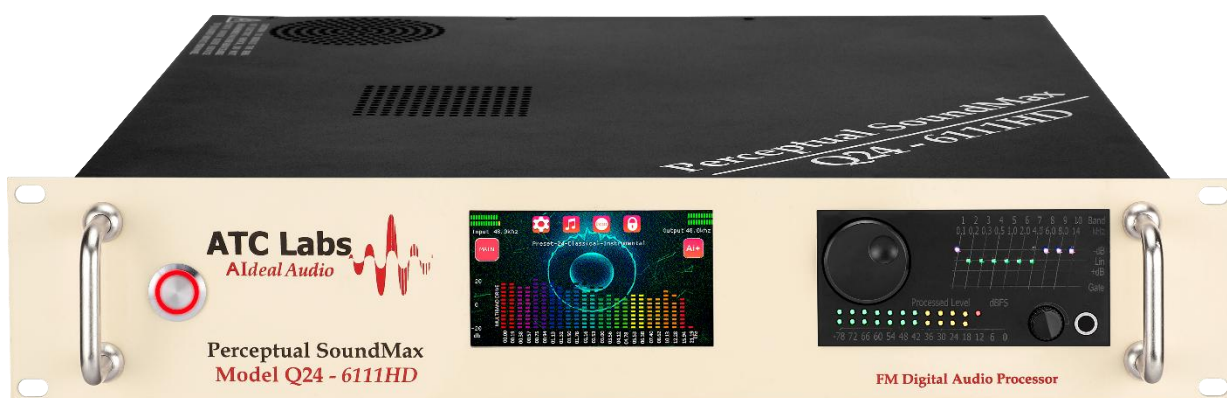




Perceptual SoundMax™

Models Q24-6111 & Q24-6111HD

24-band Audio Broadcast Processors for FM & Digital Radio

AI-Driven audio processing, Streaming and Immersive Audio

Operation Manual

Version 8.0, 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™ Q24-6111 series is a flagship *AI-driven audio processor* designed for FM, AM, and Digital/HD Radio broadcasts, as well as streaming simulcasts. Built on a sophisticated high-resolution 24-band dynamics processing engine, it leverages advanced psychoacoustic modelling, multiple deep-network ML algorithms, and accurate harmonic analysis to deliver a signature sound that is loud, punchy, and consistently balanced—all while minimizing listener fatigue.

Among the 2 processors in this series Perceptual SoundMax™ Model Q24-6111 includes all high-end features for FM broadcast audio processing including digital and balanced analog audio I/O, Composite and MPX FM signal generation, digital MPX output, in-box RDS encoder, Audio-Over-IP (AOIP), streaming simulcast and much more. Perceptual SoundMax™ Model Q24-6111HD is a further step up with high-end digital radio capabilities for 3 HD Radio™ channels – HD1, HD2, and HD3; or 3 Digital Radio Mondiale™ (DRM) channels. Streaming simulcast for all 3 digital channels is available in the Q24-6111HD model.

Three-in-One Integration:

The Perceptual SoundMax™ Q24-6111 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 3 Digital Radio channels (digital radio channels available in Model Q24-6111HD)
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. This series of audio processors also includes our **Aideal+™ genie** with 2 additional unique capabilities:
 - i. Profile Recommendation Tool – the genie listens to your radio station contents and recommends a profile from the available presets including guidance for further tuning.
 - ii. Auto Enhance Tool – choose a target sound such as enhanced vocals or enhanced instruments and the genie will continuously adjust the processing parameters to hit those targets consistently.

- **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.
- **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) 2 x FM RDS/Pilot Inputs (BNC) 1x Digital MPX (AES/XLR 192 kHz)
Digital Radio (3 additional processing paths)	3 x Digital Outputs (XLR/AES) – HD1, HD2, HD3 2 x Aux Digital Inputs (XLR/AES) – HD2 and HD3 Diversity Delay between Main and HD1 outputs Independent Processing Profile Selection for each output
Encoding	HE/LC-AAC V1/V2: 24-384kbps, MP3, Scalable HLS
Streaming, AOIP, and Networking	Configurable Audio-Over-IP (AOIP) input/output – multicast/unicast Streaming Simulcast: 1 channel (Model Q24-6111) or 3 channels (Model Q24-6111HD). Distribution server & CDN connectivity Dual Ethernet (RJ-45): Control/Stream and AoIP
Compliance	Internal Digital RDS Encoder & Nielsen™ Watermarking
Additional Tools	SpatialMax™ Spatial Audio, 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 Q24-6111 and Q24-6111HD, unless noted otherwise.

Section 2 **Installation And Quick Start**

Guide

2.1 Supplied Accessories

Package Contents

Your Perceptual SoundMax™ Model Q24-6111 or Model Q24-6111HD shipment includes the following items:

1. Perceptual SoundMax™ Audio Processor for Digital and FM (Model Q24-6111 or Model Q24-6111HD) hardware unit
2. Power cord
3. Rack-mounting kit with 10-32 x ½" screws and washers
4. Ethernet cable
5. SoundMaxManager™ software installation CD-ROM or USB pen
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 audio processor 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 Q24-6111 or Model Q24-6111HD requires 2U (3.5 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:** Do not use the equipment near source of moisture. Exposure to moisture increases the risk of damage and electrical hazards.
- **Heat:** Keep the equipment well separated 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 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 Q24-6111 and Q24-6111HD respectively.

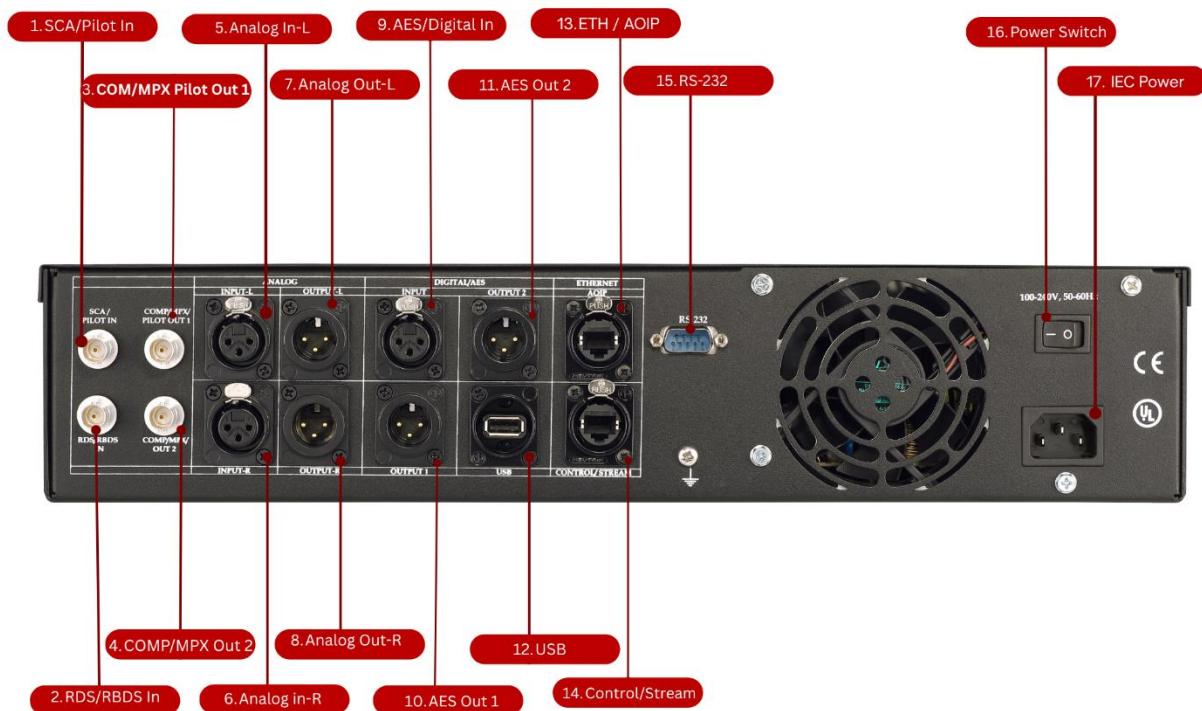


Figure 2.3.1: Model Q24-6111 REAR/BACK Panel

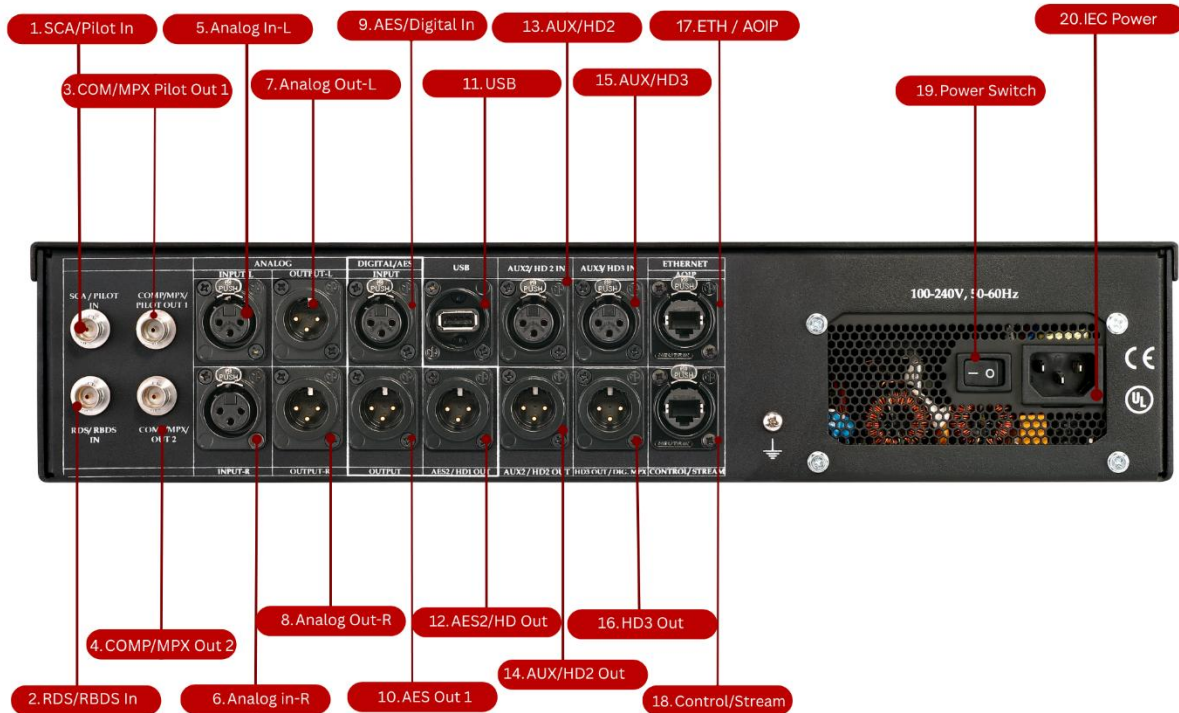


Figure 2.3.2: Model Q24-6111HD REAR/BACK Panel

All the I/O and power connectors 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 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 Rear/Back Panel Details and Input and Output Connections:

Composite / Baseband Connectors (BNC)

- **Baseband Input SCA/PILOT** – BNC female input for receiving an external SCA (Subsidiary Communications Authorization) subcarrier signal or an external 19kHz pilot signal (software configurable). This signal is used in composite generation (PILOT) or combined with the composite baseband output (SCA) for transmission
- **Baseband Input RDS/RBDS** – BNC female input for receiving an external RDS (Radio Data System) or RBDS (Radio Broadcast Data System) subcarrier signal. When an external RDS encoder is used instead of the internal RDS encoder, connect it here. The signal is injected into the composite output.
- **Composite Baseband Output 1 (COMP/MPX/PILOT)** – BNC female output providing the primary composite or multiplex (MPX) signal including Mid/Side stereo & pilot tone. Connect this output

to the FM exciter's composite input. Software configurable to output one of COMP, MPX, or PILOT. MPX output also includes RDS/RBDS/SCA subcarriers.

- **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) — Main Path

- **Balanced Analog Input (Right)** - XLR-3F (female) balanced analog audio input for the right channel. Accepts line-level audio signals with configurable nominal level and headroom.
- **Balanced Analog Input (Left)** – XLR-3F (female) balanced analog audio input for the left channel. Accepts line-level audio signals with configurable nominal level and headroom.
- **Balanced Analog Out (Right)** – XLR-3M (male) balanced analog audio output for the right channel. Output level is software configurable.
- **Balanced Analog Out (Left)** – XLR-3M (male) balanced analog audio output for the left channel. Output level is software configurable.

Digital AES Connectors (XLR) — Main Path

- **Digital/AES INPUT** – XLR-3F (female) AES3 (AES/EBU) digital audio input. Accepts a stereo AES3 digital audio stream. Supports multiple sample rates with in-line Sample Rate Converter (SRC).
- **Digital/AES OUTPUT (OUTPUT1 in Model Q24-6111)** – XLR-3M (male) AES3 (AES/EBU) digital audio output. Provides the processed stereo digital audio output. Output sample rate is software selectable.
- **Digital/AES OUTPUT2 (Model Q24-6111 only)** Software-configurable as a second Stereo Main or Digital Composite/MPX output at 192 kHz.

Digital AES Connectors (XLR) — HD/Digital Radio Paths (Model Q24-6111HD only)

The Q24-6111HD model adds two additional AES digital inputs and three additional AES digital outputs for independent HD/Digital Radio digital:

- **HD2/AUX2 IN** (XLR-3F female) - AES3 digital input for HD2/second digital channel, supports 44.1kHz or 48 kHz input (autosense)
- **HD3/AUX3 IN** (XLR-3F female) - AES3 digital input for HD3/third digital channel, supports 44.1kHz or 48 kHz input (autosense)
- **HD1/AES2 OUT** (XLR-3M male) — AES3 digital output carrying the processed HD1 channel audio. The input audio source for this output is the Main path analog or digital input. Diversity delay between Main output and HD1 is software configurable. Can also serve as a secondary main path AES2 output when HD Radio is not in use.
- **HD2 OUT** (XLR-3M male) — AES3 digital output carrying the independently processed HD2 channel audio. Sample rate follows the HD2/AUX2 input.
- **HD3/DIG MPX OUT** (XLR-3M male) — AES3 digital output carrying the independently processed HD3 channel audio. Sample rate follows the HD3/AUX3 input. Digital Composite/MPX output at 192 kHz.

Note: Each HD output (HD1, HD2, HD3) carries an independently processed audio stream with its own profile and tuning, allowing broadcasters to deliver different program material on each HD subchannel. HD1 output can be forced to be a mirror of the main digital output with the same processing profile, with a configurable diversity delay between the two, and independent level adjustment on HD1.

Network and Data Connectors

- **Ethernet 1 — AOIP (RJ-45):** Dedicated Ethernet port for AES67 Audio-over-IP (AOIP) 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.
- **Ethernet 2 — CONTROL / STREAM (RJ-45):** Ethernet port for remote control access via the SoundMaxManger™ PC software, firmware updates, configuration management. It also serves as Encoded Internet Streaming Output: Main stream (Model Q24-6111) or Main+HD2+HD3 streams (Model Q24-6111HD). Connect to the facility's control / management network using Cat5e or Cat6 cable.
- **USB Connector:** USB Type-A port for firmware updates, configuration backup and restore, and diagnostic log retrieval via USB flash drive.
- **RS-232 Serial Port (Model Q24-6111 only):** DB-9 male serial port for legacy serial communication and direct-connection remote control. Used for integration with automation systems or direct PC control without a network connection.

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 10kΩ; balanced and floating. Inputs can accommodate up to +24 dBu peak signal level without clipping (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 50Ω; balanced and floating. They can drive 600Ω or higher impedances, balanced or unbalanced. The peak output level is adjustable from –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. Input has in-line sample rate converters supporting 32 kHz to 192 kHz. Input sample rate is auto sensed. For best peak control, operate at 44.1 kHz or higher input sample rate. The output sample rate is software configurable to either 44.1 kHz or 48 kHz. Digital input is via one XLR-type female connector; digital outputs via two XLR-

type male connectors (4 male connectors in Model Q24-6111HD). Levels are front-panel configurable – section 3.

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 75Ω source impedance and can drive up to 8V peak-to-peak into 75Ω. 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 both Models Q24-6111 and Q24-6111HD

1. **Power Switch:** Use this switch to turn the unit ON or OFF. It remembers the last power state.
2. **TFT Touch Screen:** The equipment is equipped with a front panel TFT touch screen for the purpose of controlling and monitoring the unit.
3. **Rotary Selector:** This can be used instead of the touch screen control for controlling the unit. Generally, a user can also select the option displayed over the TFT screen by pressing the rotary selector switch once and move around different regions/options of the screen by rotating the rotary encoder in either direction.



Figure 2.4.1: Model Q24-6111/Q24-6111HD FRONT PANEL

4. **Headphone Volume Control:** To control the volume of headphone monitoring output
5. **Headphone Audio Output – ¼" jack:** Used for quick monitoring of the processed output of the audio encoder. A user can listen to the processed output through this audio jack.

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

Perceptual SoundMax™ Model Q24-6111 and Q24-6111HD hardware is supplied with SoundMaxManager™ PC software for remote configuration and control. This Network Management System (NMS) software is provided on a 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 Perceptual SoundMax™ audio processors connected to a network. The presence of one or more Perceptual SoundMax™ processors on the network is automatically sensed by the SoundMaxManager™ software running on any computer connected to that network.

Functionality

By using this software, users have complete flexibility to configure and monitor the audio processor. 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™ software setup on a Windows PC and follow the instructions
- After the SoundMaxManager™ installation, the user interface screen (UI) shown in Figure 2.5.1 below will appear.

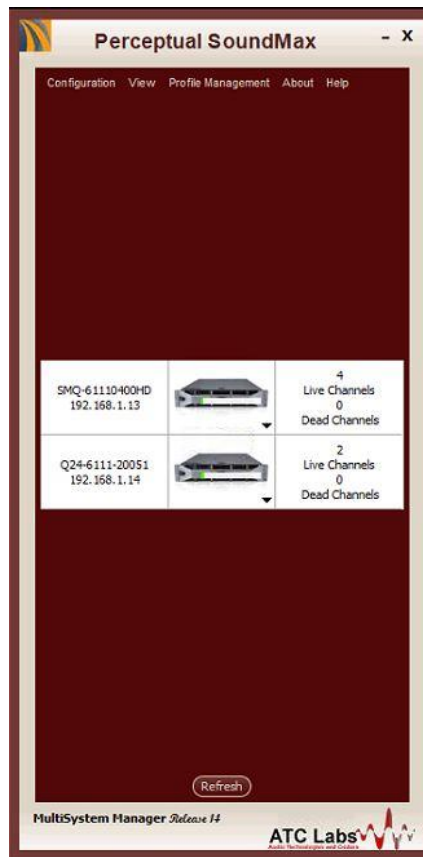


Figure 2.5.1: SoundMaxManager™ UI Screen

- SoundMaxManager™ will auto-detect the connected and powered audio processors as Live Stations. It will display their serial numbers and names including the model names. 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 audio processor.

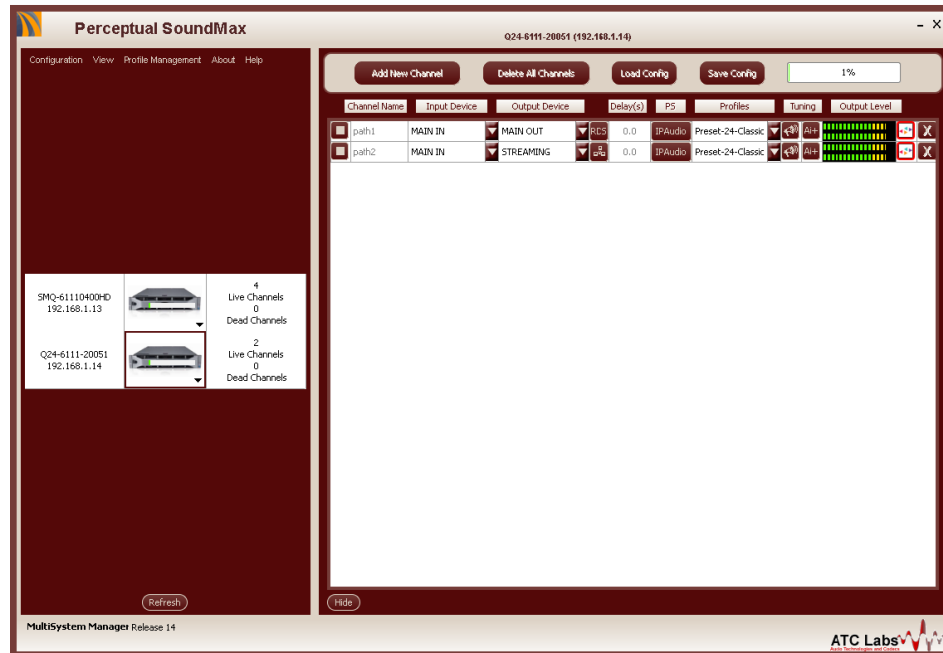


Figure 2.5.2: SoundMaxManager™ Detailed Right Panel View (Model Q24-6111)

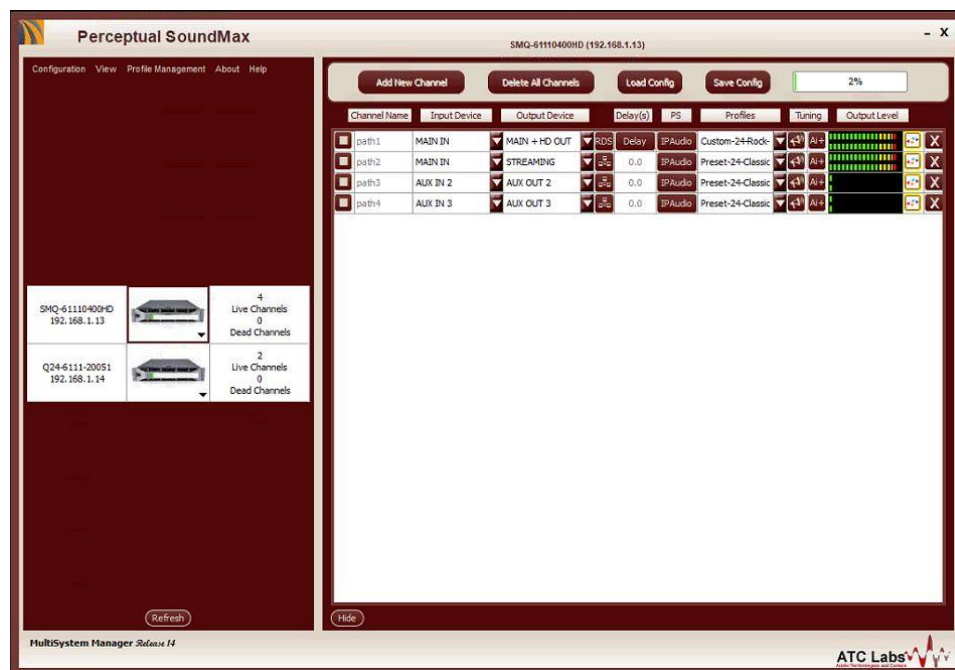


Figure 2.5.3: SoundMaxManager™ Detailed Right Panel View (Model Q24-6111HD)

This view provides the ability to configure the following:

- **Model Q24-6111:** For this model, the expanded right panel displays 2 paths (Figure 2.5.2). The first path is used to configure processing parameters, RDS encoding, AOIP, Aideal+, and Nielsen™ Watermarking settings for the MAIN PATH. The second path is used to configure streaming simulcast including the audio processing and encoding parameters for streaming output.

- **Model Q24-6111HD:** For this model, the expanded right panel displays 4 paths (Figure 2.5.3). The first path is used to configure processing parameters, RDS encoding, AOIP, Aideal+, Diversity Delay, and Nielsen™ Watermarking settings for the MAIN PATH and HD1 output. The second path is used to configure streaming simulcast of MAIN/HD1 source, including the audio processing and encoding parameters for this streaming output. Third path is used to configure processing parameters, Aideal+, Streaming Simulcast, and Nielsen™ Watermarking settings for the HD2 output. Fourth path is used to configure processing parameters, Aideal+, Streaming Simulcast, and Nielsen™ Watermarking settings for the HD3 output.

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

Note: If SoundMaxManager™ is controlling the audio processor, then the front-panel TFT interface will be temporarily disabled with a modal popup on the TFT screen (which may be used to reclaim control at the TFT).

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 Q24-6111 and Model Q24-6111HD 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 TFT 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: Stating Selection Screen on Front TFT Panel

Step 3 — Navigate the Selection Screen

Controls on the Selection Screen (and all other screens) can be operated in two ways:

- **Touchscreen** — Tap the on-screen buttons directly using the supplied stylus pen or a fingertip.
- **Rotary Encoder** — Rotate the encoder to move the highlight around the screen in a pre-defined pattern. Push in the encoder to activate the currently highlighted control.

Step 4 — Default I/O Configuration

The Model Q24-6111 and Q24-6111HD audio processor 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, 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.

First-time startup: The unit starts with the default I/O settings.

Subsequent startups: If the user has changed any settings and shut down the unit, it will power up with the last saved configuration.

Step 5 — Verify Audio

With audio connected to the analog input, confirm that:

- The input VU meters on the Main Screen show signal activity.
- The output VU meters show the processed signal.
- 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 — Detailed System Setup Instructions**.

Section 3 **Model Q24-6111 and Model Q24-6111HD TFT Screen Controls**

Perceptual SoundMax™ Model Q24-6111 and Q24-6111HD audio processors feature an intuitive and attractive touch-screen interface 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 touch-screen controls and their functionalities.

3.1 Home Screen

The Main Screen Display (Figure 2.6.1) highlights the following items:

- **Softkeys:** Setting, Tuning, Bypass, Unlock/Lock, AI+, Channel Profile Selection View
- Input and Output VU meter
- Output Spectrum in 24-Bands (center frequencies of the bands are also displayed)
- On-Air profile indicating the currently active audio processing profile. Repeatedly pressing on the MAIN button toward the top left cycles through all the channels (like MAIN, STREAM, HD1, HD2, HD3) and displays the currently active processing profile for the selected channel

Navigation Softkeys/Buttons:

- **Settings:**
Click Setting to access Analog I/O, Digital I/O, Composite, SCA/RBDS setup, and Info screens (section 3.3)
- **TUNING:**
Click the TUNING button to open the CONFIGURE screen (section 3.2) and view all available factory presets and custom profiles & apply any one of them to each of the active channels.
- **Bypass:**
 - Click Bypass to switch between processed and unprocessed output
 - 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”)
- **LOCK / UNLOCK:**
Click the LOCK/UNLOCK key to lock or unlock the screen.
Note: If you lock the screen, you will need a password to unlock it. The default password is *atclabs* which can be changed via the Setting->Info tab (section 3.6).
- **AI+ (Aideal+) Configuration**
Opens the Auto Tuning & AI Genie configuration panel. See section 3.4 for details.
- **Screen Navigation Notes**
 - Press the **HOME** soft key to return to the Main Screen from any other screen.

