** Unique 16-bit code for station identity.
- **PS (Program Service):** 8-character display name.
- **AF (Alternative Frequencies):** List of frequencies for automated tuning.
- **RT (RadioText):** Up to 64 characters of scrolling text (e.g., Artist/Song).

Section 6 General Purpose Inputs/Outputs (GPIO)

Optional – Available Only on Products Ordered with this Option

Perceptual SoundMax™ Models Q24-1111 and Q24-1111HD audio processors offer a total of 12 General Purpose Input/Outputs (GPIO) on a DB25 connector (Figure 6.1) at the back of the chassis. Eight of these (1 to 8) are fixed as inputs while the last 2 (9 and 10) are configured as outputs. The inputs and outputs are optically isolated, and these are also floating such that the ground pins of different inputs (or outputs) are not tied together. For the inputs a voltage difference of +5.0 V between the +ive and ground pin reliably inputs a logical 1 input to the audio processor while a voltage below +0.5 V is reliably input as a logical 0. If the input pins are not connected to electrical circuitry the input to the processor is a logical 0. For the outputs a logical 1 outputs a voltage of +5 V between the +ive and ground pins of the corresponding output, and a near zero voltage guaranteed to be below 0.25V for logical 0. The role of inputs can be configured using the SoundMaxManager™ panel. The outputs are used to flag various run conditions as described in a table below.

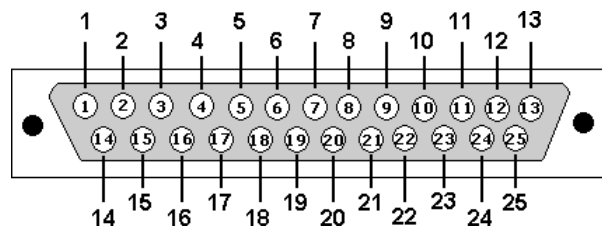


Figure 6.1: Connector Pin Configuration

PIN NO	Configuration	PIN NO	Configuration
1	INPUT 1: +ive	14	INPUT 1: Ground
2	INPUT 2: +ive	15	INPUT 2: Ground
3	INPUT 3: +ive	16	INPUT 3: Ground
4	INPUT 4: +ive	17	INPUT 4: Ground
5	INPUT 5: +ive	18	INPUT 5: Ground
6	INPUT 6: +ive	19	INPUT 6: Ground
7	INPUT 7: +ive	20	INPUT 7: Ground
8	INPUT 8: +ive	21	INPUT 8: Ground
9	OUTPUT 1: +ive	22	OUTPUT 1: Ground
10	OUTPUT 2: +ive	23	OUTPUT 2: Ground

GPIO Input Configuration

The 8 inputs can be configured using the SoundMaxManager™ Panel as shown in Figure 6.2 below.



Figure 6.2: GPIO Settings Panel

The Options will be interpreted as shown below:

For Bits b1 and b0:

Selected Options	Bits values (b1b0)	Interpretation
Option 1	00	No Effect
	01	Switch To Analog
	10	Switch To Digital
	11	No Effect
Option 2	00	No Effect/Stereo Audio
	01	Switch to Mono using Left channel
	10	Switch to Mono using Right channel
	11	Switch to Mono using Sum (Left+Right)
Option 3	00	Normal operation
	01	Test Tone 1KHz
	10	Pink Noise
	11	Bypass

For Bits b7,b6,b5,b4,b3 and b2:

Selected Options	Bits values (b7b6b5b4b3b2)	Interpretation
Option 1	000000	No Effect Normal Operation
	Xxxxxx	Switch to factory preset profile with the decimal number calculated by xxxxxx
Option 2	000000	No Effect Normal Operation
	Xxxxxx	Switch to user preset profile with the decimal number calculated by xxxxxx

GPIO Output Configuration

The 2 output bits are used to flag normal operation, error conditions, and integrity of GPIO input bits.

Bit pattern b0 b1	Interpretation
00	Normal Operation
01	Input bits parity is 0
10	Input bits parity is 1
11	Error/Stress Condition

Section 7 Hardware Specifications

MAIN Analog Audio Input:

- **Configuration:** Left and Right/Stereo
- **Maximum Input Level/Impedance:** +27dBu, > 10kΩ
- **Input Nominal Level:** Adjustable, -10.0 to +13 dBu
- **Input Headroom Level:** Adjustable, -10.0 to +14 dBu
- **A/D Convertor Resolution:** 24-bit
- **EMI Suppressed:** Yes
- **Connectors:** Balanced XLR Female

MAIN Analog Audio Output:

- **Configuration:** Left and Right / Stereo, Flat or Pre- Emphasized (50μs or 75μs software selectable)
- **Output level:** Adjustable, -6.0 to +24 dBu
- **Frequency Response:** ±0.3 dB (20 Hz – 20kHz),
- **SNR:** ≥ 90dB (20 Hz – 20kHz), unweighted
- **Crosstalk:** ≤ -90 dB (20 Hz to 20 kHz)
- **Distortion:** THD+Noise ≤ 0.01% (20 Hz to 20 kHz)
- **D/A Convertor Resolution:** 24-bit
- **EMI Suppressed:** Yes
- **Connectors:** Balanced XLR Male

MAIN Digital Audio Input (AES/ EBU):

- **Configuration:** Stereo
- **Sampling Rate:** 32, 44.1 or 48kHz (Auto Sense)
- **Word length:** 16 or 24 bit
- **Input Reference level :** -25 to 0 dBFS
- **EMI Suppressed:** Yes
- **Connector:** AES– XLR Female, 110Ω

MAIN Digital Audio Output (AES/EBU):

- **Configuration:** Stereo
- **Output level:** -20 to 0.00 dBFS
- **Frequency Response:** ± 0.3dB (5 Hz to 20 kHz)
- **Sampling Rate:** 32, 44.1, 48kHz (User selectable)
- **EMI Suppressed:** Yes
- **Connector:** AES – XLR Male
- **Pre- Emphasized:** 50μs or 75μs (software enabled)

Secondary Digital Audio Output HD1/DRM-Service A Output (AES/EBU):

(Model Q24-1111HD Only)

- **Configuration:** Stereo output with individual software level control.
- **Output level:** -20 to 0.00 dBFS
- **Frequency Response:** $\pm 0.3\text{dB}$ (5 Hz to 20 kHz) Sampling Rate: 32, 44.1, 48kHz (User selectable)
EMI Suppressed: Yes
- **Connector:** AES – XLR Male
- **Diversity Delay:** Up to 12 seconds delay in MAIN (software configurable with high precision). Accurate diversity delay ± 2 samples

Audio Over IP Input over Ethernet Port 1 (AOIP):

- **Configuration:** Stereo, AES 67 standard.
- **Connector:** RJ-45, EMI -suppressed.

Audio Over IP Output over Ethernet Port 1 (AOIP):

- **Configuration:** Stereo, AES 67 standard.
- **Connector:** RJ-45, EMI -suppressed.

Automatic failsafe input switching:

- Automatic failsafe switching of audio input Analog/Digital/composite/ Audio over IP in case of failure of primary input to secondary input (user selectable)

Composite Baseband Output:

(Model Q24-1111HD Only)

- **Configuration:** Two Outputs with Individual software level control, ITU-R BS 412 compliant
- **Maximum Output Level:** +12dBu
- **D/A Conversion:** 24 Bit
- **Signal to Noise Ratio:** $\geq 90\text{ dB}$, (20Hz – 15kHz)
- **THD :** $\leq 0.02\%$, (20Hz – 15kHz)
- **Stereo Separation :** $\geq 60\text{ dB}$ (20Hz – 15kHz)
- **Cross Talk:** $\leq -90\text{ dB}$, channel to channel
- **38kHz Suppression:** $\geq 90\text{ dB}$
- **Pilot Protection:** 65 dB relative to 9% injection, $\pm 500\text{ Hz}$
- **Pilot Stability:** 19 kHz $\pm 1.0\text{ Hz}$
- **Pilot Level:** Adjustable from 6% to 12%
- **57 kHz (RDS) Protection:** $> 60\text{ dB}$ relative to 4% injection, $\pm 2\text{ kHz}$
- **Connectors:** Two Coaxial BNC, EMI suppressed

Streaming Audio Output

- **Configuration:** 1 Internet stream out

- **Encoding format:** Enhanced AAC+ with ADTS encapsulation
- **Bit rates:** 32 kbps – 256 kbps
- **Player compatibility:** VLC, Windows media player, Real Player
- **Browser compatibility:** IE, Chrome, Safari, Mozilla
- **Protocols:** Unicast, Multicast, TCP, UDP
- **Distribution servers:** Icecast, Shoutcast

Management Control:

- Powerful SoundMaxManager™ on any networked PC for auto sense monitoring configuration and control, SNMP compliant.

USB Connection:

- **Connector:** 1 USB 2.0 (EMI Suppressed)

Ethernet & Remote PC Control:

- **Configuration:** Auto Presence Broadcast on Ethernet and RS-232
- **Connectors:** Both RJ-45 and RS-232 serial port, EMI Suppressed
- **Control:** Controller Software on any networked PC for auto sense monitoring and control.
- **Protocols:** TCP/IP , UDP , Multicast

GPIO Interface (Optional):

- **GPIO:** 8 inputs and 2 outputs opto-isolated and floating. User selectable configuration.
- **Control:** user- programable for factory preset, user preset, bypass, test tone, Analog input, digital input, stereo or mono model.
- **Connector:** DB-25 pin connector.