- Use the **rotary knob** to scroll through available controls or tap directly on any on-screen key for quick access.

3.2 Profile Selection/View

Using the CONFIGURE screen (as shown in Figure 3.2.1 below), the user can view all available processing profiles—both factory presets and custom profiles—and select the desired profile for each output path. Profiles can be independently assigned to the Main Path, the Streaming Path for both Models Q24-6111 and Q24-6111HD; and HD1, HD2, HD3, for Q24-6111HD, based on operational requirements and sound preferences.



Figure 3.2.1: Profile Configuration Screen

3.3 Input/Output Configuration

With the SETTINGS screen (as shown in Figure 3.3.1 below), a user can set up input and output levels according to studio appliances.

3.3.1 Input Configuration

The Model Q24-6111 and Q24-6111HD audio processors support stereo input through either the AES/EBU digital input or a balanced XLR type connection or AES67 AOIP source. Input selection is fully software configurable by pressing on the “Input” tab on the SETTINGS screen (Figure 3.3.1), allowing flexible integration into different system setups. In addition, the unit provides software-enabled optional de-emphasis at the input, ensuring compatibility with various legacy audio sources.

- **Word Length:** Supports 24-bit word length for high-resolution audio input.
- **Input Type:** The input source can be selected as ANALOG, DIGITAL, or AES67. For DIGITAL AES/EBU input, the incoming sample rate is auto sensed.
- **De-Emphasis:** If the incoming audio is pre-emphasized (J.17, 50 μ s, or 75 μ s), the corresponding de-emphasis must be applied (if not sure about input being pre-emphasized this selection is best left OFF).
- **Analog Input Controls:** Following 3 sliders are active when Input selection is Analog (Figure 3.3.1 (a)):
 - 1) **Analog Reference Level:**
 Matches this to analog input's *nominal level* Adjustable from -10 dBu to +12 dBu.
 - 2) **Analog Peak Level:**
 Provides adjustable headroom for protection against peaks above the reference level in the analog input. Adjustable from 0 dBu to 12 dBu.
Important Note: to avoid input from being clipped the above 2 sliders must be set so that input peak level never exceeds Analog Reference Level + Analog Peak Level; furthermore, for best processing performance it is helpful to not have too much (e.g., > 15 dB) or too little (e.g., < 3 dB headroom).
 - 3) **Prescale:** Optional additional input prescale for analog input. Adjustable from -10 dB to +20 dB.
- **Digital Input Controls:** Following 1 slider is active when Input selection is Digital (Figure 3.3.1 (b)):
 - 1) **Digital Reference Level:** Adjustable from -25 dBFS to 0 dBFS. Setting Digital Reference Level to -10 dBFS results in *flat input* without any pre-scaling. Setting it to -10 -x dBFS results in fixed x dB of input prescaling (amplification for +ive x and attenuation for -ive x).
- **AES67 Input Controls:** Refer to section 3.3.2

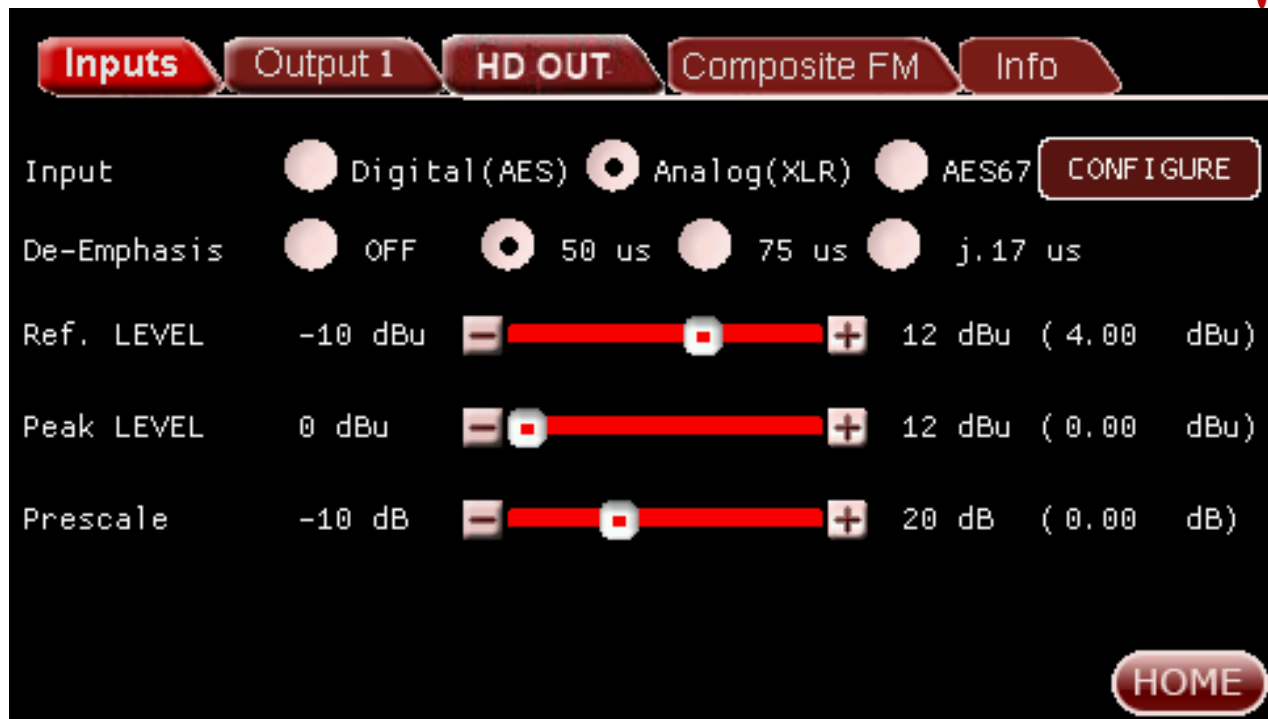


Figure 3.3.1(a): Input Configuration from the Input Tab (view when input selection is Analog)

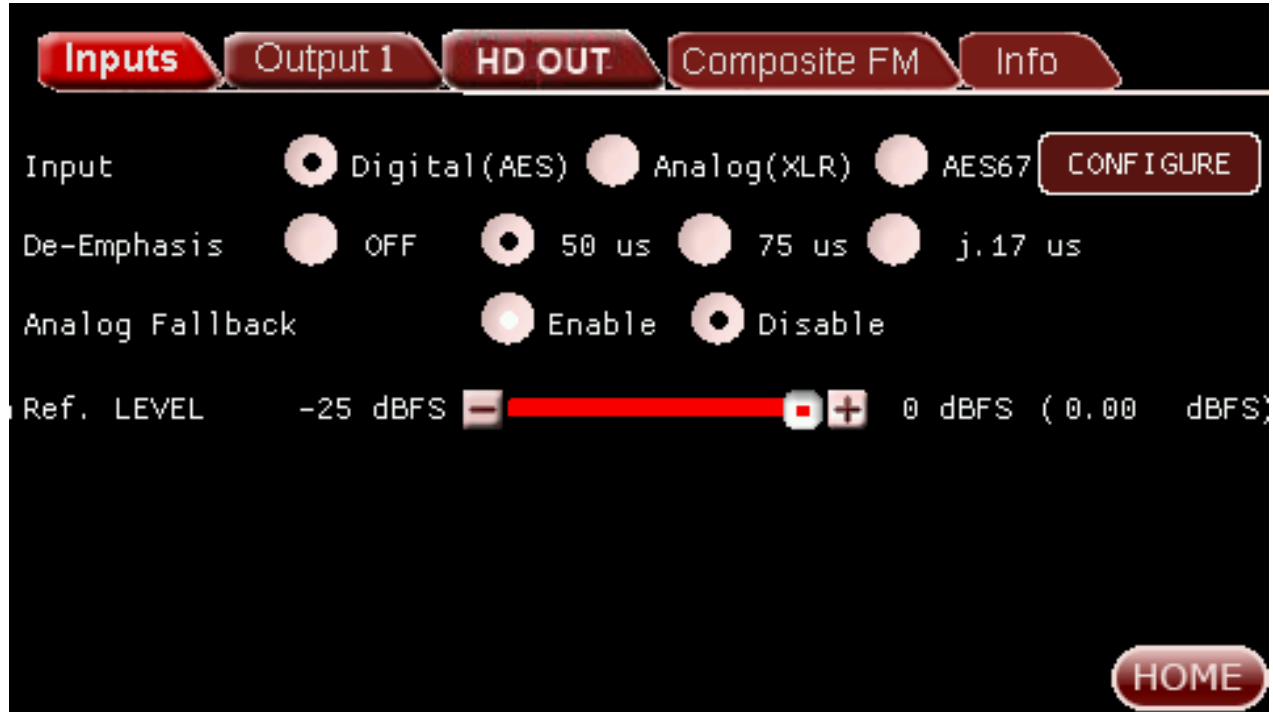


Figure 3.3.1(b): Input Configuration from the Input Tab (view when input selection is Digital)

3.3.2 Stream Audio Input (Configurable with SoundMaxManger™ Only)

The Model Q24-6111 and Q24-6111HD audio processors support the audio source for the MAIN PATH to be a streaming source in compressed audio format (HTTP/UDP AAC-ADTS/TS MP3-TS or HLS streams). To redirect the input source to a live stream the configuration must be done via SoundMaxManager™. The Input Device drop down selection on the SoundMaxManager™ expanded screen allows for the input to be toggled between MAIN IN hardware input and a NETWORK STREAM; if the second option is selected, a panel as shown in Figure 3.3.3 opens which allows for streaming input configuration. The display on the Inputs Tab of the front screen TFT then changes to show that source is pointed away to a stream (Figure 3.3.2); reselecting Digital/Analog/AES67 input options directs the source back to the corresponding inputs.

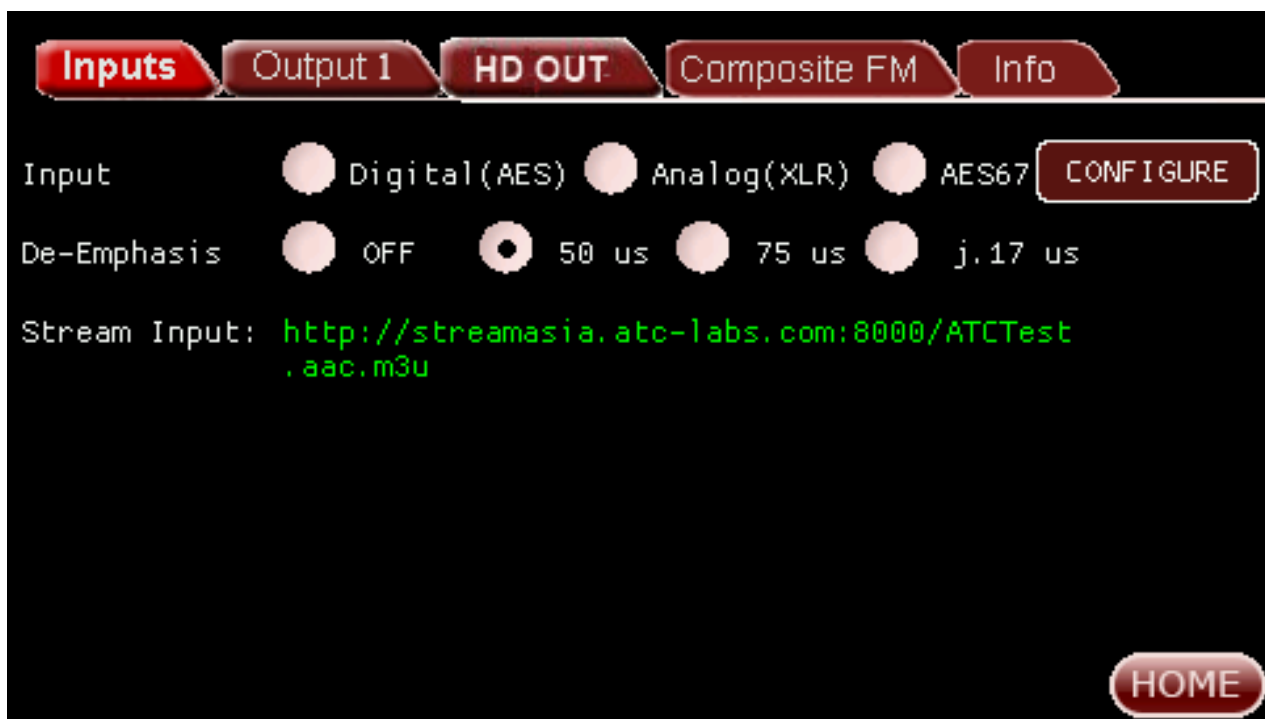


Figure 3.3.2: Stream Audio informational display on front TFT Inputs Tab

Figure 3.3.3: Stream Audio Input configured from “input Device” drop down arrow on SoundMaxManager™ Expanded Screen (Figures 2.5.2 & 2.5.3)

3.3.3 AES67 Input Configuration Screen

The AES67 Input Settings screen (as shown in Figure 3.3.4 below) enables setup of IP-based audio input streams compliant with AES67 standards. This allows the audio processor to receive high-quality uncompressed audio over an IP network for broadcast and studio applications.

Configurable Parameters:

- **PORT** — Specifies the UDP port number for the incoming AES67 audio stream (e.g., 7745). Enter the port value that matches the sending device's configuration.
- **IP ADDRESS** — Specifies the multicast or unicast IP address of the AES67 source stream (e.g., 192.168.1.54). This must correspond to the IP address being used by the transmitting device on the network.

- **SAMPLE RATE** — Selects the audio sampling rate of the incoming stream. Two options are available:
 - 44.1 kHz
 - 48 kHz
- **FORMAT** — Selects the audio bit depth (sample format) of the incoming stream:
 - L16 — 16-bit linear PCM
 - L24 — 24-bit linear PCM
- **CHANNELS** — Selects the channel configuration of the incoming audio stream:
 - STEREO — Two-channel (Left/Right) audio
 - MONO — Single-channel audio

AES67 INPUT SETTINGS

PORT: 7745

IP ADDRESS: 192.168.1.54

SAMPLE RATE: ☒ 44.1 ☐ 48 kHz

FORMAT: ☒ L16 ☐ L24

CHANNELS: ☒ STEREO ☐ MONO

CONFIGURE

SAVE **BACK**

Figure 3.3.4: AES67 Input Settings

Control Buttons:

- **CONFIGURE** — Applies the current PORT and IP ADDRESS settings and establishes the AES67 input connection. Press this button after entering or modifying the PORT and IP ADDRESS values.
- **SAVE** — Saves all current AES67 input settings to persistent memory so they are retained after a power cycle or system reboot.
- **BACK** — Returns to the previous screen without saving unsaved changes.

Usage Note: Ensure proper network configuration and PTP (Precision Time Protocol) synchronization before enabling AES67 input to avoid audio dropouts or stream synchronization issues. The PORT and IP ADDRESS values must match the source device's AES67 transmit settings.

3.3.4 Output Configuration

The Perceptual SoundMax™ Model Q24-6111 and Model Q24-6111HD audio processors provide stereo XLR balanced analog outputs as well as AES digital outputs – both the outputs are simultaneously active. The output configuration (as shown in Figures 3.3.5 and 3.3.6 below for Model Q24-6111 and Model Q24-6111HD respectively) is user-controlled. Available pre-emphasis selection allows the output to be either flat or with pre-emphasis at 50 μ s or 75 μ s, depending on FM broadcast and system requirements. The digital output can be configured as either a stereo main signal or as a Digital MPX output operating at 192 kHz, providing flexibility for different broadcast and audio processing applications.

- **TRIM:**
 - Digital output trim applied to both analog and digital outputs.
 - Adjustable Range from -20 dBFS to 0 dBFS
- **Analog Output Level:**
 - Users should calibrate and set output levels by playing a full-scale tone as input.
 - Adjustable Range from -6 dBu to +24 dBu (0 dBFS digital signal results in the set dBu analog level)
 - The analog output level is further attenuated if a TRIM, as set above, is applied.
- **Output Sample Rate:**
 - Software-selectable options: 32 kHz, 44.1 kHz, and 48 kHz
- **Pre-emphasis:**
 - Provides pre-emphasized analog and digital outputs.
 - Types available: 50 μ s and 75 μ s
 - Notes:
 - For FM broadcast pre-emphasis must be applied either in the audio processor or the transmitter (only one – please ensure that it is not activated in both). Most users will find the pre-emphasis implemented in this audio processor to be more accurate and effective than any similar option available in many transmitters.
 - The preemphasis selections should be based on the geographical region (e.g., 75 μ s in the US or 50 μ s in most parts of Asia including India).

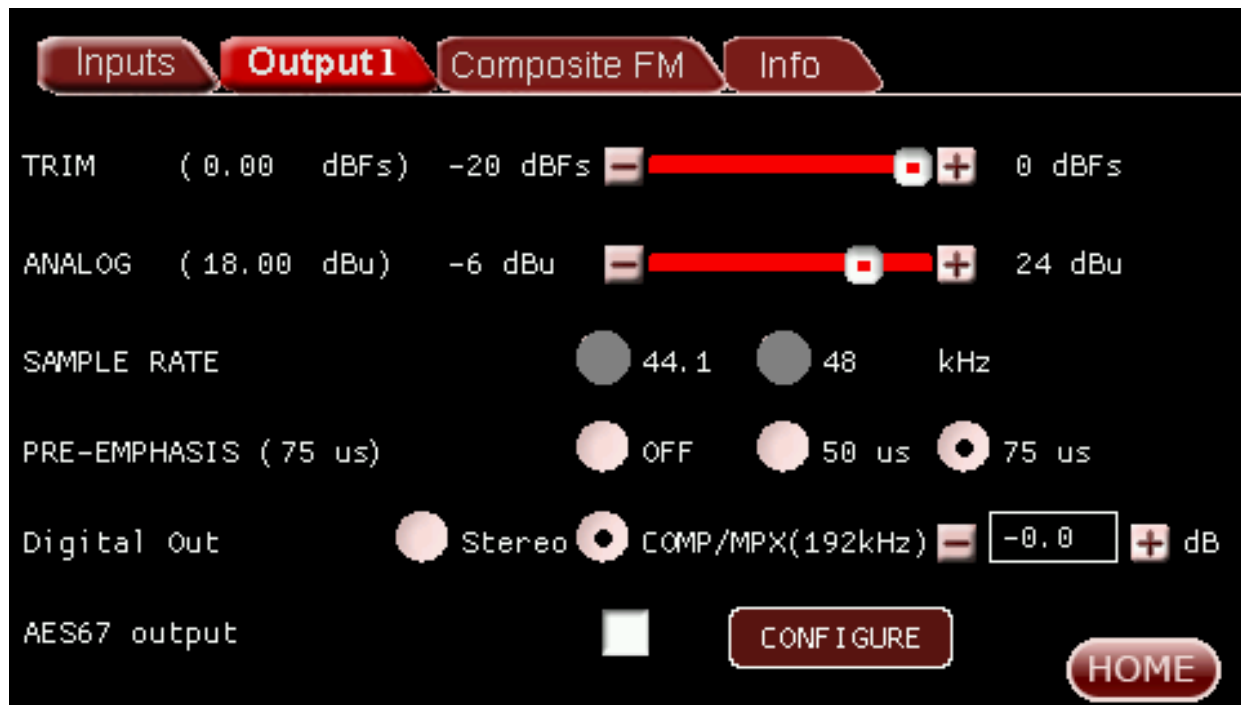


Figure 3.3.5: Output Configurations of Model Q24-6111

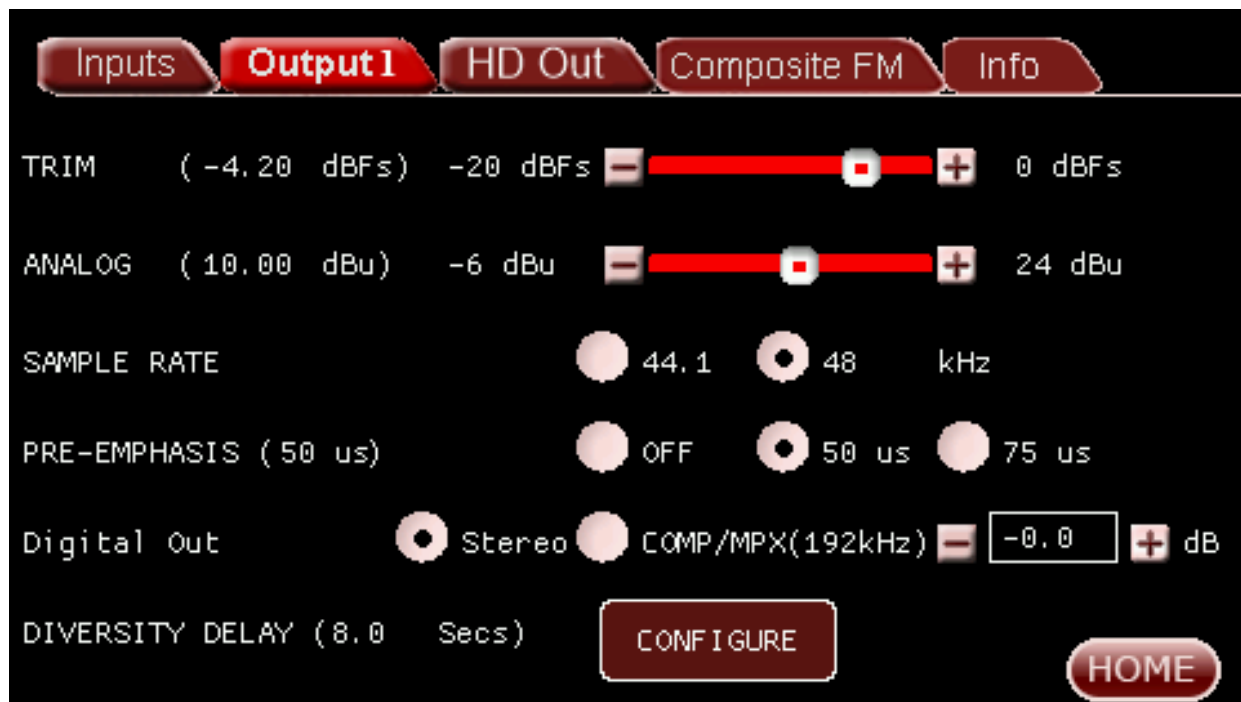


Figure 3.3.6: Output Configurations of Model Q24-6111HD

- **Digital Output Selection:**
 - Can be configured as Stereo Main or Digital Composite/MPX at 192 kHz
 - If digital output is configured as Digital MPX additional attenuation may be applied to this signal using the selection box on the right
- **AES67 Output Selection:** See section 3.3.5
Note: AES67 Output enable/disable & configuration controls are found on this tab in Model Q24-6111 only (Figure 3.3.5). For Model Q24-6111HD, these controls are found on the **HD Out** tab instead (see section 3.4.3).
- **Diversity Delay Configuration (Model Q24-6111HD Only):** See section 3.3.6

3.3.5 AES67 Output Configuration Screen

The AES67 Output Settings screen (as shown in Figure 3.3.7 below) enables the user to configure IP-based audio output streams compliant with AES67 standards. This allows the audio processor to transmit high-quality uncompressed audio over an IP network for broadcast, studio distribution, and remote monitoring applications.

Configurable Parameters:

- **PORT** — Specifies the destination UDP port number for the outgoing AES67 audio stream (e.g., 4428).
- **IP ADDRESS** — Specifies the multicast or unicast IP address to which the AES67 output stream will be transmitted (e.g., 198.168.1.65). The receiving device must be configured to listen on this same address.
- **SAMPLE RATE** — Selects the audio sampling rate of the outgoing stream. Two options are available:
 - **44.1 kHz**
 - **48 kHz**

Figure 3.3.7: AES67 Output Settings

- **FORMAT** — Selects the audio bit depth (sample format) of the outgoing stream:
 - **L16** — 16-bit linear PCM
 - **L24** — 24-bit linear PCM
- **CHANNELS** — Selects the channel configuration of the outgoing audio stream:
 - **STEREO** — Two-channel (Left/Right) audio
 - **MONO** — Single-channel audio
- **PACKET_SIZE** — Sets the packetization interval (in milliseconds) for the AES67 output stream (e.g., 0.333 ms). Use the – and + buttons to adjust. Smaller packet sizes reduce latency but increase network packet rate; larger values reduce network overhead but add latency. Common values are 0.333 ms (for low-latency applications) and 1.000 ms (for standard operation).

Control Buttons:

- **CONFIGURE** — Applies the current PORT and IP ADDRESS settings and initiates the AES67 output stream. Press this button after entering or modifying the PORT and IP ADDRESS values.
- **SAVE** — Saves all current AES67 output settings to persistent memory so they are retained after a power cycle or system reboot.
- **BACK** — Returns to the previous screen without saving unsaved changes.

Usage Note: Ensure that the receiving device or network infrastructure supports the configured sample rate, format, and packet size. PTP synchronization must be active on the network for reliable AES67 stream delivery. Verify that firewall and network switch settings allow multicast traffic on the configured IP address and port.

3.3.6 Diversity Delay (Model Q24-6111HD Only)

The Diversity Delay screen (as shown in Figure 3.3.8 below) provides precise control over the audio delay applied to the MAIN output signal (with respect to the HD1 output). This is essential for HD Radio (IBOC) installations where the analog FM signal must be time-aligned with the digital HD Radio signal in the receivers to ensure seamless blending during receiver transitions between analog and digital modes.

The screenshot shows the 'Diversity Delay' screen with a dark blue background. At the top, there is a title bar with 'Diversity Delay' and a 'BACK' button. Below the title bar, there are several input fields with labels and units, each with a minus button on the left and a plus button on the right. The fields are: 'Sec(2-9)' with a value of '8.00', 'ms(0-99)' with a value of '4', and 'us(0-999)' with a value of '188'. Below these are two read-only fields: 'Total Delay(secs):' with a value of '8.004188' and 'Total Delay(Samples):' with a value of '384201'. At the bottom, there is a field for 'Samples Adjusted(Min -10):' with a value of '0' and a '(Max 10):' label. At the very bottom, there are three buttons: 'APPLIED' (green), 'CALIBRATE NOW', and 'RESET'.

Figure 3.3.8: Diversity Delay Screen

Delay Configuration:

The total delay is specified using three fields for precise adjustment:

- **Sec (2–9)** — Sets the delay in whole seconds. Adjustable from 2 to 9 seconds.
 - **ms (0–99)** — Sets the delay in milliseconds. Adjustable from 0 to 99 ms.
 - **us (0–999)** — Sets the delay in microseconds. Adjustable from 0 to 999 μ s.
- Use the – and + buttons beside each field to increment or decrement the value.

Display Fields:

- **Total Delay (secs)** — Displays the combined total delay in seconds (e.g., 8.004188 seconds).
- **Total Delay (Samples)** — Displays the equivalent delay expressed in audio samples (e.g., 384201 samples at the system sample rate).
- **Samples Adjusted (Min –10 / Max 10)** — Allows fine-tuning of the delay by adjusting a small number of samples (range: –10 to +10). Use this for minor calibration corrections without modifying the primary delay settings.

Control Buttons:

- **APPLIED** — Indicates that the current delay settings have been applied and are active. Press to apply newly configured delay values.
- **CALIBRATE NOW** — Initiates an automatic delay calibration procedure. The system measures the actual time difference between the analog and digital signals and adjusts the delay accordingly for optimal blend alignment.
- **RESET** — Resets all delay values to the factory default settings.

Press **BACK** to return to the previous screen.

3.4 Aideal+ Audio AI Genie Controls

The **Aideal+ Audio Processing Genie** is an additional Aideal Audio™ AI-driven tool available in Perceptual SoundMax™ Models Q24-6111 and Q24-6111HD. It intelligently analyzes your on-air audio feed for content and provides automated recommendations and adaptive tuning to optimize processing performance. It can also be used to continuously auto tune the processor to emphasize a certain attribute such as *Vocal Clarity* or *Instrument Clarity*. Access this screen (Figure 3.4.1) by pressing the **AI+** soft key on the Main Screen.

The Aideal+ Audio Processing Genie screen provides two primary functions:

(I) RECOMMEND PROFILE

- The Aideal+ AI-based Audio Processing Genie listens to your radio station's on-air audio and recommends the best-matching processing profiles. The Station Engineer should audition and approve all On-Air Profile Configurations before applying. See section 3.4.1 for details.

(II) ADAPTIVE PROCESSING PARAMETER ADJUSTMENT (Auto Tune)

- The Aideal+ AI-based Audio Processing Genie continuously adapts profile parameters based on changing audio characteristics, ensuring optimal processing in real time. See section 3.4.2 for details.

Press **HOME** to return to the Main Screen.

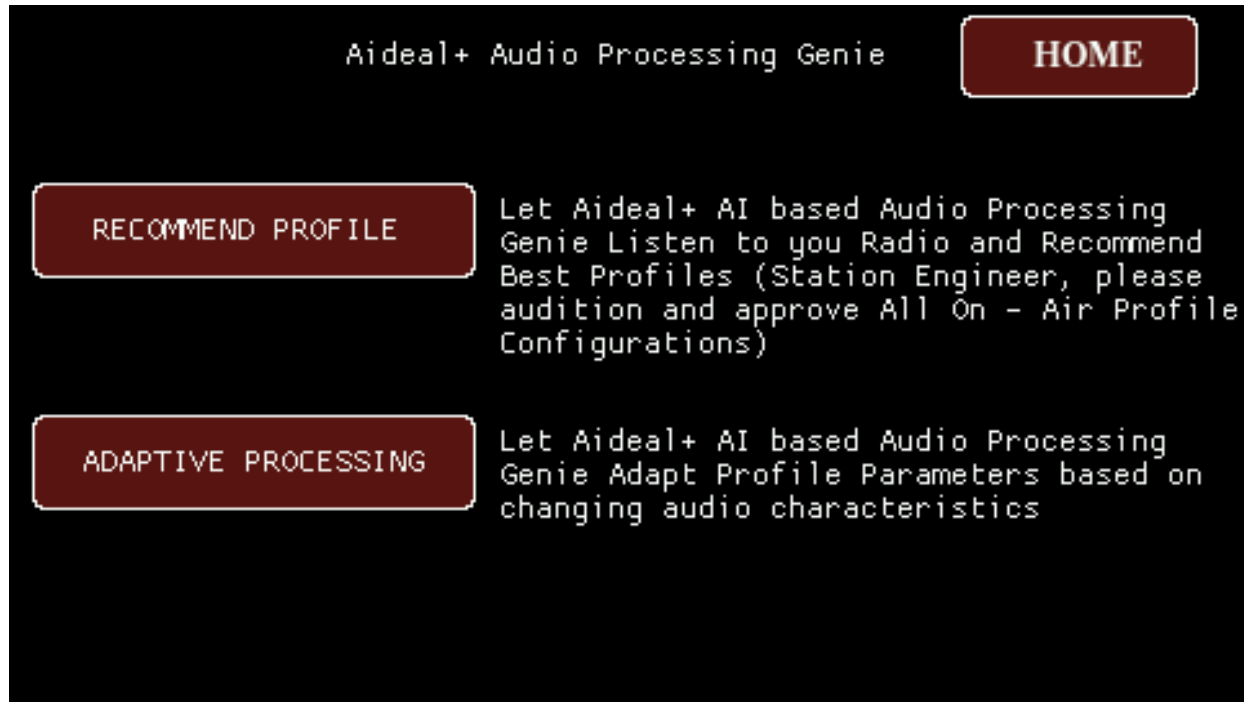


Figure 3.4.1: Aideal+ Audio Processing Genie Main Screen

3.4.1 Aideal+ Profile Recommendation

Pressing **RECOMMEND PROFILE** on the Aideal+ main screen opens the Profile Recommendation screen (Figure 3.4.2 below). This feature enables the AI engine to listen to the incoming audio signal, analyze its characteristics, and recommend the most suitable processing profiles from the available factory presets and custom profiles.

Operating Procedure:

1. Set Listening Duration

Use the – and + buttons to adjust the listening duration (in minutes). A longer listening window allows the AI to capture a broader range of program material for more accurate recommendations.

2. Start Listening

Press the **START LISTENING** button to begin the analysis. The screen will transition to a listening-in-progress state.

While listening is active, the screen displays:

- **LISTENING.** indicator (animated) confirming that audio analysis is in progress

- **Listening Started at** — timestamp showing when the analysis began
- **Listening will End at** — timestamp showing the scheduled end time based on the configured duration

The user may press **STOP LISTENING** at any time to end the analysis early and in that case previous analysis results will be shown. Press **BACK** to return to the Aideal+ main screen.



Figure 3.4.2: Aideal+ Profile Recommendation — Listening In Progress

3. Review Recommendations

Once the listening period is complete, the system displays its analysis results (Figure 3.4.3 below)

The results screen includes the following information:

- **Based on Listening on** — Shows the date and time range of the analyzed audio segment.
- **Best Profile Match** — Displays the top recommended processing profile (e.g., "Custom-24-Jazz-Classical"). Press the corresponding **APPLY** button to activate this profile on the current output path.
- **Next Best Match** — Displays the second-best recommended profile (e.g., "Custom-24-Country"). Press the corresponding **APPLY** button to apply this profile instead.
- **Further Tuning Guidance** — Provides specific parameter adjustment suggestions to further refine the selected profile. These recommendations may include:
 1. Loudness Control: Release Time range
 2. Loudness Control: Attack Time range
 3. Loudness Control: GainExp range
 4. Loudness Control: Noise Thr Hi range

- The Station Engineer should evaluate these recommendations and apply them using the Tuning interface (refer to Section 4) as needed to achieve the desired on-air sound.
- Press **BACK** to return to the Aideal+ main screen. Press **START LISTENING** again to run a new analysis session.

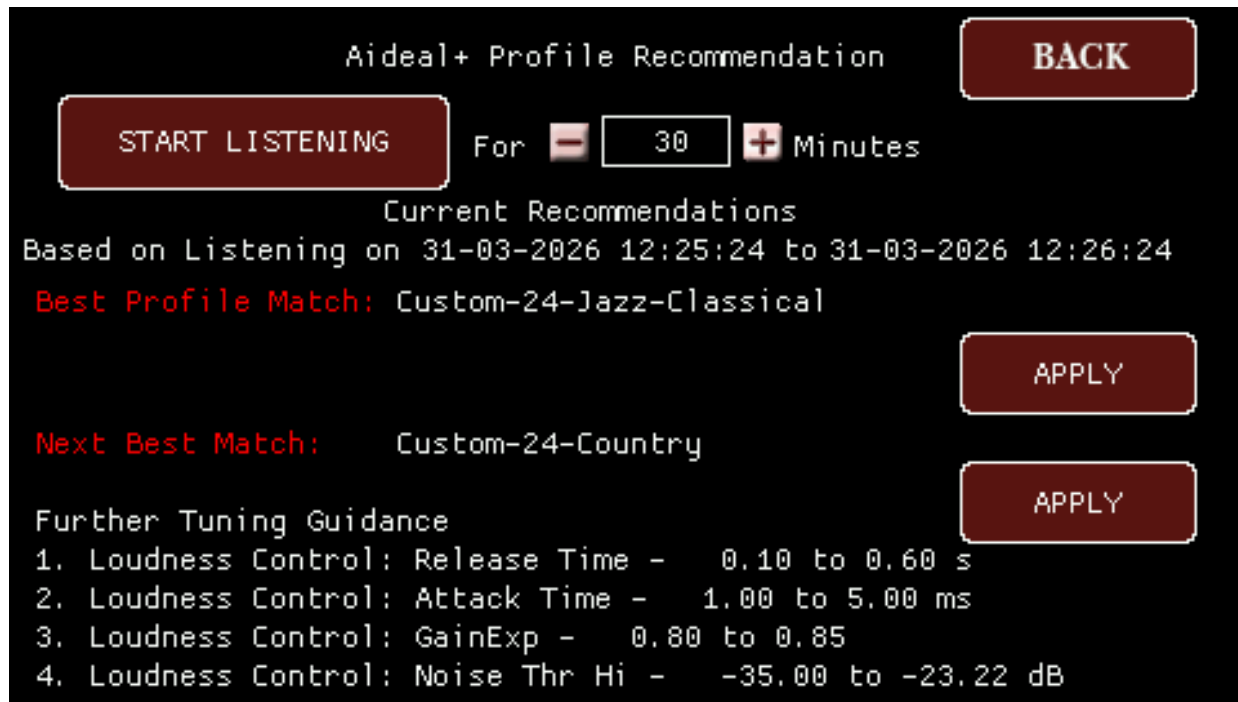


Figure 3.4.3: Aideal+ Profile Recommendation — Results Screen

3.4.2 Aideal+ Adaptive Processing (Auto Tune Genie)

Pressing **ADAPTIVE PROCESSING** on the Aideal+ main screen opens the Auto Tune Genie screen (Figure 3.4.4 below). This feature allows the AI engine to dynamically adjust key audio enhancement parameters based on the characteristics of the incoming audio, providing real-time intelligent processing optimization.

The Auto Tune Genie provides four adjustable attributes, each controlled via a slider, and the users can choose to emphasize one or more attributes for their broadcast sound:

- **Vocal Clarity** — Controls the degree of vocal enhancement and intelligibility improvement applied to speech content. Adjustable range: 0.00 to 1.00.
- **Music Clarity** — Controls the degree of musical detail enhancement and spectral clarity. Adjustable range: 0.00 to 1.00.
- **Stereo Enhancement** — Controls the degree of stereo image widening and spatial enhancement applied to the audio. Adjustable range: 0.00 to 1.00.
- **Noise Control** — Controls the aggressiveness of the AI-driven noise reduction system. Adjustable range: 0.00 to 1.00.

Applying and Removing Adjustments:

- Select **Apply** to enable the current Auto Tune Genie settings on the active processing path.
- Select **Unapply** to disable the Auto Tune Genie adjustments and revert to the standard profile settings.
- Use the checkboxes next to each parameter to individually enable or disable specific enhancements.



Figure 3.4.4: Aideal+ Auto Tune Genie Screen

Saving Settings:

- **SAVE TO PROFILE** — Saves the current Auto Tune Genie parameter values into the active processing profile for persistent use.
- **SAVE TO CHANNEL** — Saves the current settings to the selected output channel configuration, allowing different channels to have independent adaptive processing configurations.

Press **BACK** to return to the Aideal+ main screen.

3.4.3 HD Out Tab (Model Q24-6111HD Only)

This screen consists of two primary sections:

HD Channel Output Controls: Figure 3.4.5 shows how all three HD path Trim can be controlled through sliders ranging from -20dBfs to 0 dBfs. The sample rates displayed next to each of HD1/HD2/HD3 paths are for information only. For HD1, the sample rate matches that of the MAIN output setting (section 3.3.4); for HD2/HD3 it is sensed from the corresponding digital input.

AES67 Output: Enable/disable & configure; details are in earlier section 3.3.5 AES67 Output Configuration Screen.

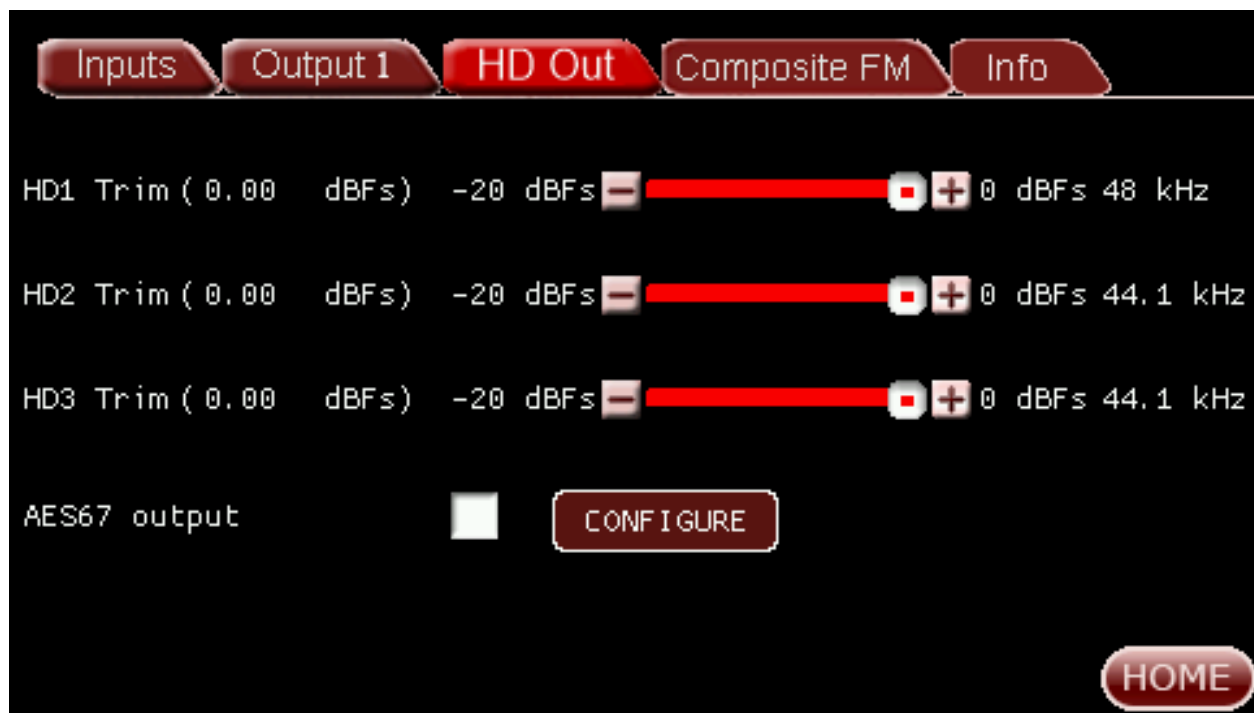


Figure 3.4.5: HD Out Controls

3.5 Composite FM Controls

The Perceptual SoundMax™ Models Q24-6111 and Q24-6111HD are equipped with two composite baseband outputs and two baseband inputs, each featuring independent user-controlled output levels and dedicated connectors. This design provides flexibility for system integration, allowing precise adjustment of each input or output to meet specific installation or operational requirements (Figure 3.5.1 below).

- **OUT1:** It's the coaxial output and software selectable as COMP, MPX, or PILOT tone output of 19 kHz.
- **OUT2:** It's the coaxial output and software configurable as COMP or MPX or RDS output on this screen.

- **OUT1 LEVEL:** OUT1 has a range from -12 dBu to +12 dBu, configurable using the corresponding slider shown in Figure 3.5.1.
- **OUT2 LEVEL:** OUT2 has a range from -12 dBu to +12 dBu, configurable using the corresponding slider shown in Figure 3.5.1.
- **PRE-EMPHASIS:** If pre-emphasis is already applied on the main path, this option becomes disabled, and the same pre-emphasis value is automatically applied to the composite path. If pre-emphasis is *not* applied on the main path, then this option allows the user to manually configure the type of pre-emphasis applied on the composite path only
- **PILOT LEVEL:** Adjustable between 6%–12%.
- **RDS INT/EXT SELECTION:** Use these controls to set RDS (output as part of the MPX outputs) source to be INT (internally encoded) or EXT (externally provided on IN2)
- **IN1:** Input 1 can be selectable as OFF, SCA, or PILOT.
- **IN2:** Input 2 is dedicated to RDS. The RDS option can be internal RDS encoder or external as configured above. IF internal RDS encoder is chosen then this input is ignored.

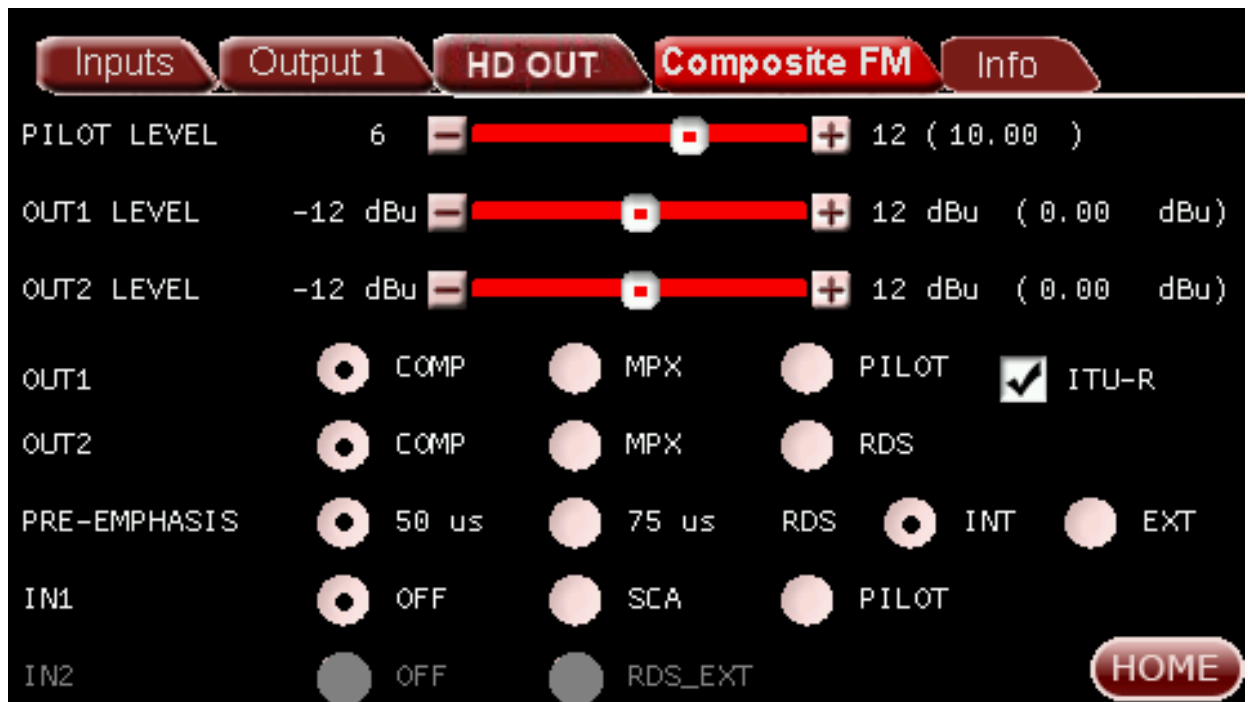


Figure 3.5.1: Composite FM Configuration Tab

3.6 Information & Diagnostics Controls

The Info Screen (Figure 3.6.1 below) provides a consolidated summary of all important system settings in one convenient view. It displays key operational parameters and offers access to additional configuration and diagnostic features.



Figure 3.6.1: Audio Settings Info Screen

This centralized information screen ensures quick access to vital configuration details and essential maintenance controls.

The following advanced functions are available through the corresponding soft keys:

- **UPGRADE** – Allows the user to update the system software or firmware.
- **RUN DIAGNOSTICS** – Provides access to built-in diagnostic tools for system health monitoring, troubleshooting, and performance checks.
- **SET PASSWORD** – The screen in Figure 3.6.2 appears when the **SET PASSWORD** soft key is pressed.

Password Requirements:

- A valid password must contain **more than 5 characters**.
- After entering the password, press the **SAVE** soft key to store the new password.

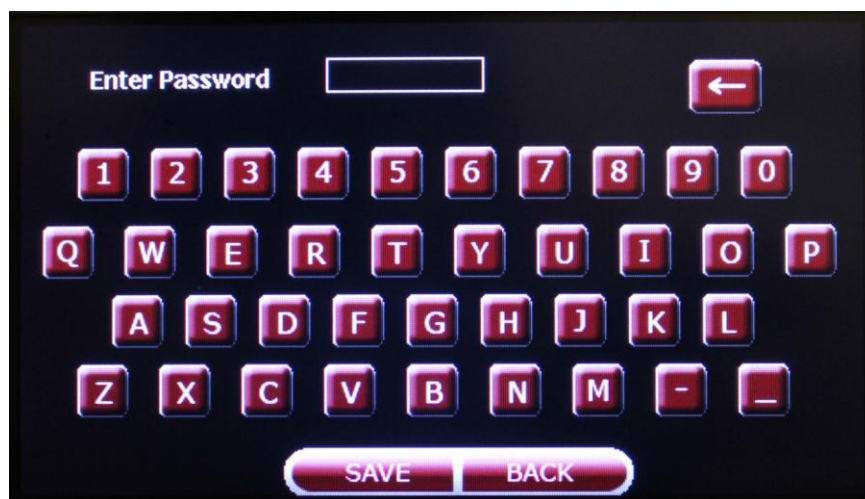


Figure 3.6.2: Set Password Screen

RUN DIAGNOSTICS

The Run Diagnostics feature helps determine whether the unit is malfunctioning.

When the RUN DIAGNOSTICS soft key is pressed, the diagnostic screen (Figure 3.6.3 below) appears, allowing the user to test each of the three processing paths—Analog, Digital, and Composite FM.

Diagnostic Procedure:

1. Select Path

Choose one of the following paths to diagnose:

- Digital
- Analog
- Composite FM

2. Test Signal Options

Diagnostics can be performed using built-in test tones:

- L+R
- L–R
- Left Only
- Right Only

These tones are available at all Digital, Analog, and Composite outputs.

3. Start Test Playback

Press the Playback key and select one of the following signal types:

- Tone — fundamental frequency adjustable from 1 kHz to 24 kHz
- Sweep — user can configure Start and Stop frequencies
- Pink Noise

4. Loopback Mode

The unit can also be tested in loopback configuration if required.

5. Return to Normal Operation

Press the GO LIVE key to return to standard Processing Mode.



Figure 3.6.3: Diagnostic Screen

3.7 Processing Parameters Touch-Up

The Tuning screen (Figure 3.7.1) allows the user to enable or disable individual processing blocks in the active profile and access their detailed parameter screens. It is accessed by pressing the TUNE soft key on **Profile Selection/View** screen in Figure 3.2.1.

The screen header shows the active path and current on-air profile (e.g., "Main: OnAir: Preset-24-Classical-Instrumental").

Controls

This section presents a brief overview of the Tuning Screen elements. Detailed control and parameter descriptions are provided in Section 5.3 – Processing Parameter Touch-Up via TFT Screen. Each processing block has two controls:

- **Checkbox** — Enables (✓) or disables the block. Disabling bypasses the block without erasing its stored parameters.
- **VIEW** — Opens the detailed parameter screen for that block (refer to Section 5 for full parameter descriptions).

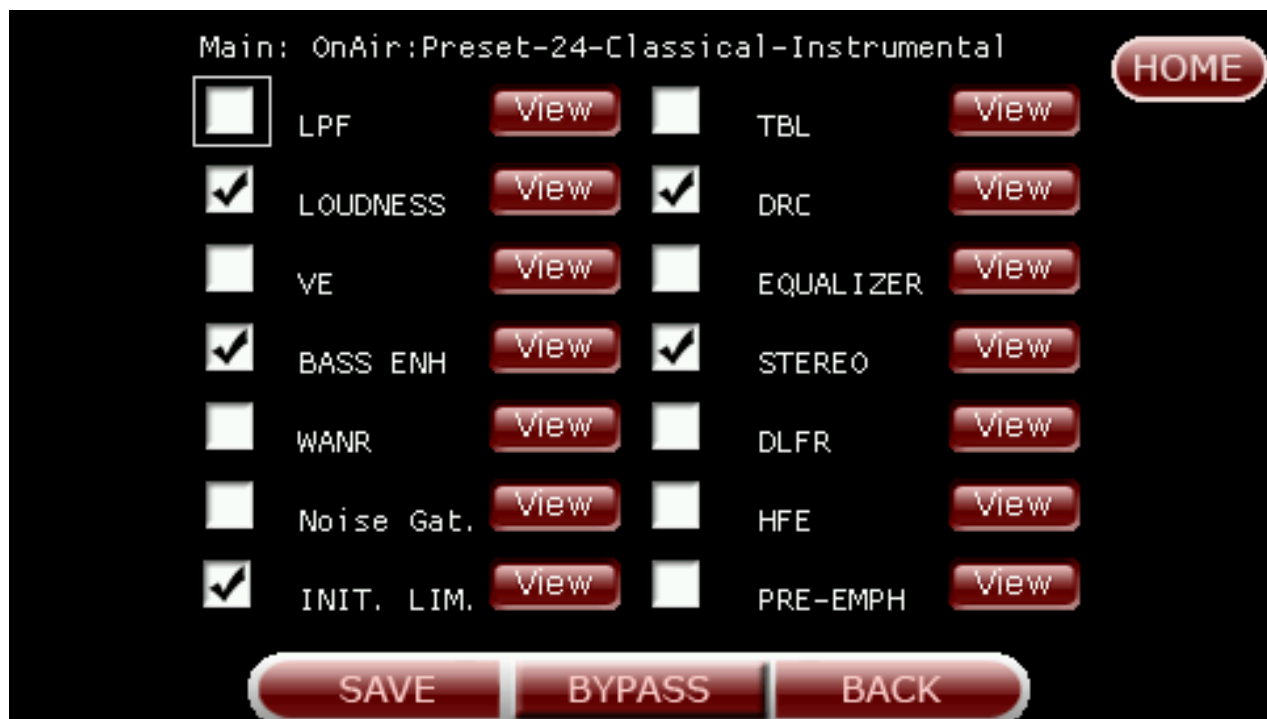


Figure 3.7.1: Tuning Screen.