Power

- **Power Requirements:** 105-260 V AC Single Phase, 45-60 Hz.
- **Connector:** IEC, EMI Suppressed, Detachable 3-wire cord with earthing.
- **Safety Standards:** ETL Listed to UL Standards, CE marked.
- **Power factor:**

Power factor : A/C Input	Input line current	Full load P.F.
115 V	<5.5 Amps-rms	>0.95
230V	<3Amps-rms	>0.9

Operating Temperature range: 0° to 50° C.

Working Altitude: 3000 meters ASL Relative.

Humidity: 95% non-condensing.

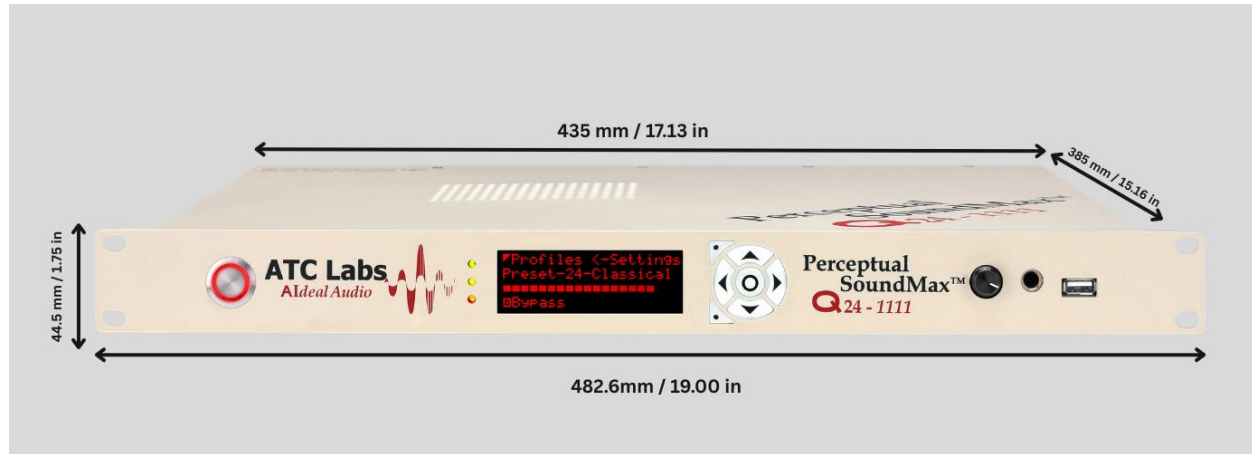
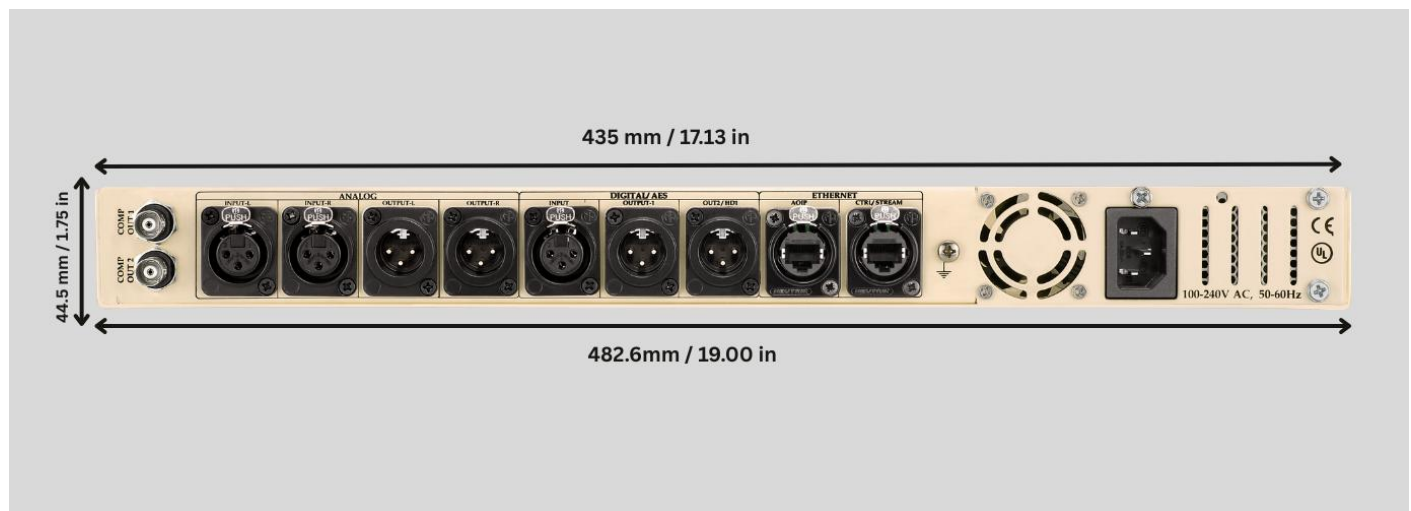
Weight: 5.5 Kg.