Available Processing Blocks

- **LPF:** Low-pass filter

- **LOUDNESS:** Master loudness control
- **VE:** Voice enhancement
- **BASS ENH:** Bass enhancement
- **WANR:** Wideband auto noise reduction
- **Noise Gat.:** Noise gate
- **INIT. LIM.:** Initial limiter
- **TBL:** Two-band limiter
- **DRC:** Dynamic range control
- **EQUALIZER:** Graphic equalizer
- **STEREO:** Stereo processing
- **DLFR:** Dynamic listening fatigue reduction
- **HFE:** High-frequency enhancement
- **PRE-EMPH:** Pre-emphasis filter

Action Buttons

- **SAVE:** Saves the current enable/disable settings to the active profile.
- **BYPASS:** Bypasses the entire processing chain for A/B comparison with unprocessed audio.
- **BACK:** Returns to the previous screen without saving.
- **HOME:** Returns directly to the Main Screen.

Section 4 SoundMaxManager™ Controls

SoundMaxManager™ controls can be adjusted from the detailed view in Figure 4.1 below. The description in this section is from the point of view of higher capability Model Q24-6111HD (which has 4 paths as shown in this Figure). Model Q24-6111 offers 2 of these paths (i.e., no HD1/HD2/HD3); and all the description in this section applies to that model as well, except for HD/Digital Radio specific features such as Diversity Delay, and of course information related to paths 3 and 4.

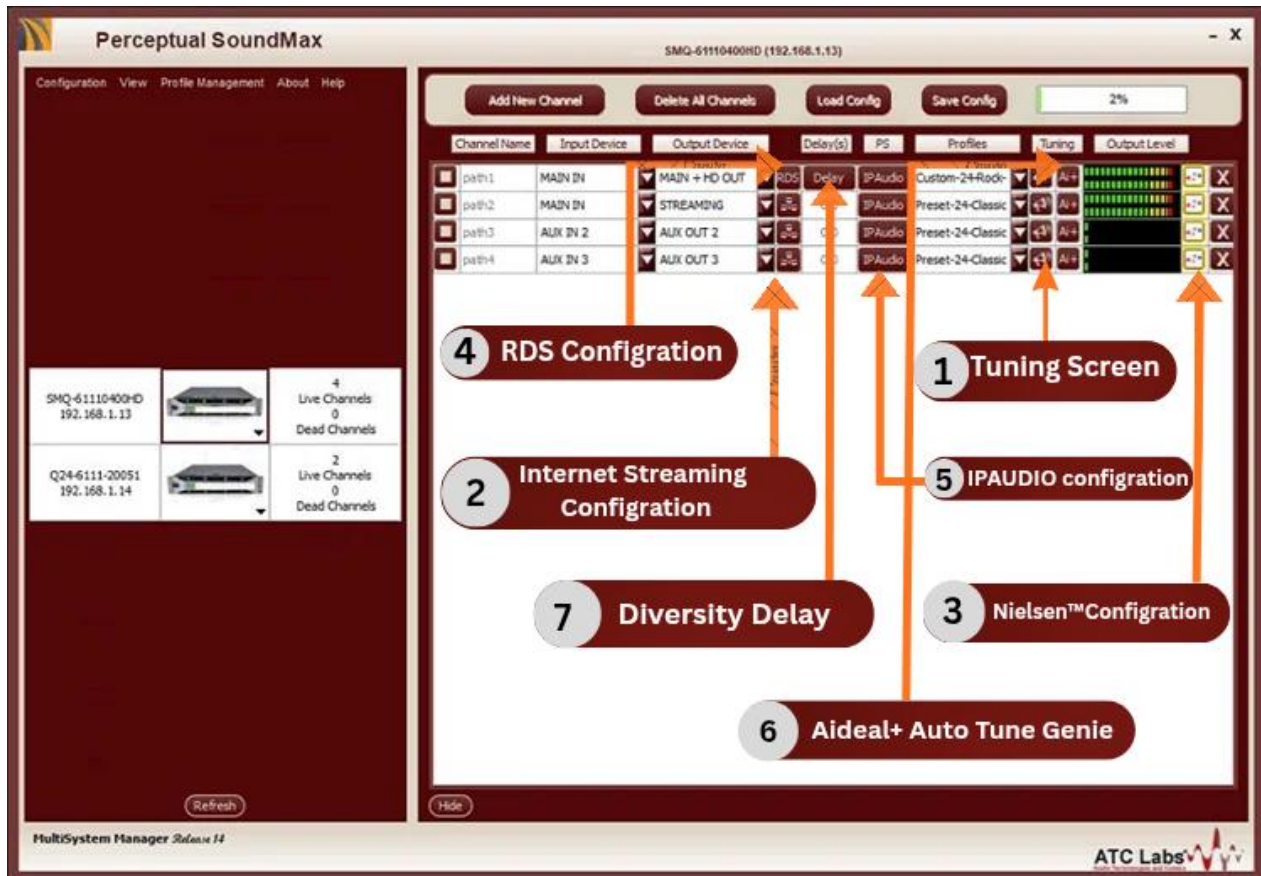


Figure 4.1: SoundMaxManager™ Detailed View (Model Q24-6111HD) – adjustments & controls

4.1 Advanced Tuning Screen for Creating & Fine-Tuning Processing Profiles

The Advanced Tuning screen (Figure 4.1.1) is accessed by clicking on the SoundMaxManager™ detailed view Tuning Screen button ❶. As described in Section 5.2, the user is then given an option to choose either “Basic Tuning” controls or “Advanced Tuning” controls; choosing Advanced Tuning Control opens the screen in Figure 4.1.1. Detailed Description of the effect of each of these controls are described in Section 5.2. Here we briefly describe the aspects of creating custom profiles. You may start with one of the available factory preset profiles, selectable from the **Profiles** drop-down menu on the top left of this

screen, and then create a custom signature sound for your station by adjusting the parameters on this screen. After tune ups profiles are saved as a Custom profile – an option also available in the **Profiles** drop-down menu.

Figure 4.1.1 below shows the tuning screen where various parameters can be controlled/set.

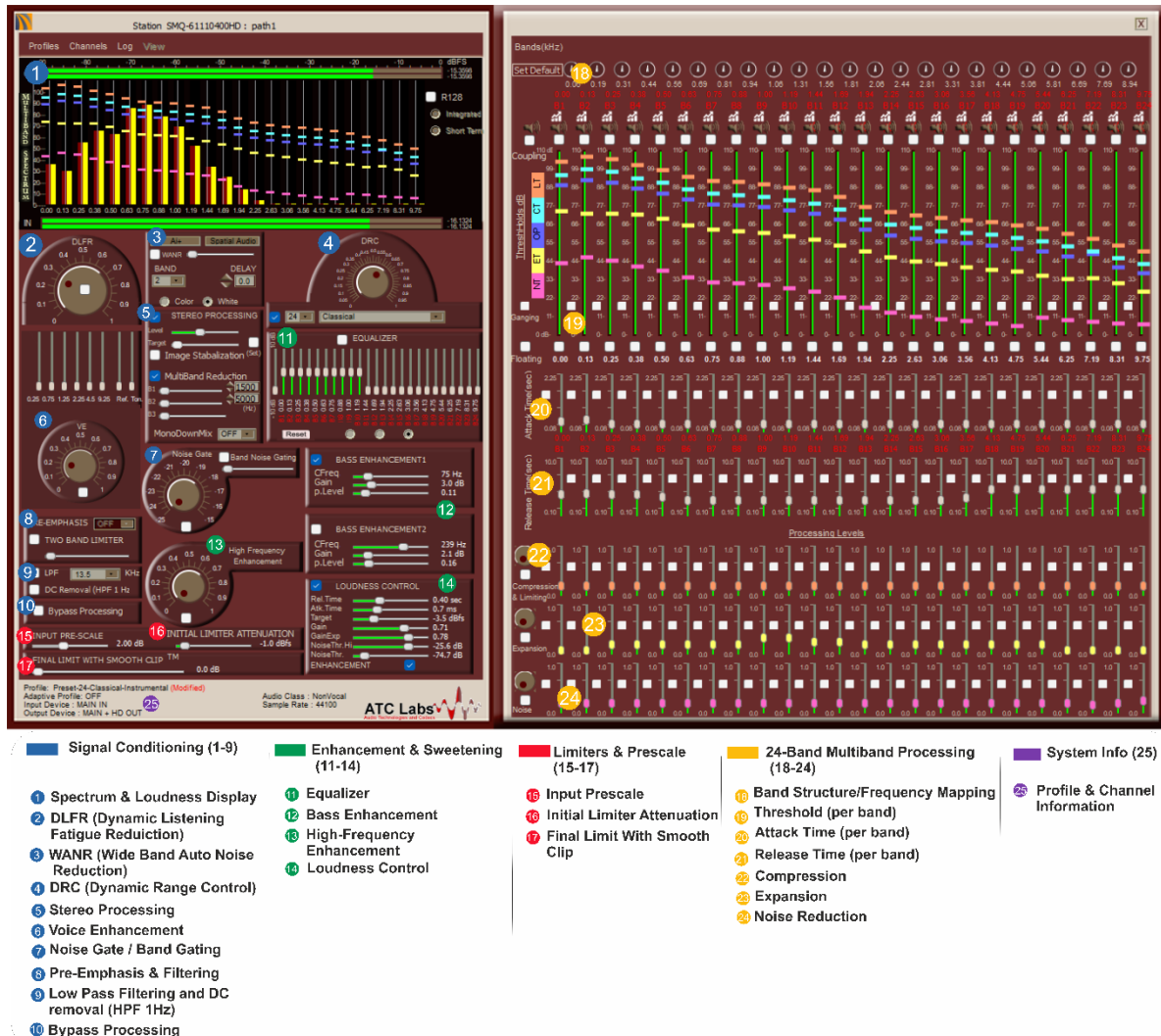


Figure 4.1.1: Tuning screen to control/set various parameters

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:** This preset is 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. The user may also want to invoke and configure a processing block which was previously unused. While limited tuning is possible using front panel TFT as described in Section 3 and 5, SoundMaxManager™ provides a much higher level of control in terms of tuning the processing parameters – this is described in detail in Section 5. Once a user is satisfied with the sound the resulting profile may then be saved as a custom profile and is then added to the list of available profiles for any of the processing paths. Also, custom created profile names are added

with a “Custom-” prefix in addition to the user specified names at the time of Profile save operation. The augmented list of processing profiles with all Custom-** profiles become available either from the front TFT (Section 3.2) or from the SoundMaxManager™.

4.2 Internet Streaming Configuration

Customized Internet Radio integrated in Perceptual SoundMax™ gives you a totally new experience in terms of versatility, quality, and ease of configuration. Internet streaming can be configured by pressing Button labels ② on the SoundMaxManager™ detailed view panel in Figure 4.1. This Internet Stream Configuration button is available for *path 2* (Models Q24-6111 and Q24-6111HD) and *path 3* and *path 4* (Model Q24-6111HD only); in *path 2* it is used to configure the streaming for the MAIN PATH audio, while in *path 3* and *path 4* it is used to configure streaming for H2 and HD3 audio sources respectively. The panel shown in Figure 4.2.1. opens when the button labeled ② is pressed on any of these paths. Multiple simultaneous streaming options are available:

1. AAC Streaming with and/or without HLS chunking via push/post to a CDN server and/or via an in-built HTTP distribution server. Optional TS encapsulation (Section 4.2.2)
2. MP3 Streaming with and/or without HLS chunking via push/post to a CDN server and/or via an in-built HTTP distribution server. Optional TS encapsulation (Section 4.2.3)
3. AAC/MP3 Distribution via an Icecast distribution server or UDP unicast/multicast over network (with optional TS encapsulation) (Section 4.2.4)

Following are the steps of Stream configuration from this panel.

4.2.1 Stream’s Common Settings Panel

The Common Settings tab (Figure 4.2.1) defines the identity of the stream and the network interface it uses. The configuration window is titled **StreamMax**, and the **Operator Guide** at the bottom of the window summarises what each tab (Common, AAC, MP3, Icecast/UDP) does.

- **Streaming Interface:** Selects the network adapter this channel streams from. Leave it on **AUTO** to let the system choose the interface automatically, or select a specific adapter.
- **Stream Name:** The base name used in every stream URL (for example, name.aac or name_128kbps.m3u8). Each channel must have a distinct stream name to avoid conflicts.
- **Local HTTP Port (aac):** Port used for local AAC HTTP-pull streaming from the built-in server.
- **Local HTTP Port (m3u8):** Port used for local HLS (.m3u8) HTTP-pull streaming from the built-in server.

These local-server ports are used only for local .aac/.mp3 and .m3u8 pull streaming and are configured under **Server Settings** (right-click the server in SoundMaxManager™), not on this tab.

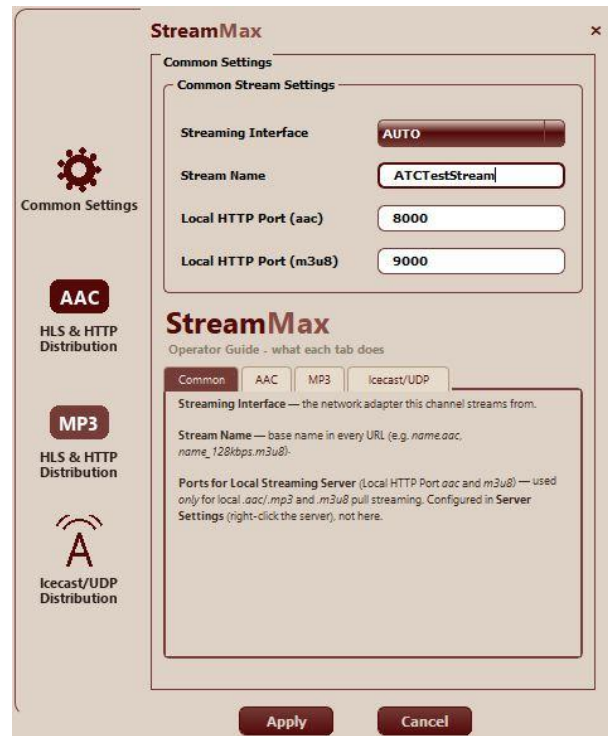


Figure 4.2.1: Common Settings Panel

4.2.2 AAC HLS & HTTP Distribution Settings

The **AAC HLS/HTTP Distribution** tab (Figure 4.2.2) provides complete control over AAC encoding, HLS segmenting, and delivery. Tick **AAC HLS/HTTP Distribution Enable** to activate AAC streaming, then configure the four groups below.

(A) AAC Encoder Settings

- **Codec Profile (LC / HE-v1 / HE-v2):** Selects the AAC profile. **LC** (Low Complexity) is ideal for high-fidelity streaming at higher bitrates, while **HE-v1/HE-v2** (High Efficiency) deliver superior quality at lower bitrates.
- **Dual Rate Encoding (ON/OFF):** When set to **ON**, two selected bitrates are encoded simultaneously so the player can switch quality based on the listener's network speed. When **OFF**, a single selected bitrate is encoded.
- **Bitrates Selection:** Choose the bitrate(s) to encode from **24, 32, 64, 96, 128, 192, 256, or 384 kbps**. With Dual Rate Encoding **ON**, select two — it is recommended to pair one higher bitrate (e.g., 128 or 192) with one lower bitrate (e.g., 32 or 64) so mobile listeners stay connected in poor-signal areas.

(B) HLS Chunking & M3U8 Playlist Generation

Enable this group to prepare the stream for adaptive-bitrate (HLS) playback.

- **Chunk Duration (3–10 sec):** Length of each HLS audio segment (default 10 s).
- **Encapsulation:** Segment container format — **ADTS** (default) or **TS** (MPEG-TS).

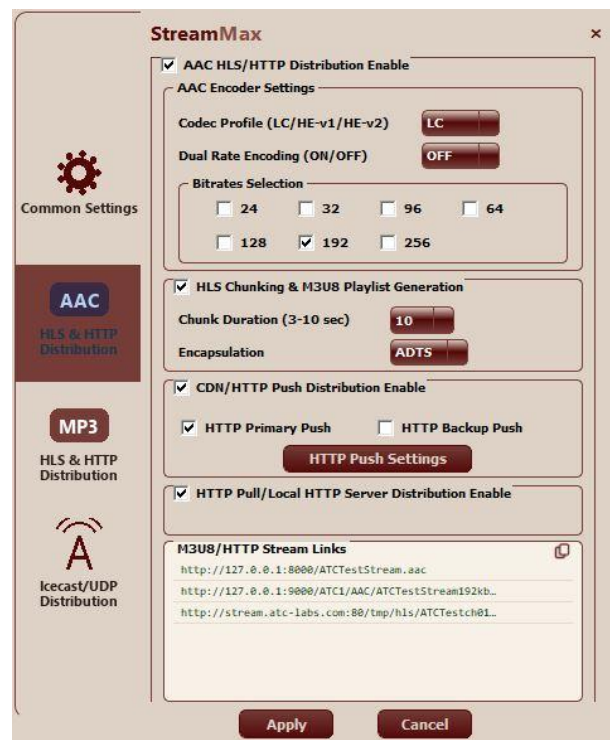


Figure 4.2.2: AAC HLS & HTTP Distribution Settings Panel

(C) CDN/HTTP Push Distribution Enable

Enable this group to push the stream to remote destinations such as a CDN (AWS, Akamai, BitGravity, etc.).

- **HTTP Primary Push / HTTP Backup Push:** Send the stream to a primary and, optionally, a backup remote destination for redundancy.
- **HTTP Push Settings:** Opens the HTTP Push Settings panel (Figure 4.2.3), where the destination URLs and credentials are configured.

The **HTTP Push Settings** panel (Figure 4.2.3) defines where the HLS playlist and segments are published — a CDN or an ingest endpoint — and how that connection is authenticated. The panel is arranged in three groups: **Destination**, **Authentication** and **Transport**. Fields marked with an asterisk are mandatory; the asterisk set changes with the authentication method selected.

Destination

- **Primary host *:** The destination server address, for example stream.atc-labs.com. Enter the host name or IP only — no scheme and no path.
- **Primary path *:** The directory on that server where the .m3u8 playlist and audio segments are written, for example tmp/hls/ATCTestch03.

- **Backup host / Backup path:** An optional second destination used for redundancy. Leave both set to NULL if no backup is required. When a backup is used, give it a path distinct from the primary (for example by adding a -b suffix) so the two sources stay separate.

Authentication

- **Method:** Selects the scheme the destination expects. The fields below this selector change to match the method chosen — the illustration shows AWS SigV4, used for publishing to Amazon S3 and S3-compatible endpoints.
- **Access key ID * / Secret key *:** The credential pair issued by the cloud provider for the target bucket or endpoint. Both are required for SigV4.
- **Region *:** The region the destination endpoint belongs to, for example us-east-1. The value must match the bucket's region or signing will fail.
- **Service:** The signing service name used in the SigV4 signature. This defaults to s3 and should only be changed if the destination documents a different service name.

HTTP Push Settings
Publish HLS segments to a CDN or ingest endpoint

Destination

Primary host *

Primary path *

Backup host

Backup path

Authentication

Method

Access key ID *

Secret key *

Region *

Service

Transport

☐ Use TLS (HTTPS)

Port

HTTP verb

Test Connection

Fill the details above, then click Test Connection.

Figure 4.2.3: HTTP Push Settings Panel

Transport

- **Use TLS (HTTPS):** Tick to publish over HTTPS instead of plain HTTP. Most CDNs and cloud ingest endpoints require this.
- **Port:** The destination network port — typically 80 for HTTP or 443 for HTTPS. Port 80 is often blocked by strict firewalls, so 443 with TLS enabled is the safer default.
- **HTTP verb:** The method used to upload each segment. PUT is normal for object-storage destinations; POST is used by ingest endpoints that expect posted segments. Set this to whichever the destination documents.

After filling in the details, click **Test Connection** to verify that the destination accepts the credentials and path, then click **Save** to store the settings. **Cancel** closes the panel without applying changes.

(D) HTTP Pull / Local HTTP Server Distribution Enable

Enable this group to distribute the stream from the unit's own built-in HTTP server. The Models Q24-6111 and Q24-6111HD then act as an internal Icecast-compatible origin, accepting HTTP-pull requests for the HLS .m3u8 playlist and chunks (or the single-rate AAC stream) directly from local listeners or players at this unit's IP address — ideal for in-house monitoring, office-wide broadcasts, or low-latency distribution within a private network.

M3U8 / HTTP Stream Links

Once settings are applied, the generated URLs for the AAC HLS playlist and the raw HTTP stream appear in this list in the lower section of the panel, ready to be copied and distributed to listeners or embedded into players.

4.2.3 MP3 HLS & HTTP Distribution Settings

The **MP3 HLS/HTTP Distribution** tab (Figure 4.2.4) mirrors the AAC tab for applications requiring legacy compatibility or specific bandwidth. Tick **MP3 HLS/HTTP Distribution Enable** to activate MP3 streaming, then configure the groups below.

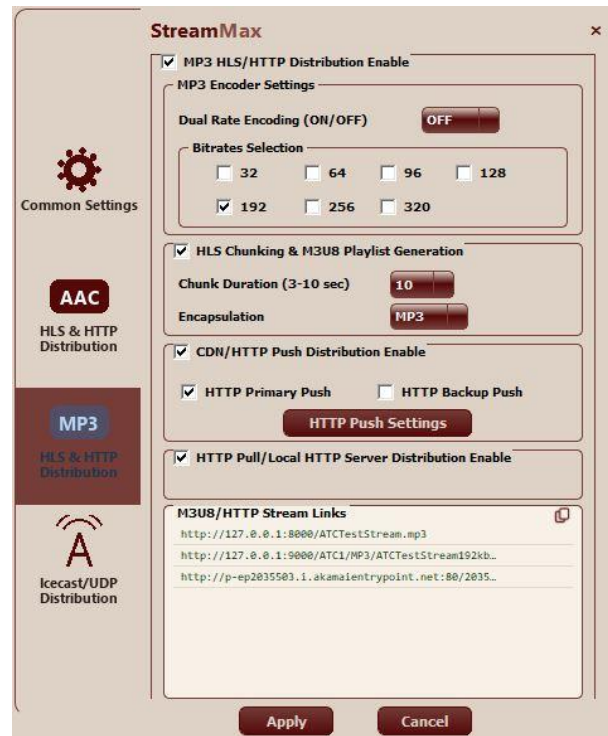


Figure 4.2.4: MP3 HLS & HTTP Distribution Settings Panel

(A) MP3 Encoder Settings

- **Dual Rate Encoding (ON/OFF):** When ON, two selected bitrates are encoded simultaneously for adaptive switching; when OFF, a single bitrate is used. MP3 has no codec-profile selection.
- **Bitrates Selection:** Choose from **32, 64, 96, 128, 192, 256, or 320 kbps**. MP3 supports up to 320 kbps.

(B) HLS Chunking & M3U8 Playlist Generation

- **Chunk Duration (3–10 sec):** Length of each HLS segment (default 10 s).
- **Encapsulation:** Fixed to **MP3** for MP3 HLS output.

(C) CDN/HTTP Push Distribution Enable

As with AAC, enable **HTTP Primary Push** and/or **HTTP Backup Push** to send the stream to remote servers, and use **HTTP Push Settings** to configure the destination URLs and credentials (the panel is the same as Figure 4.2.3).

(D) HTTP Pull / Local HTTP Server Distribution Enable

Distributes the MP3 HLS playlist and stream from the unit's built-in HTTP server for local pull requests, exactly as described for AAC.

M3U8 / HTTP Stream Links

The generated URLs for the MP3 HLS playlist or the raw HTTP stream appear here for copying. Use MP3 HLS for maximum compatibility with older web players and legacy mobile devices that may struggle with AAC/HE-AAC profiles.

4.2.4 Icecast / UDP Distribution Settings

The **Icecast/UDP Distribution** tab (Figure 4.2.5) handles web broadcasting via an Icecast distribution server or network streaming via UDP unicast/multicast. Tick **Icecast/UDP Distribution Enable** to activate this output.

Server / Protocol Settings

- **Protocol:** Select the delivery method — **Icecast** (delivers the stream to an authorised Icecast server), **Unicast**, or **Multicast** (transmit over a private LAN).
- **Select Server:** Choose a pre-authorised ATC Labs Icecast server — **ATC Labs Asia** or **ATC Labs North America** — or select **Custom** to enter your own server.
- **Host:** The Icecast server address (e.g., streamasia.atc-labs.com). For Unicast/Multicast, enter the target IP address.
- **Port:** The destination network port (default 8000).
- **Server Password:** The source/mount password used to connect to the Icecast server.
- **Organisation Tag:** The mount-point/station tag that identifies your stream on the server.
- **Use ATC Icecast Default Password:** When ticked, the built-in default password for the selected ATC Labs Icecast server is used instead of a manually entered one

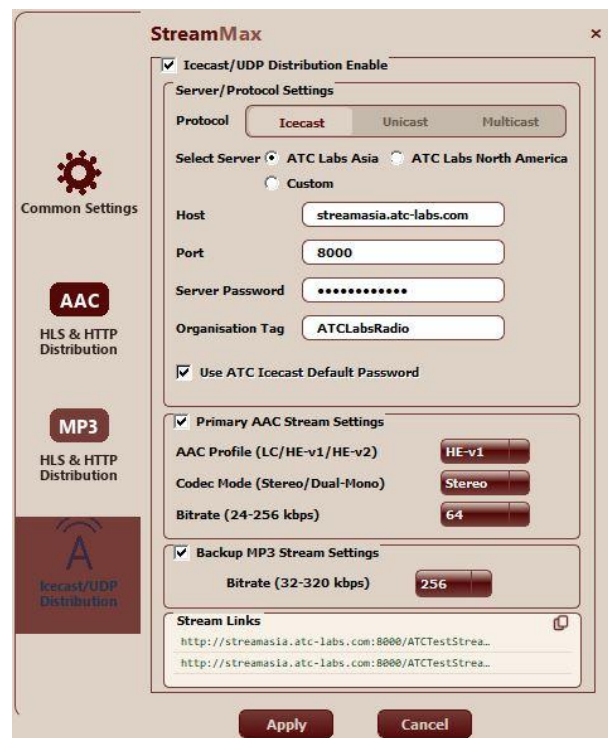


Figure 4.2.5: Icecast / UDP Distribution Panel

Primary AAC Stream Settings

Tick the checkbox to enable the primary high-efficiency AAC stream.