Dimensions:

- **Width:** 482.6 mm (19")
- **Height:** 44.5 mm (1 U/1.75")
- **Depth:** 385 mm (15.2")

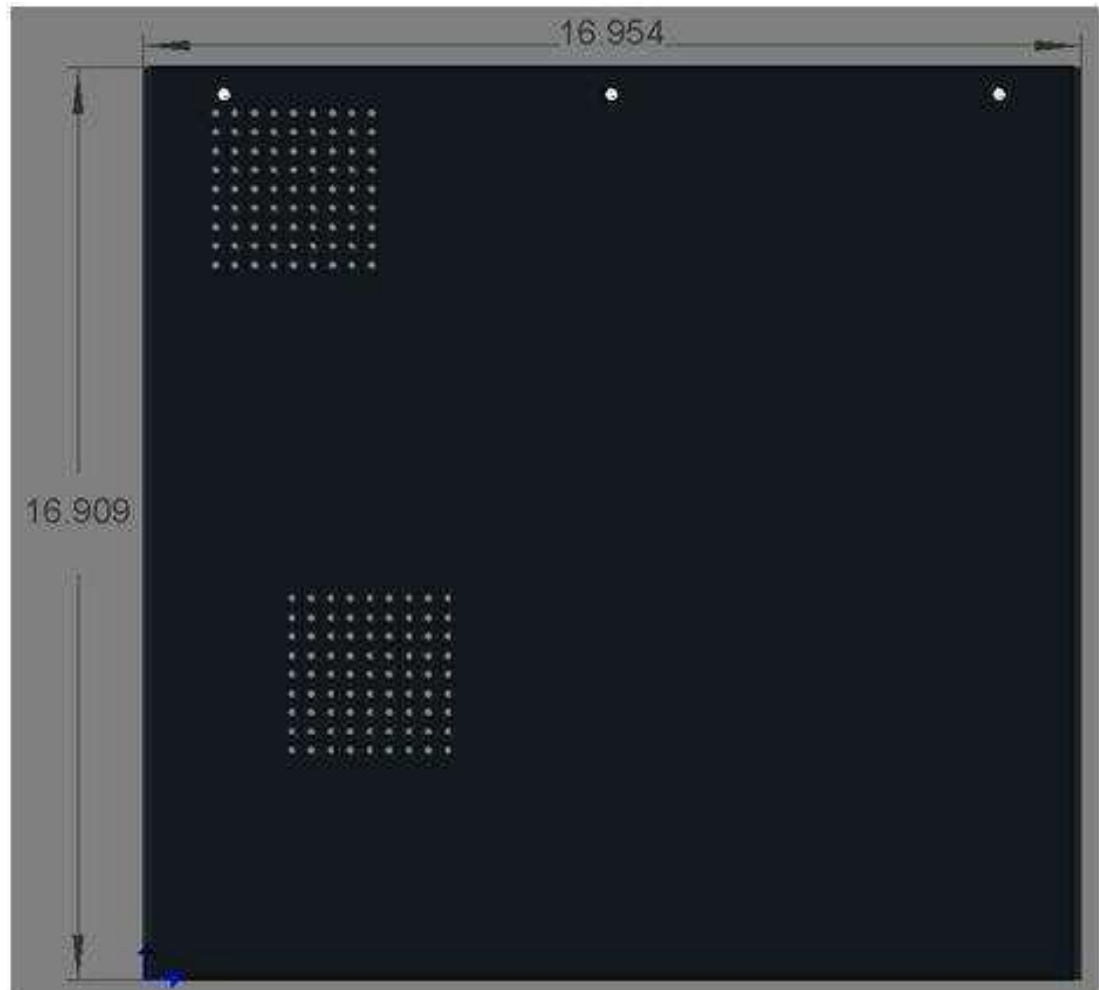
Section 8 Dimensioned Product View

Isometric View with Dimensions(mm)



Back Panel View with Dimensions (in meters & inches)

Top Cover View with Dimensions in meters



Section 9 Notices and Contact Information

For sales inquiries, technical support, or any other assistance, contact ATC Labs at the offices below.

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Online Resources

- [Perceptual SoundMax™ Product Page](#)
- [SoundMaxManager™ Download Page:](#)
<https://dwywczovty6aj.cloudfront.net/server/webphp/new-en.php>
- [Sales Enquiry](#)
- [Support Enquiry](#)

Perceptual SoundMax™ Model Q24-1111 and Model Q24-1111HD User Manual Version 8.1

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