- **AAC Profile (LC / HE-v1 / HE-v2):** Select the AAC profile for the primary stream.

- **Codec Mode (Stereo / Dual-Mono):** Choose the channel mode for the primary stream.
- **Bitrate (24–384 kbps):** Set the target bitrate for the primary AAC stream.

Backup MP3 Stream Settings

Tick the checkbox to enable a secondary MP3 stream for devices or legacy software that do not support AAC.

- **Bitrate (32–320 kbps):** Set the bitrate for the backup MP3 stream.

Stream Links

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

4.3 Nielsen™ Configuration

Perceptual SoundMax™ supports fully certified in-box Nielsen™ watermarking. The Nielsen™ Configuration Panel (Figure 4.3.1) is accessed via the SoundMaxManager™ detailed view (button labeled as ③ in Figure 4.1). It 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 image shows a 'NIELSEN CONFIGURATION PANEL' window with a close button (X) in the top right corner. The panel contains several 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

Perceptual SoundMax™ supports full functionality in-box RDS Encoding. The Internal RDS Encoder Panel (Figure 4.4.1) is accessed via the SoundMaxManager™ detailed view (button labeled as ④ in Figure 4.1). It 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 numeric input field 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 labeled "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.

Program 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 IP Audio functionality is also fully configurable via the SoundMaxManager™ detailed view (button labeled as  in Figure 4.1). 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

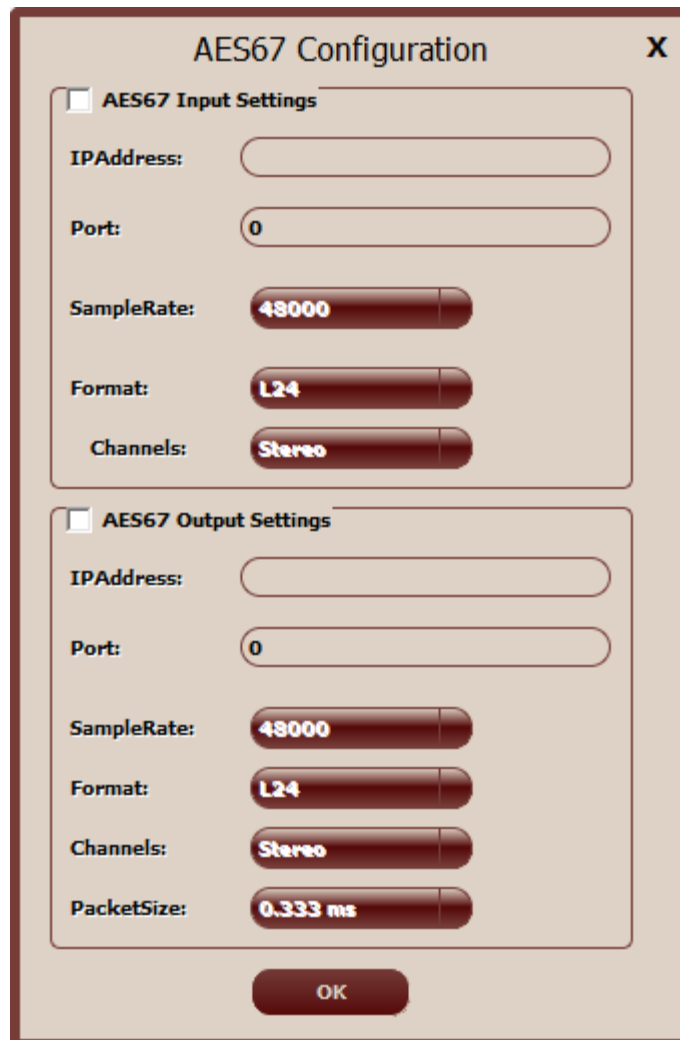
The **AES67 Configuration Panel** (Figure 4.5.1) enables high-performance Audio-over-IP (AoIP) interoperability. This allows the system to send and receive uncompressed, low-latency audio across a local network.

Important Note: The Q24-6111 / 6111HD 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. The "Input Settings" section includes fields for IP Address, Port (set to 0), Sample Rate (set to 48000), Format (set to L24), and Channels (set to Stereo). The "Output Settings" section includes fields for IP Address, Port (set to 0), Sample Rate (set to 48000), Format (set to L24), Channels (set to Stereo), and Packet Size (set to 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 Aideal+ Audio AI Genie Controls

The Aideal+ Audio Processing Genie is an enhanced AI-driven tool that intelligently analyzes your on-air audio and provides automated recommendations and adaptive tuning to optimize processing performance. As described in Section 3.4 it is fully configurable via the front screen TFT of the hardware unit. Optionally, on the SoundMaxManager™ interface, the Aideal+ Genie is accessed by clicking the AI+ button on the Main Detailed View Screen (Button labeled as ❹ in Figure 4.1), which opens the Aideal+ Audio Processing Genie window (Figures 4.6.1 and 4.6.2).

The Aideal+ window provides two primary functions, selectable from the persistent navigation column on the left side of the screen:

- **Aideal+ Profile Recommendation:** Activates the AI-based Audio Processing Genie to listen to your station's on-air audio and recommend the best-matching processing profiles. The Station Engineer should audition and approve all on-air profile configurations before applying.
- **Adaptive Processing:** Engages the AI-based Audio Processing Genie to continuously adapt profile parameters based on changing audio characteristics, ensuring optimal processing in real time.

Both functions can be opened from either side of the navigation column at any time. Click the X button in the top-right corner to close the Aideal+ window and return to the Main Screen.

Aideal+ Profile Recommendation

Clicking **Recommend Profile** in the left-hand navigation column opens the Profile Recommendation screen (Figure 4.6.1). This feature enables the AI engine to listen to the incoming audio signal, analyze its characteristics, and recommend the most suitable processing profiles from the available factory presets and custom profiles.

The header of this screen reads "Let Aideal+ AI based Audio Processing Genie Listen to your Radio and Recommend Best Profiles. (Station Engineer, please audition and approve all On-Air Profile Configurations)" — a reminder that all recommendations must be reviewed and approved by the Station Engineer prior to going on-air.

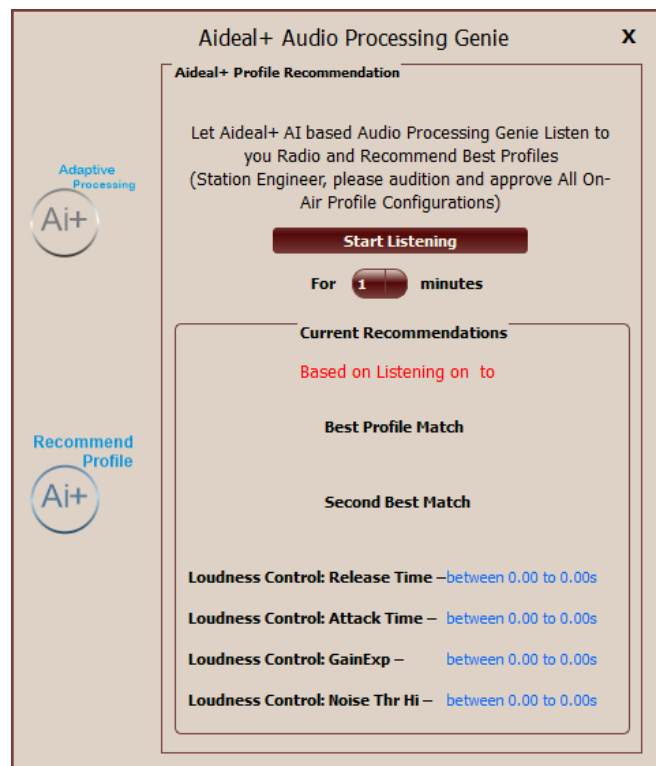


Figure 4.6.1: Aideal+ Profile Recommendation

Operating Procedure

1. Set Listening Duration

Use the numeric stepper next to "**For [] minutes**" to set the listening duration. A longer listening window allows the AI to capture a broader range of program material and produces more accurate recommendations. For most stations a listening period of 5 – 15 minutes covering a representative cross-section of the program schedule is recommended.

2. Start Listening

Click the **Start Listening** button to begin the analysis. The screen transitions to a listening-in-progress state, and the same button changes to **Stop Listening**. While listening is active, the screen displays:

- An animated **LISTENING** indicator confirming that audio analysis is in progress.
- **Listening Started at** — timestamp showing when the analysis began.
- **Listening will End at** — timestamp showing the scheduled end time based on the configured duration.

The user may click **Stop Listening** at any time to end the analysis early. In that case, results from the previous completed analysis (if any) remain displayed in the **Current Recommendations** area.

3. Review Recommendations

Once the listening period is completed, the Current Recommendations area populates with the analysis results:

- **Based on Listening on [date / time range]** Shows the date and time window of the analyzed audio segment, displayed in red text.
- **Best Profile Match** — Displays the top recommended processing profile (e.g. "Custom-24-Jazz-Classical"). Click the corresponding **APPLY** button to activate this profile on the current output path.
- **Second Best Match** — Displays the second-best recommended profile (e.g. "Custom-24-Country"). Click the corresponding **APPLY** button to apply this profile instead.

Below the profile matches, the screen lists Further Tuning Guidance — specific parameter adjustment suggestions to further refine the selected profile. The recommended ranges are shown in blue text alongside each parameter:

- **Loudness Control: Release Time** — between X.XX to Y.YY s
- **Loudness Control: Attack Time** — between X.XX to Y.YY s
- **Loudness Control: GainExp** — between X.XX to Y.YY
- **Loudness Control: Noise Thr Hi** — between X.XX to Y.YY s

(Where X.XX and Y.YY are the AI-suggested lower and upper bounds for each parameter, populated after analysis completes.)

The Station Engineer should evaluate these recommendations and apply them using the Tuning interface (refer to Section 4) as needed to achieve the desired on-air sound.

To run a new analysis session, click Start Listening again. To switch to the Adaptive Processing function, click Adaptive Processing in the left-hand navigation column.

Aideal+ Adaptive Processing (Auto Tune Genie)

Clicking Adaptive Processing in the left-hand navigation column opens the Auto Tune Genie screen (Figure 4.6.2). This feature allows the AI engine to dynamically adjust key audio enhancement parameters based on the characteristics of the incoming audio, providing real-time intelligent processing optimization.

The header of this screen reads "Let Aideal+ AI based Audio Processing Genie Adapt Profile Parameters based on changing audio characteristics."

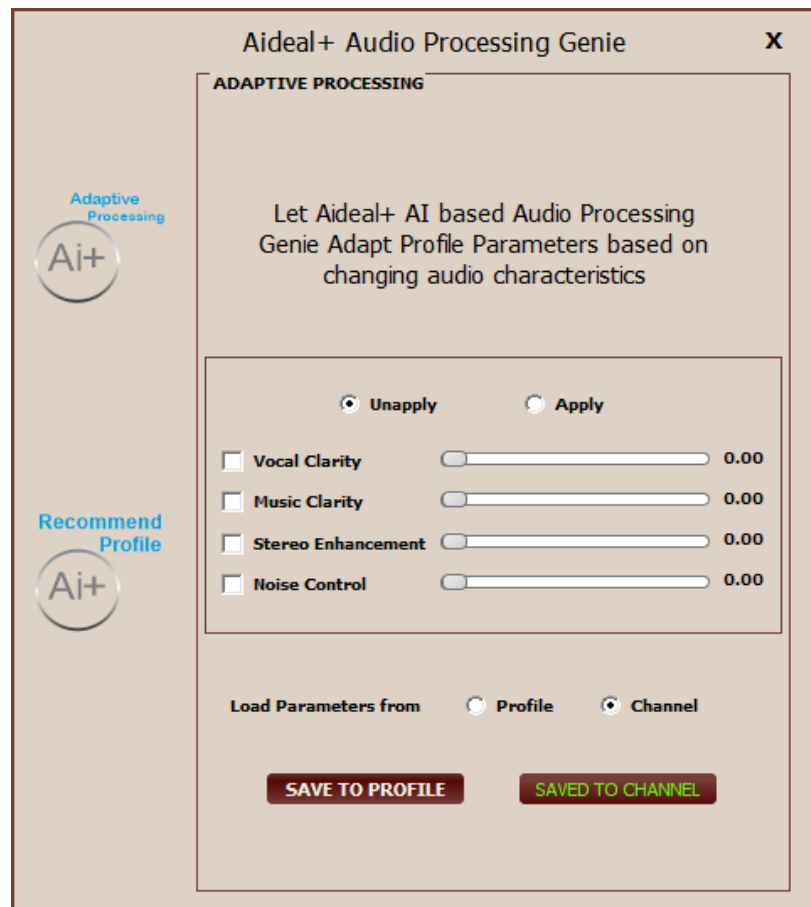


Figure 4.6.2: Aideal+ Adaptive Processing

Adjustable Parameters

The Auto Tune Genie provides four adjustable parameters, each controlled via a slider with a numeric readout on the right (range 0.00 to 1.00) and an enable/disable checkbox on the left:

- **Vocal Clarity** — Controls the degree of vocal enhancement and intelligibility improvement applied to speech content.
- **Music Clarity** — Controls the degree of musical detail enhancement and spectral clarity.
- **Stereo Enhancement** — Controls the degree of stereo image widening and spatial enhancement applied to the audio.
- **Noise Control** — Controls the aggressiveness of the AI-driven noise reduction system.

For each parameter, **0.00** disables enhancement on that dimension and **1.00** applies the maximum amount of AI-driven adjustment. Use the checkbox alongside each slider to individually enable or disable specific enhancements without losing the slider value.

Apply / Unapply

A pair of radio buttons at the top of the parameter area controls whether the Auto Tune Genie settings are active on the processing path:

- **Apply** — Enables the current Auto Tune Genie settings on the active processing path.
- **Unapply** — Disables the Auto Tune Genie adjustments and reverts to the standard profile settings.

Switching between Apply and Unapply takes effect immediately, allowing the engineer to A/B compare the Auto Tune Genie processing against the underlying profile.

Load Parameters From

Below the parameter sliders, the Load Parameters from option allows the user to choose the source of the Auto Tune Genie parameter values:

- **Profile** — Loads the saved Auto Tune Genie values stored in the currently active processing profile. Use this option when the parameter values are intended to be tied to the program content / format.
- **Channel** — Loads the saved Auto Tune Genie values stored in the currently selected output channel (e.g. a specific FM, HD, or streaming output). Use this option when the parameter values are intended to be tied to the delivery path rather than the program content.

The selected source is indicated by the active radio button.

Saving Settings

Two persistence options are available at the bottom of the screen:

- **SAVE TO PROFILE** — Saves the current Auto Tune Genie parameter values into the active processing profile for persistent use. The current Auto Tune Genie configuration will then load automatically whenever the active profile is recalled.

- **SAVE TO CHANNEL** — Saves the current Auto Tune Genie parameter values to the currently selected output channel configuration, allowing different output channels (e.g. FM vs HD vs streaming) to have independent adaptive processing configurations. After a successful save the button label changes to SAVED TO CHANNEL to confirm that the settings have been written.

4.7 Diversity Delay (Model Q24-6111HD Only)

The Diversity Delay dialog (Figure 4.7.1) 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 Diversity Delay parameters can be configured via the SoundMaxManager™ interface by clicking the DELAY button in the main path of the Detailed View Screen (Button labeled as ⑦ in Figure 4.1),

The screenshot shows the 'Diversity Delay' dialog box. At the top, it says 'Diversity Delay HD' with a close button 'X'. Below this, there's a 'TOTAL DELAY' section. It displays '8.000000 seconds' and '352800 samples'. To the right of these values is a button labeled 'SR 44100 Hz'. Below the 'TOTAL DELAY' section is section '01 COMPOSE DELAY'. It has three input fields: 'Seconds' (8.0), 'Milliseconds' (0), and 'Microseconds' (0.00). Below section '01' is section '02 FINE ADJUST'. It has one input field: 'Samples Adjusted' (0). At the bottom of the dialog are two buttons: 'Reset' and 'APPLY'.

Fig 4.7.1: Diversity Delay

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.

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 μ s.

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 describes the audio processing architectures, algorithms, and controls embedded in this product.

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 algorithms. 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 band affects a smaller portion of audio, resulting in a more open and richer sound.

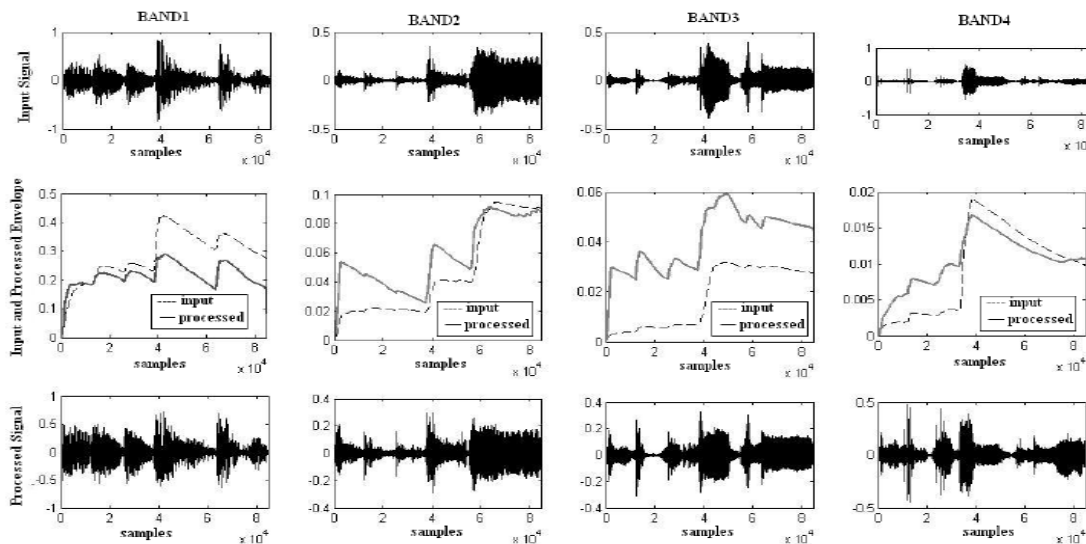


Figure 5.1.1: Perceptual SoundMax™ Envelope Processing in multiple *high resolution* frequency bands

Linear Phase Audio Processing: The multi-band processing in Perceptual SoundMax™ is completely 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 for maintaining fidelity; moreover, in digital broadcasting 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 linear-phase processing with high temporal resolution that does not cause any phase distortion 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* non-linear envelope gains shape 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 of the bands are fully configurable. Moreover, the level of processing in each of the regions is also independently controllable.

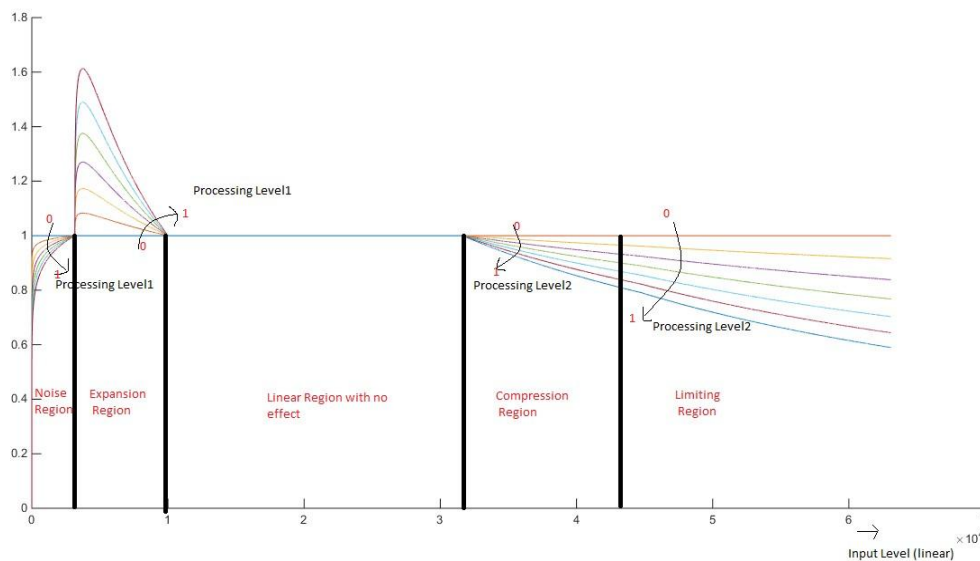


Figure 5.1.2: The configurable shape of Perceptual SoundMax™ Envelope Processing in each of the 24 bands: non-linear processing gain structure in multiple regions

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 in achieving a *clean sound* for the processed audio and in 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).

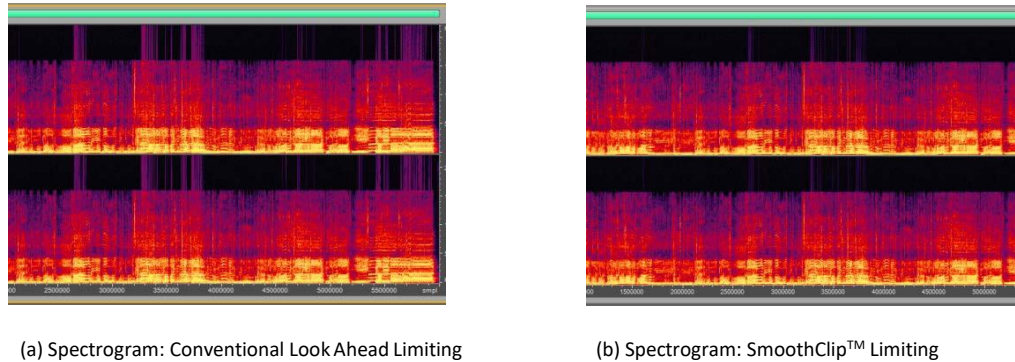


Figure 5.1.3: SmoothClip™ technology in Perceptual SoundMax™ substantially limits increase in high frequency energy during look ahead limiting

Extensive use of AI Based Audio Processing Algorithms: The Aideal 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, and also used in innovative features such as Aideal+ Profile Recommendation Tool and Auto-Tune genie.

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.

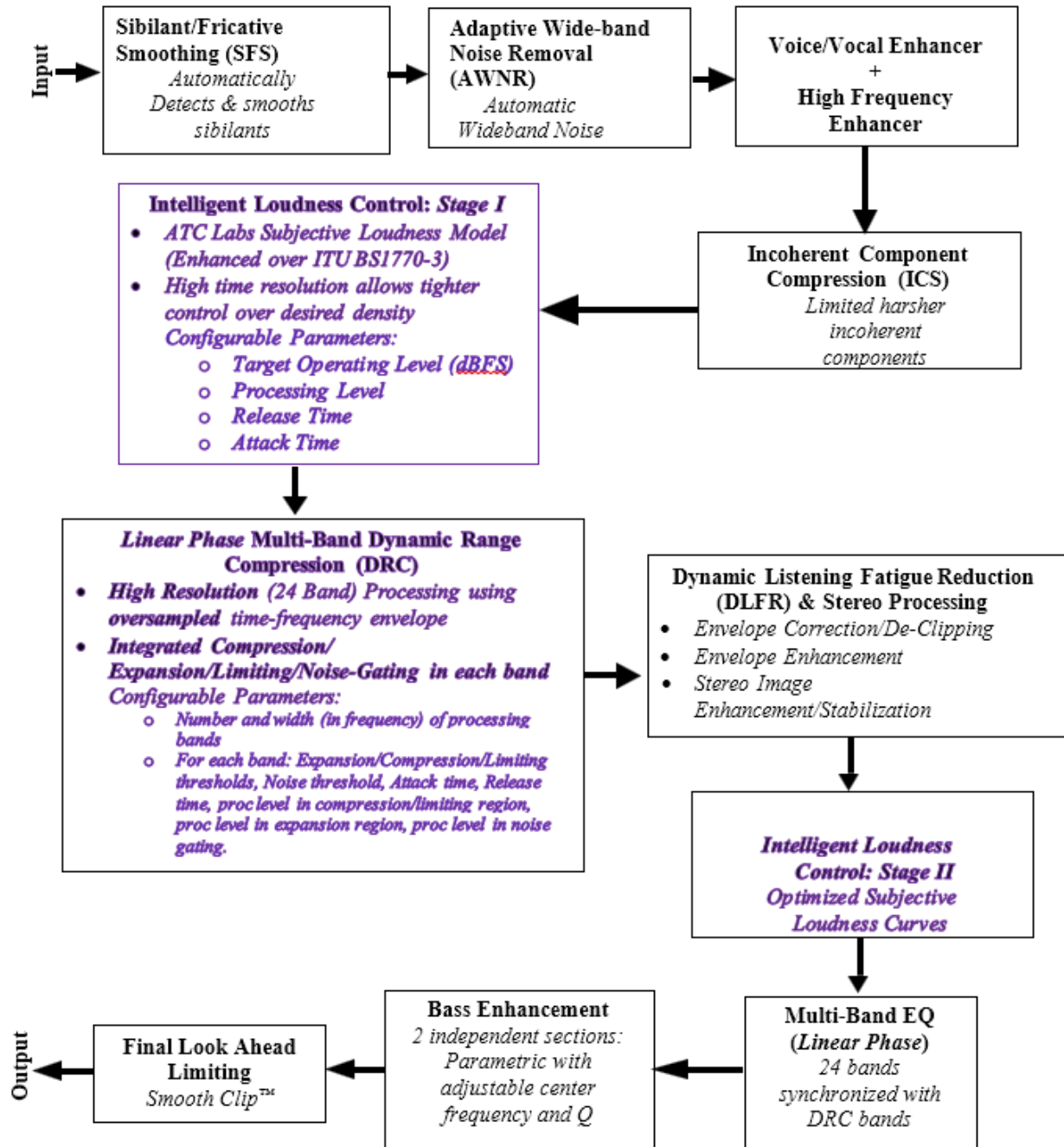


Figure 5.1.4: Processing Architecture – Signal Flow

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 TFT 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 labeled as 1); clicking on this button opens the selections panel shown in Figure 5.2.1 below:

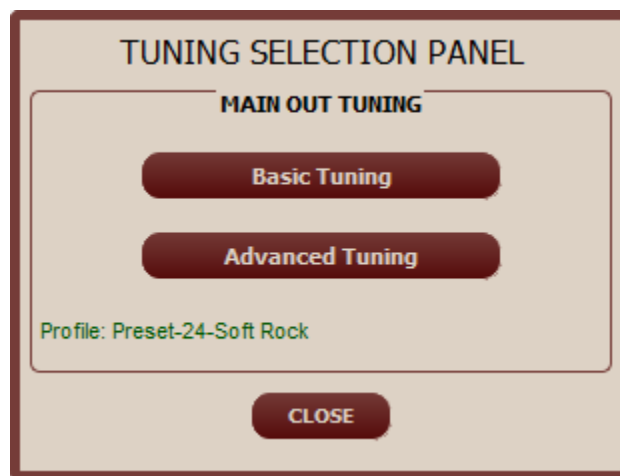


Fig 5.2.1: SoundMaxManager™ Tuning Mode Selection Screen.

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 the 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 to 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 a 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 brings the user to this screen as shown in Figure 5.2.3. There about 1000 parameters exposed on this screen, and these can be tuned independently for each of the signal paths (such as MAIN and STREAM on all models and HD1/HD2/HD3 on Perceptual SoundMax™ Q24-6111HD). 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 processor, custom profiles created on one processor may also be exported and made available on the other processors.

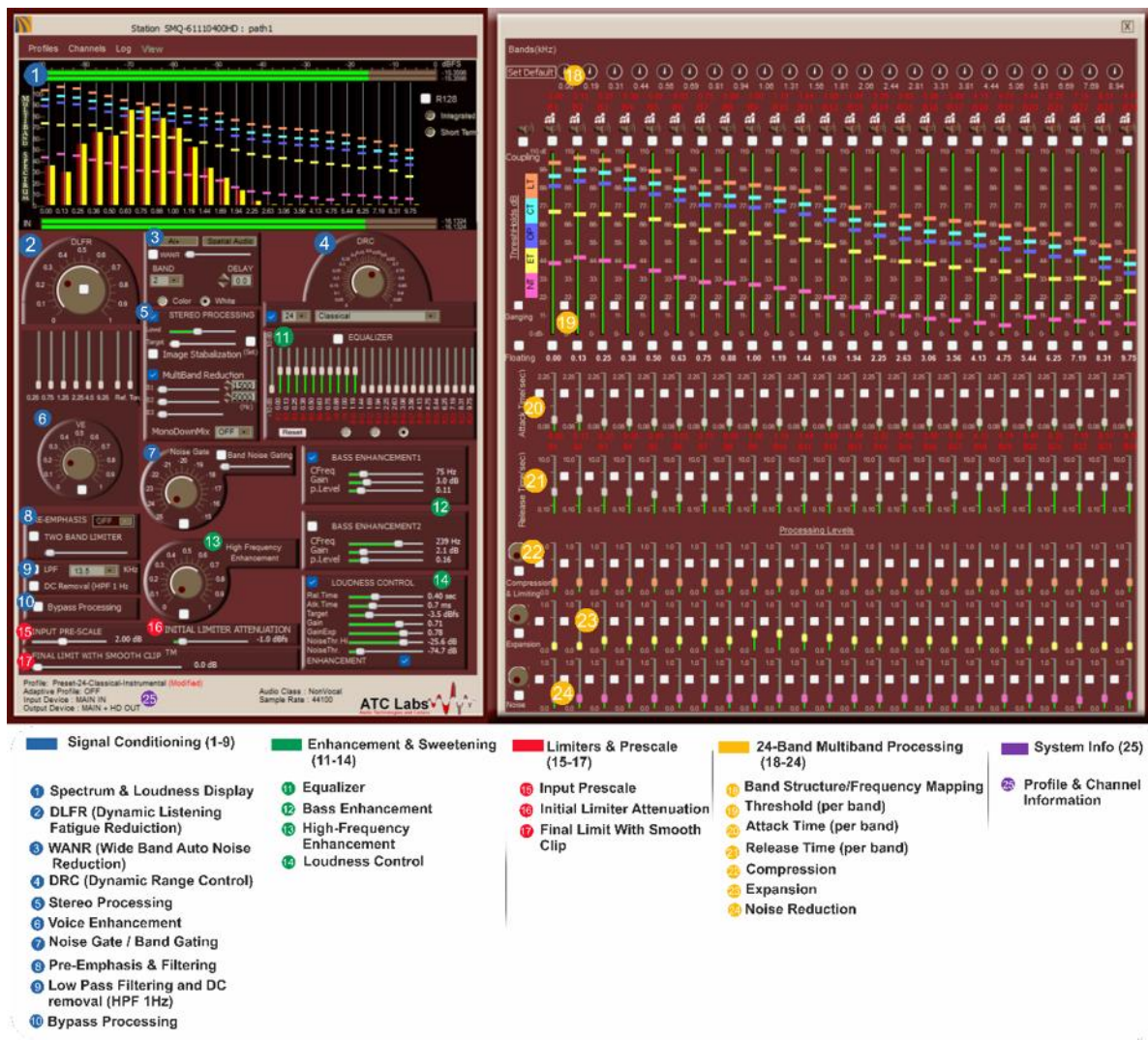


Figure 5.2.3: SoundMaxManager™ Advanced Tuning Screen

As described above following different processing modules are available:

- AI based auto tuning controls
- 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)

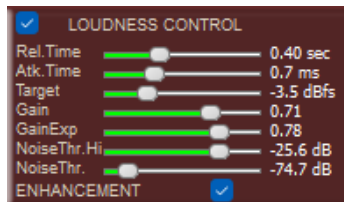
- Dynamic Listening Fatigue Reduction (DLFR)
- Pre-Emphasis
- Low Pass Filter
- Initial & Final Look Ahead Limiting
- Input pre-scale

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; higher values 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 in case of digital broadcasts. 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 Expansion):** 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

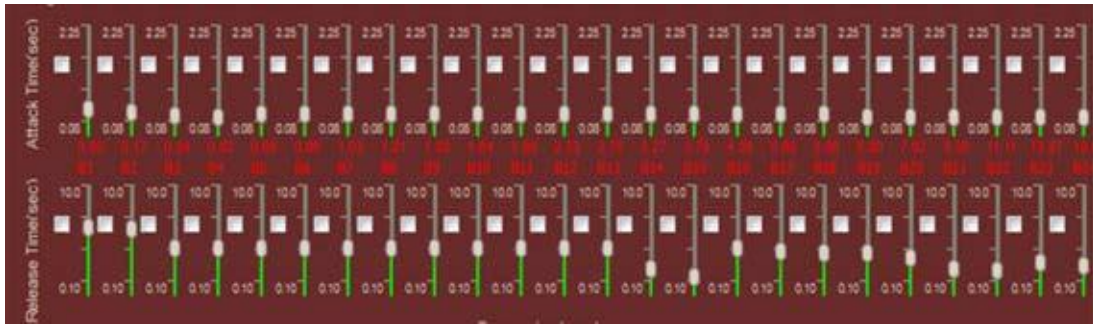
characteristic is employed. To change the dynamic range in different bands individually, this module is provided with greater functionality than any other product available in the market. At the heart of the processor is a specially designed filter bank for generating an extremely detailed time-frequency 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 helps 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 to 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.



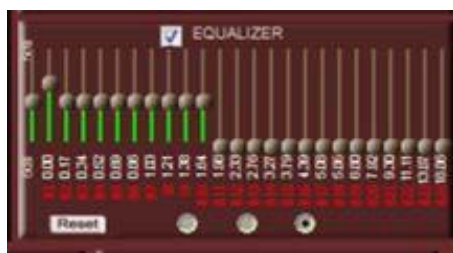
- **Processing Levels:** The interface allows for independent control of processing levels of DRC for each band, in three regions of non-linearity (Figure 5.1.2): (i) compression/limiting, (ii) expansion, (iii) noise-gating. Correspondingly there are 24 sliders for compression/limiting processing level, 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 dials (with the constraint that center frequency of each band is upper/lower bounded by respectively the adjacent higher/lower band).

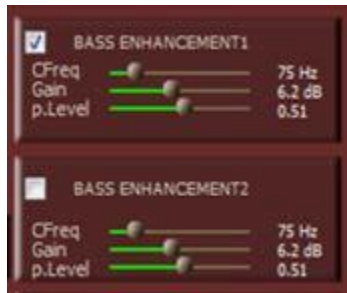


- **Multi-Band Graphic Equalizer:**



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.

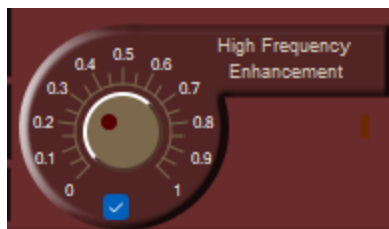
- **Bass Enhancement:**



Perceptual SoundMax™ offers the options for using up to 2 stages of parametric bass boost. Either first 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:

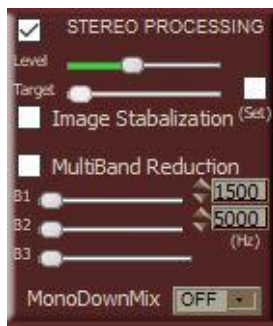
- **CFreq:** this is the center frequency of the bass boost notch or peak (in Hz)
- **Gain:** this specified in dB controls the height of the bass boost notch.
- **P. Level:** This processing level controls the width (i.e. Q-value) of the bass boost notch/peak; a higher level corresponds to a wider peak.

- **High Frequency Enhancement:**



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

- **Stereo Processing:**

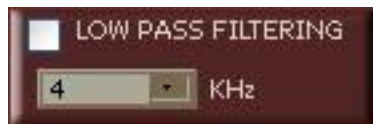


This module controls the stereo balance in the audio. It may widen or shrink the stereo image depending on the characteristics of the stereo audio and the slider setting – i.e. it serves as a dynamic range control for the stereo image. This tool helps in maintaining a consistent stereo image which would 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



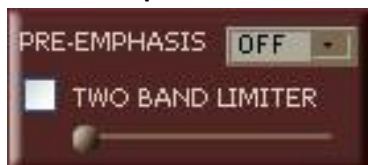
- **Noise Gating/Band Noise Gating:**



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 behaviour 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.

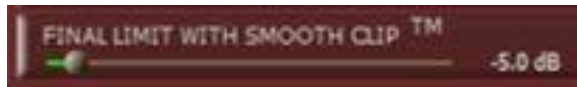
- **Pre-Emphasis:**



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, the TFT is set from the TFT/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:**



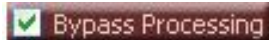
This slider displays/sets the final level for the look ahead limiter. For the MAIN path in hardware processors, the final limiter value is set from front TFT/LCD controls as described above and this slider will simply follow or display the value set there. For streaming and digital broadcast signal paths 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 enhances the vocal content by enhancing corresponding spectral vocal harmonic peaks while preserving other spectral content. This module may be used for instance in sports/news broadcasts to make the commentator's voice more intelligible, in situations where high background audience noise may 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 Processing Parameter Touch-Up via TFT Screen

While the user will typically use SoundMaxManager™ software for advanced tuning of sound quality, in Perceptual SoundMax™ Models Q24-6111 and Q24-6111HD limited profile tuning is also available via the front TFT panel. *This section describes these TFT controls in detail which mirror the corresponding **Advanced Tuning Control** adjustments described in Section 5.2.2; therefore, for the interpretation of different tunable parameters and algorithmic modules users may refer to Section 5.2.2.* The TFT Tune screen - Figure 5.3.1 is accessed by pressing the TUNE soft key on the on **Profile Selection/View** screen

in Figure 3.2.1. It provides a centralized control panel where the user can selectively enable, disable, and access more detailed configuration for each individual processing block in the active profile.



Figure 5.3.1: Tune Screen — Processing Block Selection

The screen header displays the currently active path and on-air profile (e.g., "Main: OnAir: Preset-24-Classical-Instrumental"), confirming which profile's processing blocks are being modified.

Processing Block Controls

The Tune screen presents fourteen individual processing blocks arranged in two columns. Each block has two associated controls:

Enable/Disable Checkbox: A checkbox to the left of each block name. When checked (✓), the processing block is active and contributes to the audio chain. When unchecked, the block is bypassed without removing its stored parameter values. This allows the user to A/B compare the effect of any individual processing stage without losing the configured settings.

VIEW Button: Pressing the VIEW soft key next to any block opens that block's dedicated parameter screen, where the user can adjust all configurable parameters in detail (thresholds, attack/release times, gains, frequencies, etc.).

The following processing blocks are available for selective enable/disable and detailed tuning.

5.3.1 Low-pass filter

The Low-Pass Filter screen (Figure 5.3.2) sets the output bandwidth by configuring the LPF cutoff frequency. Frequencies above the cutoff are attenuated, removing unwanted high-frequency content above the FM transmission limit.



Figure 5.3.2: Low-Pass Filter Screen

The screen header shows the active path and current on-air profile (e.g., "PATH 1 — OnAir: Preset-24-Classical-Inst..").

Controls

- **OUTPUT BANDWIDTH (LPF Cutoff Freq)** — Sets the cutoff frequency in kHz. The value is entered using three-digit fields (tens, units, and decimal), each with + and – buttons to increment or decrement the digit. Typical setting for FM broadcast is 13.5 kHz to 15.0 kHz.

Action Buttons

- **SAVE** — Saves the configured cutoff frequency to the active profile.
- **BYPASS** — Bypasses the LPF stage for A/B comparison.
- **BACK** — Returns to the Tune screen without saving.

5.3.2 Master loudness control

The Loudness Control screen (Figure 5.3.3) configures the master loudness management stage, which determines overall on-air loudness, consistency, and noise floor handling. *A brief description of these parameters is repeated below but the users are encouraged to refer to Section 5.2.2 for more detailed understanding.*

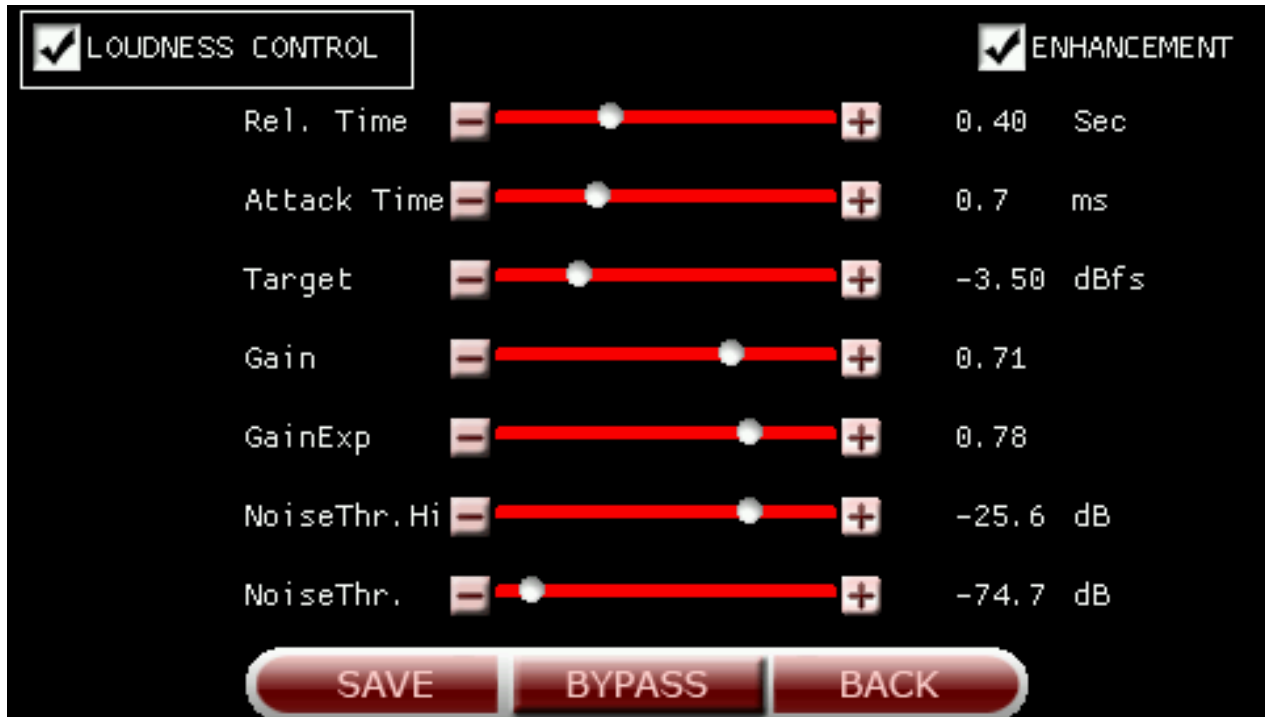


Figure 5.3.3: Loudness Control Screen

Top Toggles

- **LOUDNESS CONTROL** — Enables or disables the loudness control stage.
- **ENHANCEMENT** — Enables or disables the loudness enhancement processing.

Controls

Each parameter is adjusted using a slider with – and + buttons. The current value is displayed to the right of each slider.

- **Rel. Time** — Release time in seconds. Controls how quickly the loudness processor releases gain after a loud passage (e.g., 0.40 sec).
- **Attack Time** — Attack time in milliseconds. Controls how quickly the processor responds to incoming level increases (e.g., 0.7 ms).
- **Target** — Target output level in dBfs (e.g., -3.50 dBfs).
- **Gain** — Overall gain factor applied by the loudness stage in the compressive region (e.g., 0.71).
- **GainExp** — Gain exponent that shapes the loudness curve response in the expansion region (e.g., 0.78).

- **NoiseThr. Hi** — Upper noise threshold in dB. Sets the level above which the signal is treated as program audio (e.g., -25.6 dB).
- **NoiseThr.** — Lower noise threshold in dB. Sets the noise floor below which the signal is treated as silence/noise and may be gated (e.g., -74.7 dB).

Action Buttons

- **SAVE** — Saves the configured cutoff frequency to the active profile.
- **BYPASS** — Bypasses the LPF stage for A/B comparison.
- **BACK** — Returns to the Tune screen without saving.

5.3.3 Voice enhancement

The Voice Enhancement screen (Figure 5.3.4) sets the intensity of the vocal clarity and presence processing applied to the audio signal.

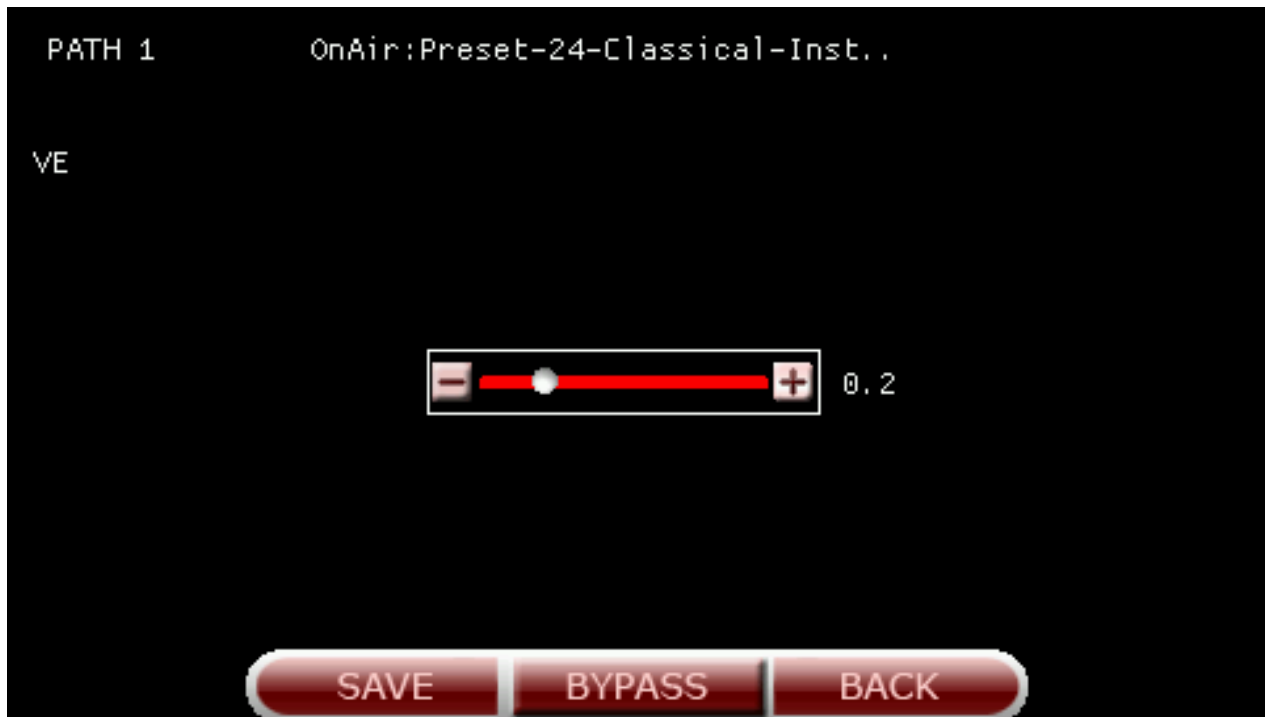


Figure 5.3.4: Voice Enhancement Screen

The screen header shows the active path and current on-air profile (e.g., "PATH 1 — OnAir: Preset-24-Classical-Inst..").

Controls

- **VE** — Sets the voice enhancement intensity. Adjustable from **0.0** (no enhancement) to **1.0** (maximum) using the slider with – and + buttons. The current value is displayed to the right of the slider (e.g., 0.2). Higher values increase vocal presence and intelligibility — particularly useful for talk-format and news programming.

Action Buttons

- **SAVE** — Saves the configured cutoff frequency to the active profile.
- **BYPASS** — Bypasses the LPF stage for A/B comparison.
- **BACK** — Returns to the Tune screen without saving.

5.3.4 Bass enhancement

The Bass Enhancement screen (Figure 5.3.5) provides access to the dual-band bass enhancement system, which adds controlled low-frequency reinforcement at two independently configurable bands.

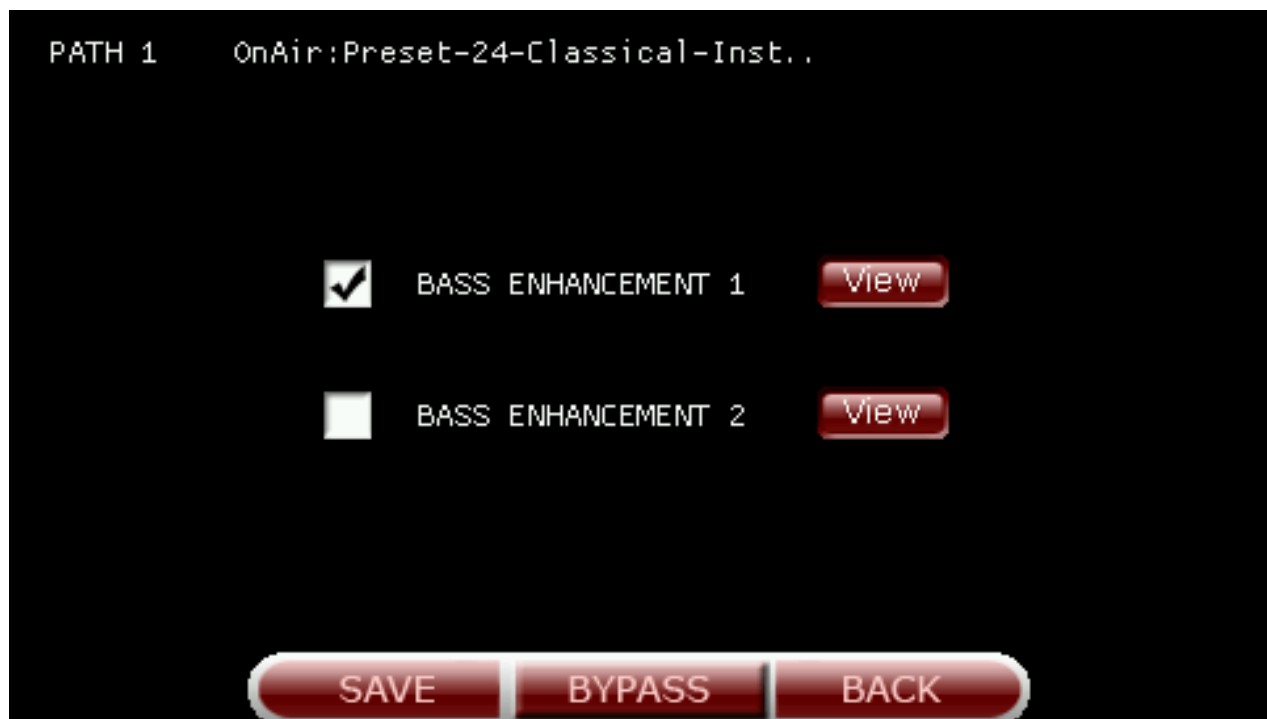


Figure 5.3.5: Bass Enhancement Screen

The screen header shows the active path and current on-air profile (e.g., "PATH 1 — OnAir: Preset-24-Classical-Inst..").

Controls

- **BASS ENHANCEMENT 1** — Checkbox to enable or disable the first bass enhancement band. Press **VIEW** to open the parameter screen for Band 1 (center frequency, gain, and p.Level).
- **BASS ENHANCEMENT 2** — Checkbox to enable or disable the second bass enhancement band. Press **VIEW** to open the parameter screen for Band 2 (center frequency, gain, and p.Level).

The two bands are typically configured at different center frequencies (e.g., sub-bass and mid-bass) to provide targeted low-frequency reinforcement without muddiness.

Action Buttons

- **SAVE** — Saves the configured cutoff frequency to the active profile.
- **BYPASS** — Bypasses the LPF stage for A/B comparison.
- **BACK** — Returns to the Tune screen without saving.

5.3.5 Wideband auto noise reduction

The WANR screen (Figure 5.3.6) configures the broadband noise reduction system, which detects and suppresses background noise while preserving program audio clarity.

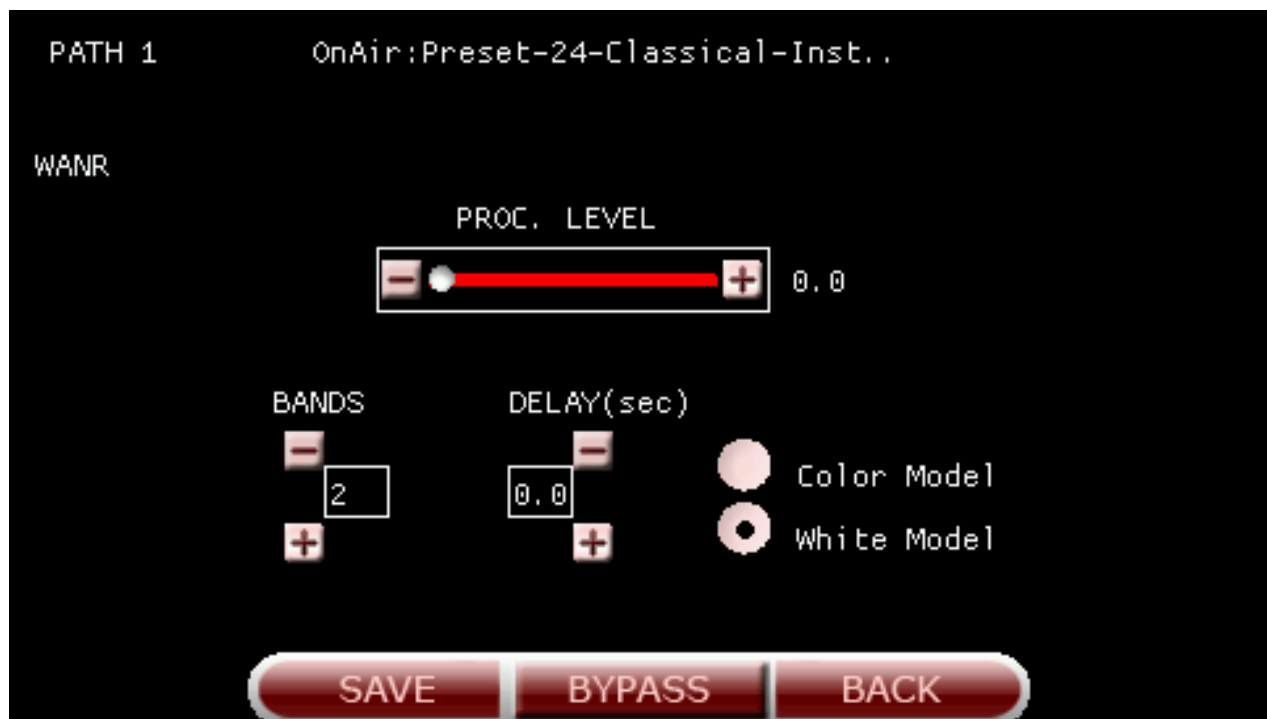


Figure 5.3.6: Wideband Auto Noise Reduction Screen

The screen header shows the active path and current on-air profile (e.g., "PATH 1 — OnAir: Preset-24-Classical-Inst..").

Controls

- **PROC. LEVEL** — Sets the noise reduction processing intensity. Adjustable using the slider with – and + buttons. The current value is displayed to the right (e.g., 0.0). Higher values apply more aggressive noise reduction.
- **BANDS** — Sets the number of analysis bands used by the noise reduction algorithm. Adjustable using the – and + buttons (e.g., 2).
- **DELAY (sec)** — Sets the processing delay in seconds. Adjustable using the – and + buttons (e.g., 0.0).

- **Noise Model** — Selects the noise model used for analysis:
- **Color Model** — Optimized for colored noise (non-uniform spectral content such as hum, rumble, or HVAC noise).
- **White Model** — Optimized for white noise (uniform spectral content such as tape hiss or microphone self-noise).

Action Buttons

- **SAVE** — Saves the configured cutoff frequency to the active profile.
- **BYPASS** — Bypasses the LPF stage for A/B comparison.
- **BACK** — Returns to the Tune screen without saving.

5.3.6 Noise gate

The Noise Gate screen (Figure 5.3.7) configures the noise gating functions, providing both wideband noise gating and per-band noise gating control.

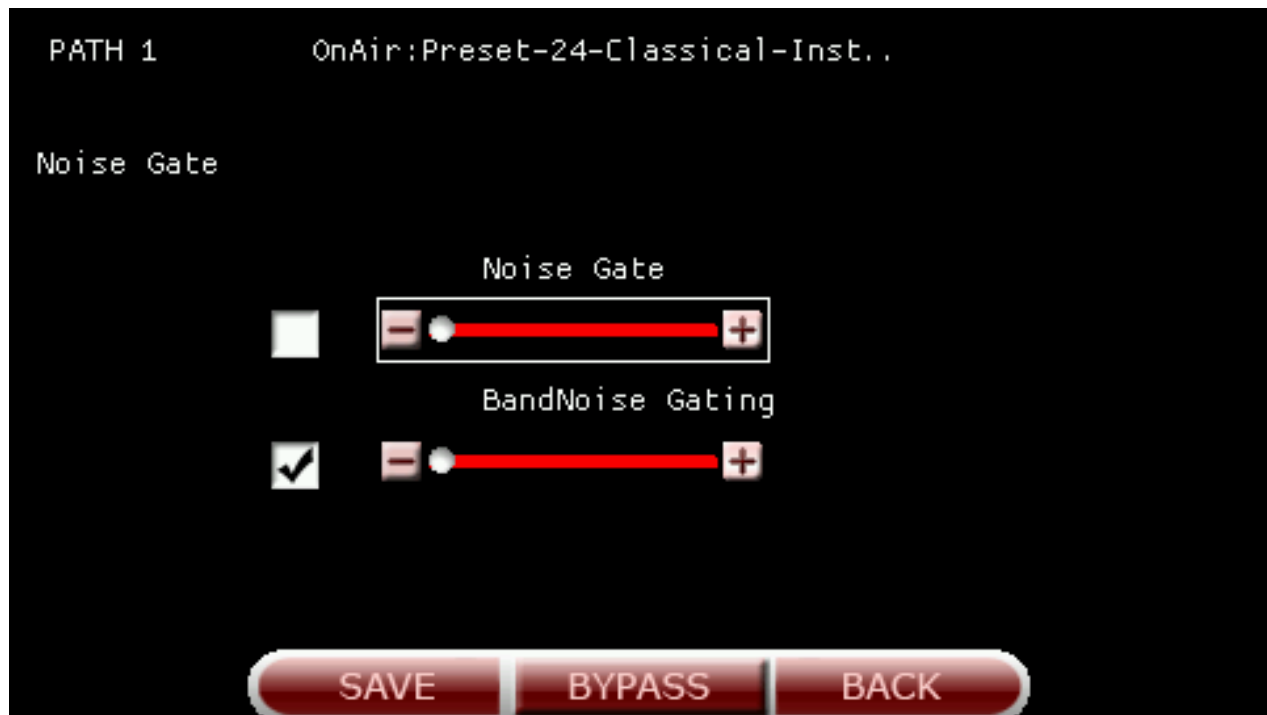


Figure 5.3.7: Noise Gate Screen

The screen header shows the active path and current on-air profile (e.g., "PATH 1 — OnAir: Preset-24-Classical-Inst..").

Controls

- **Noise Gate** — Checkbox to enable or disable the wideband noise gate. The slider with – and + buttons sets the 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.

- **BandNoise Gating** — Checkbox to enable or disable per-band noise gating. The slider with – and + buttons sets the gating intensity applied independently across the 24 frequency bands, allowing frequency-selective noise floor management.

Action Buttons

- **SAVE** — Saves the configured cutoff frequency to the active profile.
- **BYPASS** — Bypasses the LPF stage for A/B comparison.
- **BACK** — Returns to the Tune screen without saving.

5.3.7 Initial limiter

The Initial Limiter screen (Figure 5.3.8) sets the attenuation level applied by the limiter before final limiting, which provides initial peak control before the audio enters the main SmoothClip™ look ahead limiting processing engine.

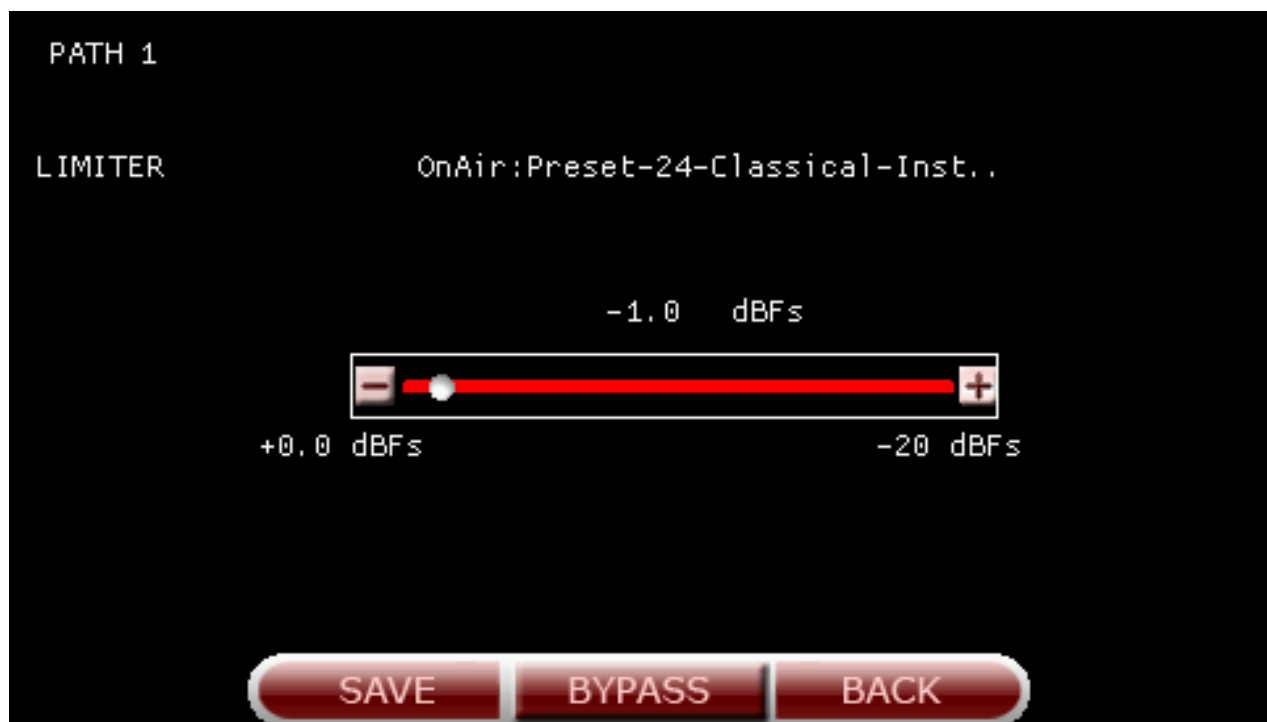


Figure 5.3.8: Initial Limiter Screen

The screen header shows the active path and current on-air profile (e.g., "PATH 1 — OnAir: Preset-24-Classical-Inst..").

Controls

- **LIMITER** — Sets the limiter attenuation in dBFs. Adjustable from **+0.0 dBFs** to **-20 dBFs** using the slider with – and + buttons. The current value is displayed above the slider (e.g., -1.0 dBFs).

Action Buttons

- **SAVE** — Saves the configured cutoff frequency to the active profile.
- **BYPASS** — Bypasses the LPF stage for A/B comparison.
- **BACK** — Returns to the Tune screen without saving.

5.3.8 Two-band limiter

The Two Band Limiter screen (Figure 5.3.9) sets the limiter attenuation level applied by the two-band limiter stage.

The screen header shows the active path and current on-air profile (e.g., "PATH 1 — OnAir: Preset-24-Classical-Inst..").

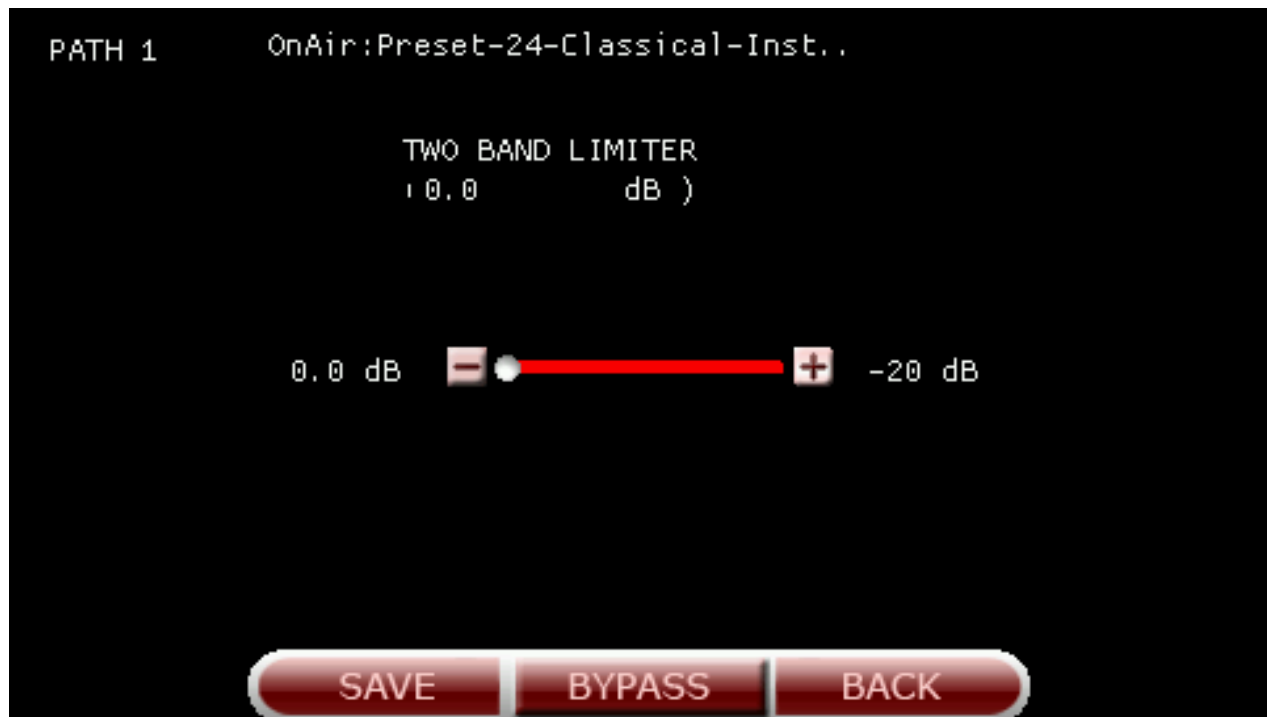


Figure 5.3.9: Two Band Limiter Screen

Controls

- **TWO BAND LIMITER** — Sets the limiter attenuation in dB. Adjustable from **0.0 dB** to **-20 dB** using the slider with – and + buttons. The current value is displayed at the top of the screen.

Action Buttons

- **SAVE** — Saves the configured value to the active profile.
- **BYPASS** — Bypasses the Two Band Limiter stage for A/B comparison.

- **BACK** — Returns to the Tune screen without saving.

5.3.9 Dynamic range control

This parameter is not configurable via the TFT screen. It is controlled through the SoundMax Station Manager. Refer to Section 5.2, SoundMax Tuning Controls: Tuning Guide for Individual Audio Processing Modules, for details

5.3.10 Graphic equalizer

The Equalizer screen (Figure 5.3.10) provides a 10-band graphic equalizer with an additional pre-gain control, allowing the user to apply spectral shaping to the processed audio.

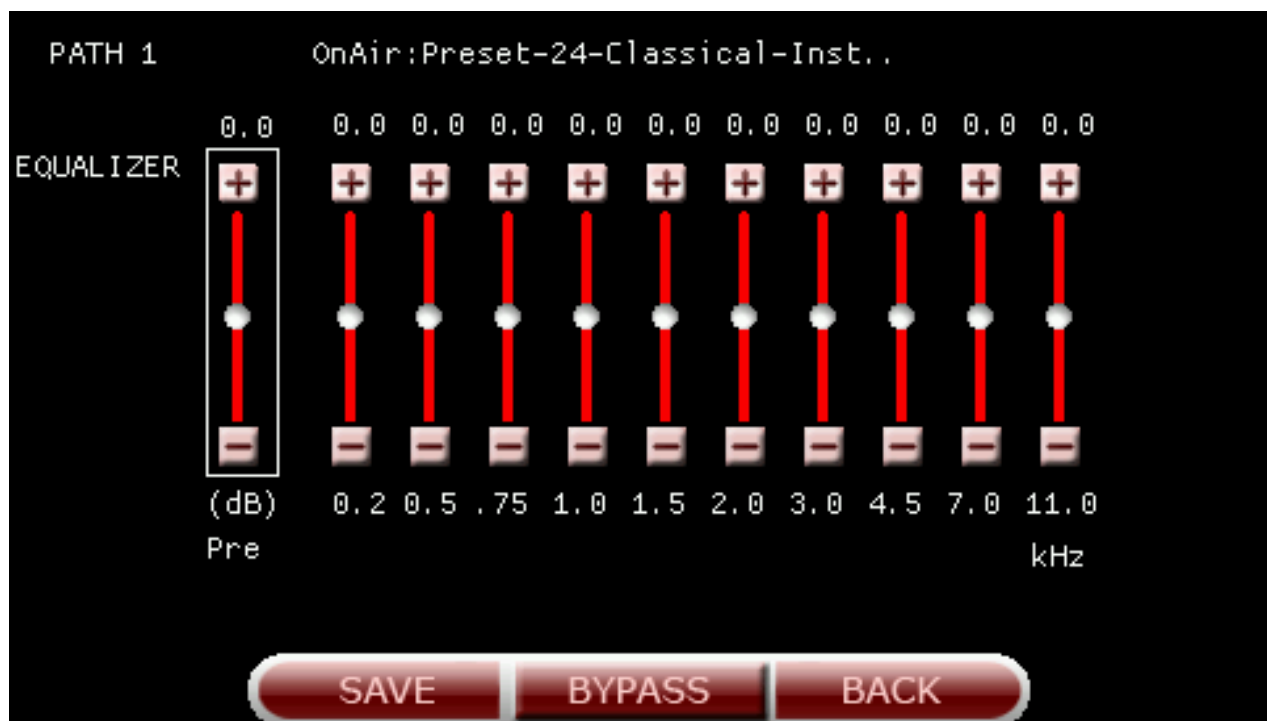


Figure 5.3.10: Equalizer Screen

The screen header shows the active path and current on-air profile (e.g., "PATH 1 — OnAir: Preset-24-Classical-Inst..").

Band Frequency

- 0.2 kHz 0.5 kHz 0.75 kHz 1.0 kHz 1.5 kHz 2.0 kHz 3.0 kHz 4.5 kHz 7.0 kHz 11.0 kHz.

Controls

- **Pre (dB)** — Pre-gain fader applied to the EQ input stage. Adjustable using the + and – buttons. The current value is displayed above the fader (e.g., 0.0 dB).

- **Band Faders** — Ten frequency band faders, each with + and – buttons for boost or cut. The current gain value (in dB) is shown above each fader. The center frequency of each band is shown below in kHz:

Each band can be boosted or cut independently to shape the overall frequency response.

Action Buttons

- **SAVE** — Saves the configured cutoff frequency to the active profile.
- **BYPASS** — Bypasses the LPF stage for A/B comparison.
- **BACK** — Returns to the Tune screen without saving.

5.3.11 Stereo processing

The Stereo Processing screen (Figure 5.3.11) configures the stereo width control and mono downmix options for the active profile.

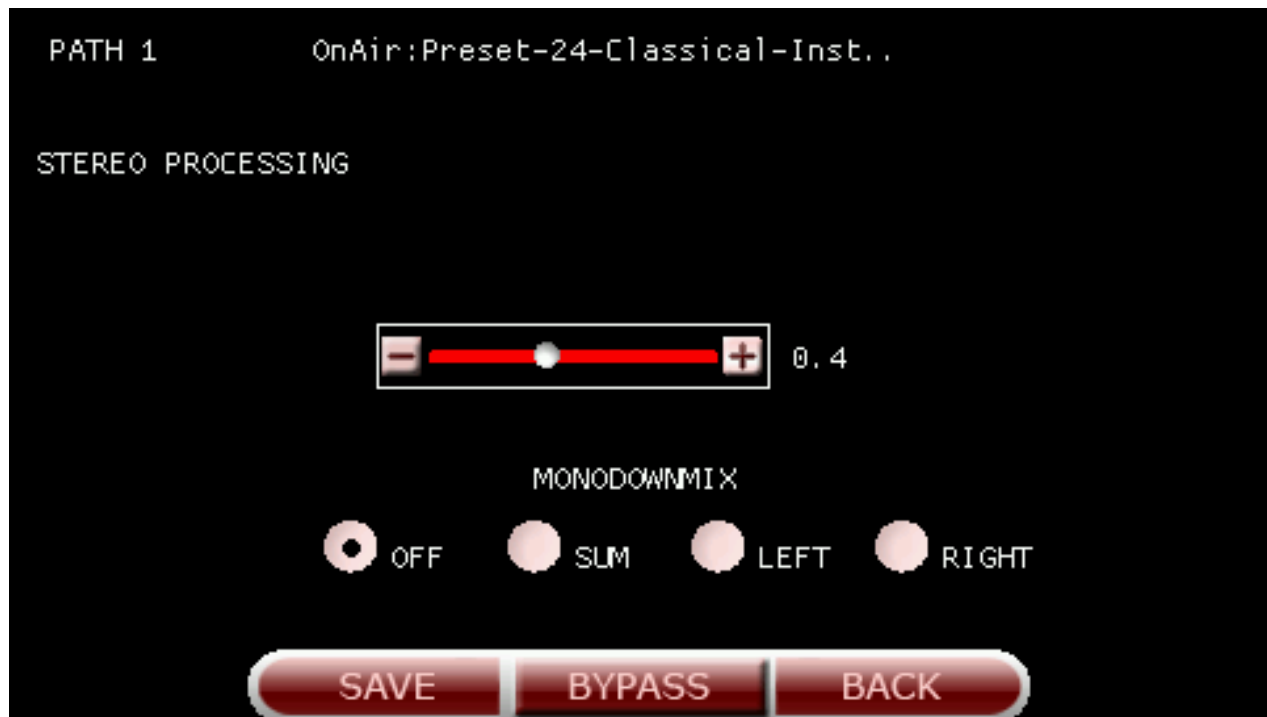


Figure 5.3.11: Stereo Processing Screen

The screen header shows the active path and current on-air profile (e.g., "PATH 1 — OnAir: Preset-24-Classical-Inst..").

Controls

- **STEREO PROCESSING** — Sets the stereo width / processing intensity. Adjustable using the slider with – and + buttons. The current value is displayed to the right (e.g., 0.4). Higher values increase stereo image width and spatial enhancement.

- **MONODOWNMIX** — Selects the mono downmix mode for stereo-to-mono conversion:
 - **OFF** — No downmix; stereo signal passes unchanged.
 - **SUM** — Combines Left and Right channels into a mono sum (L+R).
 - **LEFT** — Outputs only the Left channel as mono.
 - **RIGHT** — Outputs only the Right channel as mono.

Action Buttons

- **SAVE** — Saves the configured cutoff frequency to the active profile.
- **BYPASS** — Bypasses the LPF stage for A/B comparison.
- **BACK** — Returns to the Tune screen without saving.

5.3.12 Dynamic listening fatigue reduction

The DLFR screen (Figure 5.3.12) configures the proprietary Dynamic Listening Fatigue Reduction processor, which reduces long-term listener fatigue by intelligently managing audio dynamics across multiple frequency bands.

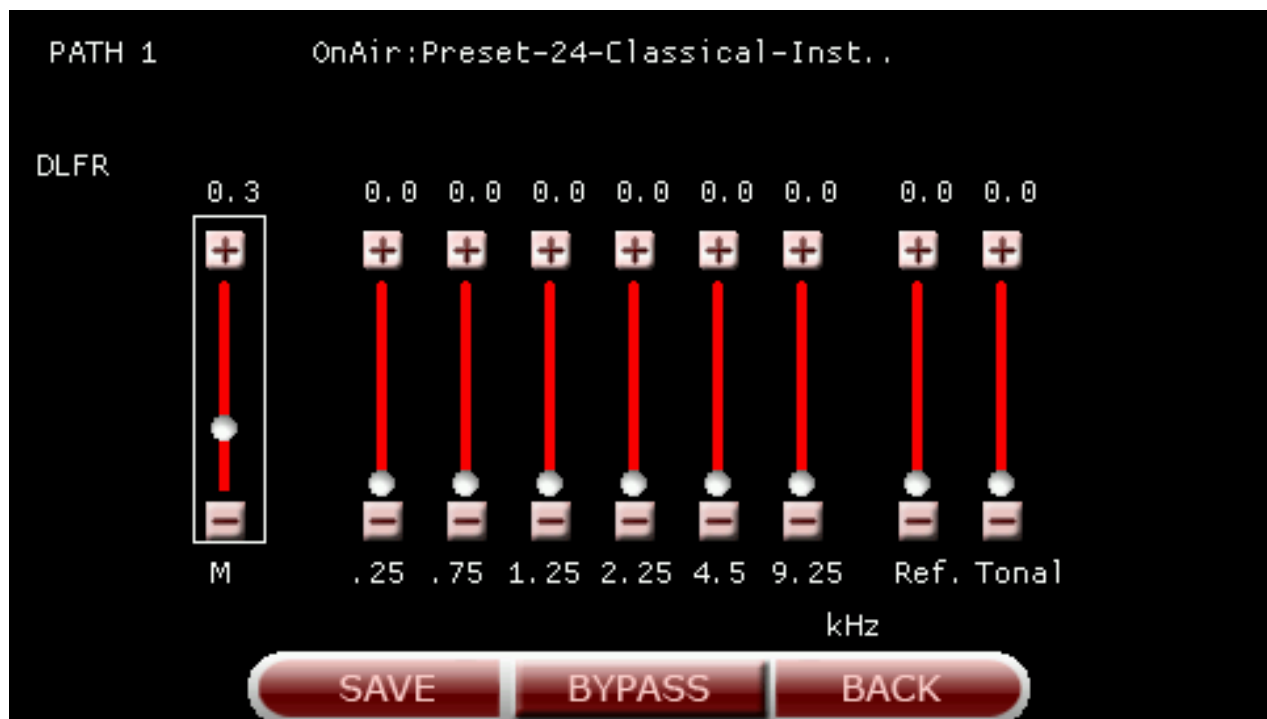


Figure 5.3.12: Dynamic Listening Fatigue Reduction Screen

The screen header shows the active path and current on-air profile (e.g., "PATH 1 — OnAir: Preset-24-Classical-Inst..").

Controls

- **M (Master)** — Master DLFR intensity fader. Adjustable using the + and – buttons. The current value is displayed above the fader (e.g., 0.3). Sets the overall DLFR processing depth applied to the audio.
- **Band Faders** — Six frequency band faders provide per-band DLFR adjustment, plus two additional control faders. Each fader has + and – buttons; the current value is shown above each. The frequency or function of each fader is shown below:

Action Buttons

- **SAVE** — Saves the configured cutoff frequency to the active profile.
- **BYPASS** — Bypasses the LPF stage for A/B comparison.
- **BACK** — Returns to the Tune screen without saving.

5.3.13 High-frequency enhancement

The High Frequency Enhancement screen (Figure 5.3.13) sets the intensity of the HFE processor, which restores high-frequency detail and adds presence and brilliance to the processed audio.

The screen header shows the active path and current on-air profile (e.g., "PATH 1 — OnAir: Preset-24-Classical-Inst..").

Controls

- **HFE LEVEL** — Checkbox to enable or disable High Frequency Enhancement. The slider with – and + buttons sets the enhancement intensity. Higher values add more high-frequency detail and air to the processed audio.

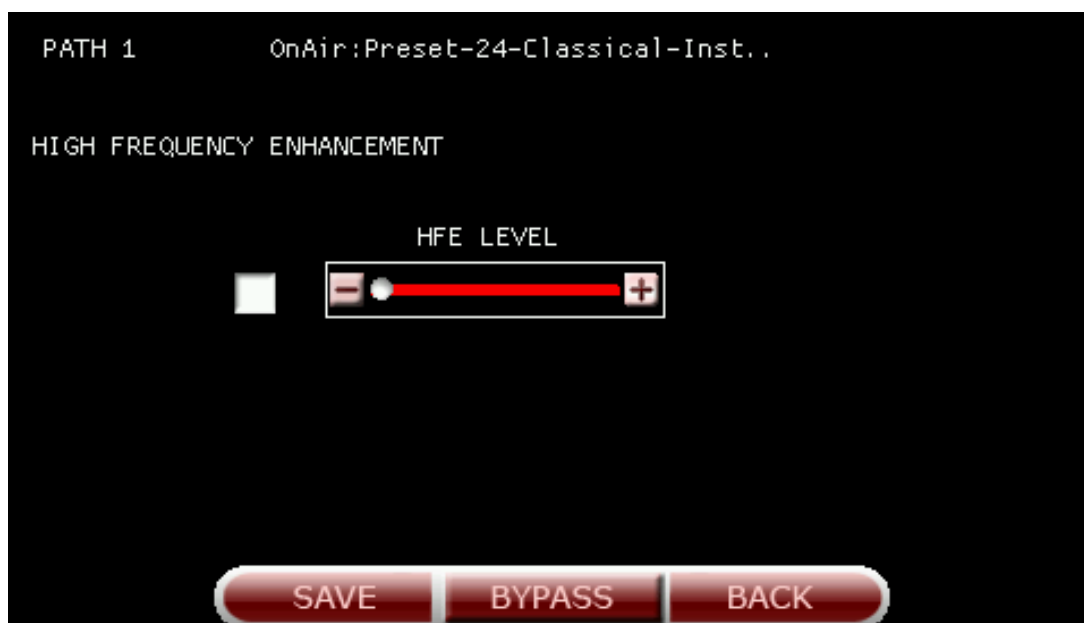


Figure 5.3.13: High Frequency Enhancement Screen

Action Buttons

- **SAVE** — Saves the configured cutoff frequency to the active profile.
- **BYPASS** — Bypasses the LPF stage for A/B comparison.
- **BACK** — Returns to the Tune screen without saving.

5.3.14 Pre-emphasis filter

The Pre-Emphasis screen (Figure 5.3.14) selects the pre-emphasis filter applied to the output signal, as required for FM broadcast transmission compliance.

The screen header shows the active path and current on-air profile (e.g., "PATH 1 — OnAir: Preset-24-Classical-Inst..").

The current pre-emphasis state is displayed at the top of the screen (e.g., "PRE-EMPHASIS (OFF)").

Controls

Three radio button options are available for selection:

- **OFF** — No pre-emphasis applied. Used for digital broadcast paths or when pre-emphasis is applied externally.
- **50 μ s** — Standard pre-emphasis time constant used in Europe, Asia, Australia, and many other parts of the world for FM broadcasting.
- **75 μ s** — Standard pre-emphasis time constant used in North America (USA, Canada) and South Korea for FM broadcasting.

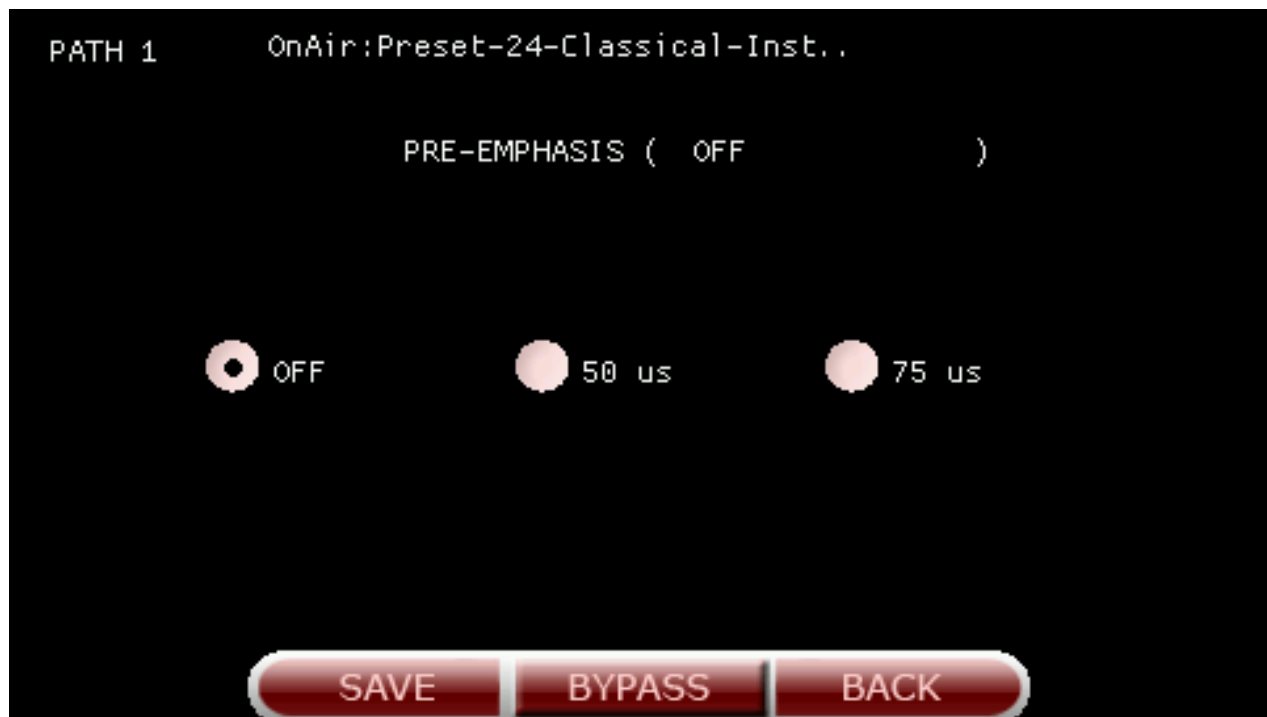


Figure 5.3.14: Pre-Emphasis Screen

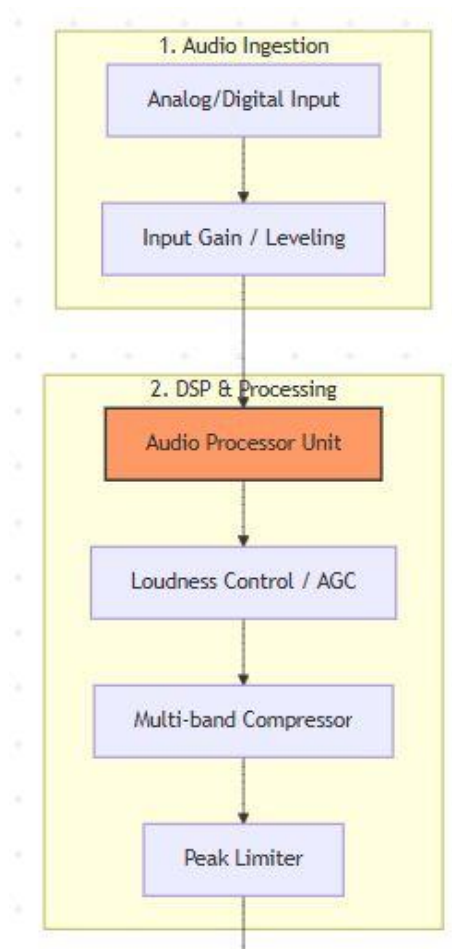
Only one option can be selected at a time. The selected value must match the corresponding de-emphasis setting at the receiver end.

Action Buttons

- **SAVE** — Saves the configured cutoff frequency to the active profile.
- **BYPASS** — Bypasses the LPF stage for A/B comparison.
- **BACK** — Returns to the Tune screen without saving.

5.4 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 Q24-6111 and Q24-6111HD 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 5.4.1 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.



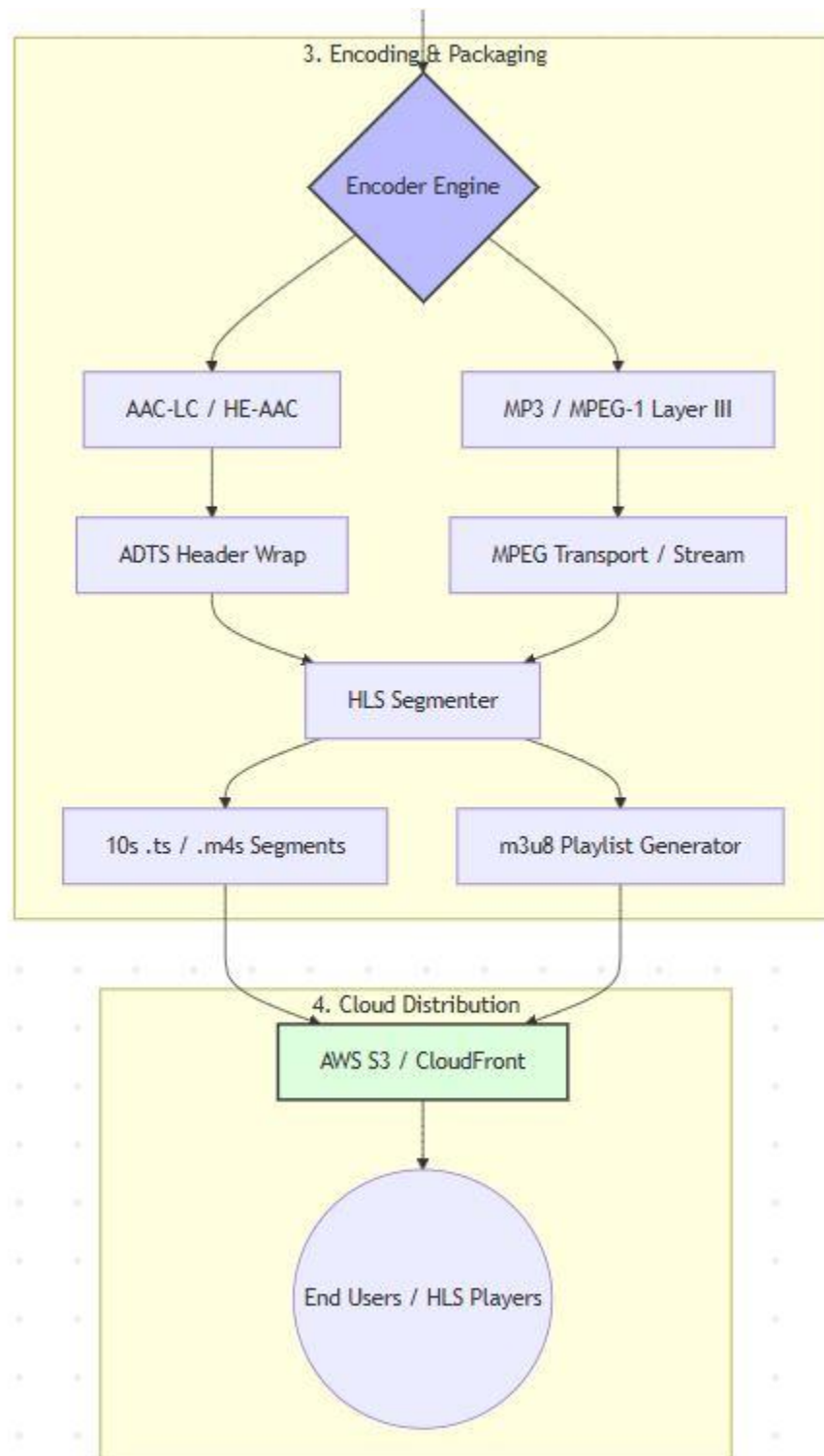


Figure 5.4.1: Internet Streaming Signal Flow in Perceptual SoundMax™

5.4.1 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 the 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 audio coding, **Spectral Band Replication (SBR)** is also 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 analyzes 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 the product uses HLS, the AAC bitstream must be "packetized" so it can be decoded starting from any 10-second segment. We therefore use the **ADTS (Audio Data Transport Stream)** encapsulation 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.

5.4.2 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 analyzes 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 Q246111 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 MP3 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" and preferably needs TS encapsulation for HLS streaming.

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.

5.4.3 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 5.4.2 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).

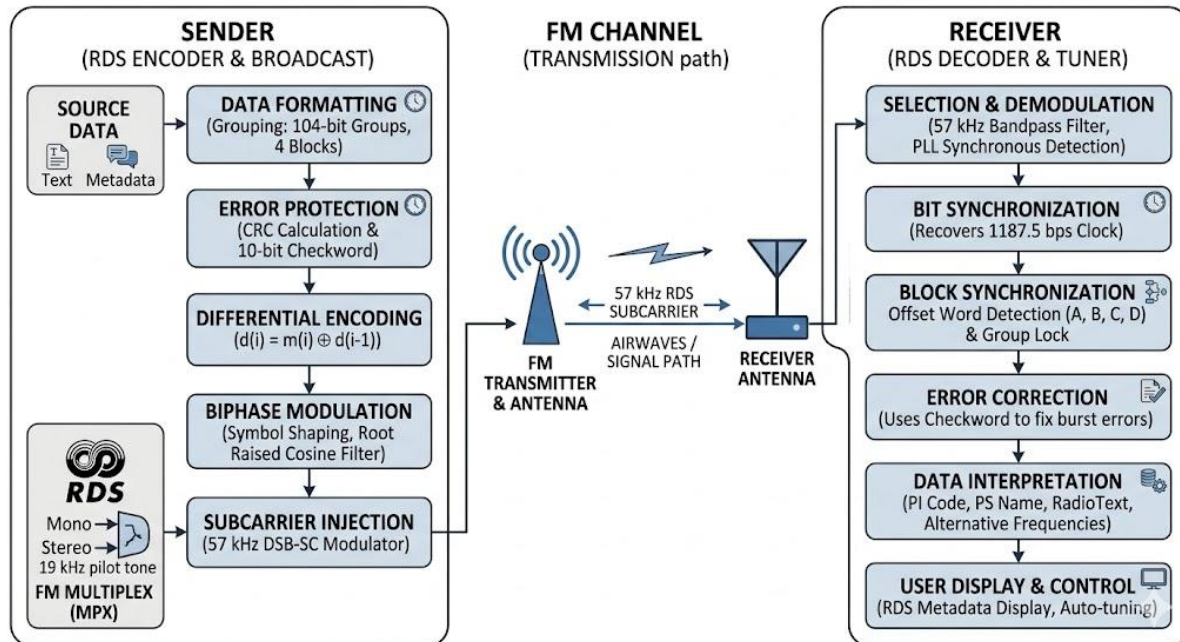


Figure 5.4.2: RDS Insertion Overview

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 checkword 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 Q24-6111 and Q24-6111HD 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.

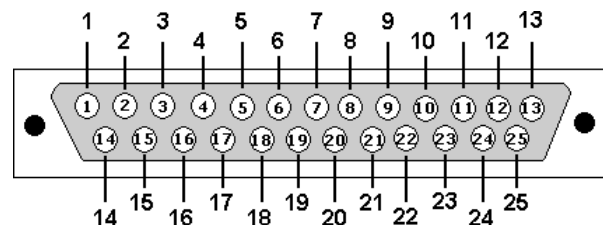


Figure 6.1: Connector Pin Configuration

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

GPIO Input Configuration

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

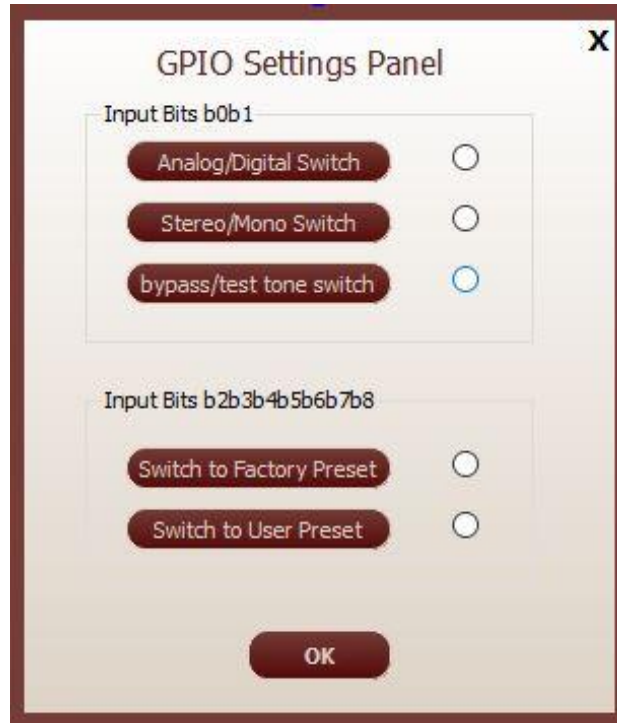


Figure 6.2: GPIO Settings Panel

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 Specifications

MAIN Analog Audio Input:

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

MAIN Analog Audio Output:

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

MAIN Digital Audio Input (AES/ EBU):

- **Configuration:** Stereo
- **Sampling Rate:** 32kHz-192 kHz (Auto Sense, SRC)
- **Word length:** 16 or 24 bits
- **Reference level:** -25 to 0 dBFS
- **EMI Suppressed:** Yes
- **Connector:** AES– XLR Female, 110 Ω

MAIN Digital Audio Output (AES/EBU):

- **Configuration:** Stereo
- **Output level:** -20 to 0.00 dBFS
- **Frequency Response:** ± 0.3 dB (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:

(AES2 on Model Q24-6111HD, OUT2 on Model Q24-6111)

- **Configuration:** Stereo
- **Output level:** -20 to 0.00 dBFS
- **Frequency Response:** $\pm 0.3\text{dB}$ (5 Hz to 20 kHz)
- **Sampling Rate:** Same as MAIN Digital Output
- **Connector:** AES – XLR Male
- **Diversity Delay:** Up to 12 seconds delay in MAIN (software configurable with high precision). Accurate diversity delay ± 2 samples

AUX1 - Auxiliary Digital Audio Input - HD2/DRM-Service B Input:

(Model Q24-6111HD Only)

- **Configuration:** Stereo
- **Sampling Rate:** 44.1 or 48 kHz (Auto Sense)
- **Word length:** 16 or 24 bits
- **Connector:** AES– XLR Female, 110Ω

AUX1 - Auxiliary Digital Audio Output - HD2/DRM-Service B Output:

(Model Q24-6111HD Only)

- **Configuration:** Stereo
- **Output level:** -20 to 0.00 dBFS
- **Frequency Response:** $\pm 0.3\text{dB}$ (5 Hz to 20 kHz)
- **Sampling Rate:** Follows AUX1 Input
- **Connector:** AES – XLR Male

AUX2 - Auxiliary Digital Audio Input – HD3/DRM-Service C Input:

(Model Q24-6111HD Only)

- **Configuration:** Stereo
- **Sampling Rate:** 44.1 or 48 kHz (Auto Sense)
- **Word length:** 16 or 24 bits
- **Connector:** AES– XLR Female, 110Ω

AUX2 - Auxiliary Digital Audio Output – HD3/DRM-Service C Output:

(Model Q24-6111HD Only)

- **Configuration:** Stereo
- **Output level:** -20 to 0.00 dBFS
- **Frequency Response:** $\pm 0.3\text{dB}$ (5 Hz to 20 kHz)
- **Sampling Rate:** Follows AUX2 Input

- **Connector:** AES – XLR Male

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)

2 X Analog Composite Baseband Outputs:

- **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:** ≥ 90 dB, (20Hz – 15kHz)
- **THD:** $\leq 0.02\%$, (20Hz – 15kHz)
- **Stereo Separation:** ≥ 60 dB (20Hz – 15kHz)
- **Cross Talk:** ≤ -90 dB, channel to channel
- **38kHz Suppression:** ≥ 90 dB
- **Pilot Protection:** 65 dB relative to 9% injection, +/- 500 Hz
- **Pilot Stability:** 19 kHz +/- 1.0 Hz
- **Pilot Level:** Adjustable from 6% to 12%
- **57 kHz (RDS) Protection:** > 60 dB relative to 4% injection, +/- 2 kHz
- **Connectors:** Two Coaxial BNC, EMI suppressed

2 X Analog Composite Baseband Inputs:

- **Configuration:** Two inputs with Individual software configuration (Pilot/RDS/SCA)
- **Connectors:** Two Coaxial BNC, EMI suppressed

Digital Composite/MPX Output:

- **Configuration:** Multiplexed with MAIN Digital Output (software configurable)
- **Sample Rate:** 192 kHz

- **Connectors:** AES – XLR Male

Streaming Audio Output (up to 3 streams):

- **Configuration:** 1 x Internet Stream Out for MAIN/HD1 Audio
1 x Internet Stream Out for AUX1/HD2 Audio (Model Q24-6111HD)
1 x Internet Stream Out for AUX2/HD3 Audio (Model Q24-6111HD)
- **Encoding formats:** Enhanced AAC Profiles: LC, HE-v1, HE-v2; MP3
- **Encapsulation:** ADTS or MPEG2-TS for AAC; MP3 or MPEG2-TS for MP3
- **Bit rates:** 24-384 kbps for AAC; 32-320 kbps for MP3
- **HLS Generation:** Up to 2 bit rates for adaptive HLS; Chunking 3-10 sec (soft-configurable)
- **Distributions:** CDN Primary/Secondary Push/Post
External Icecast Servers
In-built HTTP server for direct pull of AAC/MP3/HLS
UDP Unicast/Multicast
- **Player compatibility:** Android/iOS Native, VLC, All Browsers with HLS/AAC/MP3 support
- **Browser compatibility:** Edge, Chrome, Safari, Mozilla
- **Connector:** Ethernet-2 (see below)

Ethernet-2 (CONTROL/STREAM) for Remote PC Control & Stream Out:

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

Management Control:

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

USB Connection for Upgrades:

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

GPIO Interface (Optional):

- **GPIO:** 8 inputs and 2 outputs opto-isolated and floating. User selectable configuration.
- **Control:** user- programmable 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.

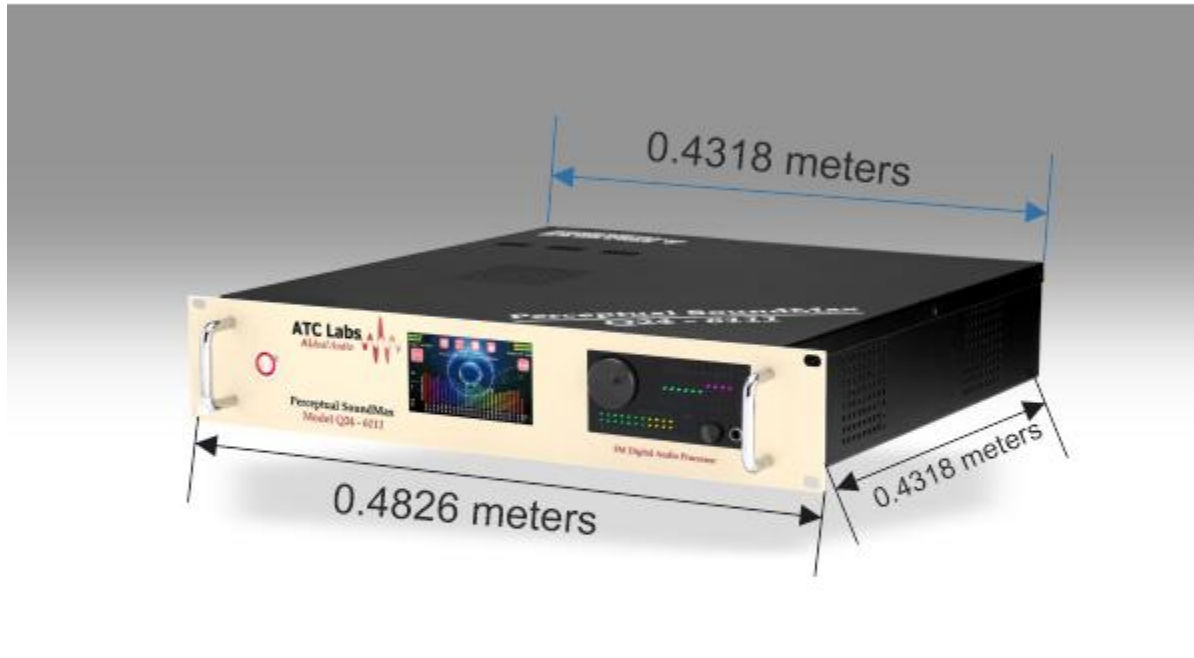
Weight: 5.5 Kg.

Dimensions:

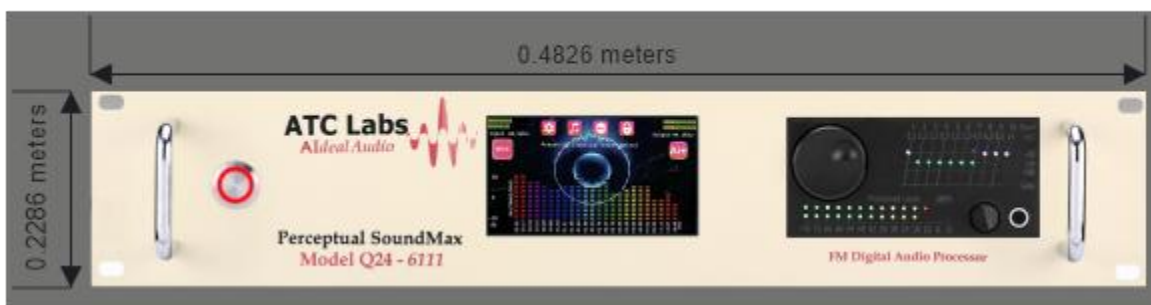
- **Width:** 0.483 m (19")
- **Height:** 0.088m (2.0U/3.5in)
- **Depth:** 0.4295m (16.9")

Section 8 Dimensioned Product View

Isometric View with Dimensions (meters)



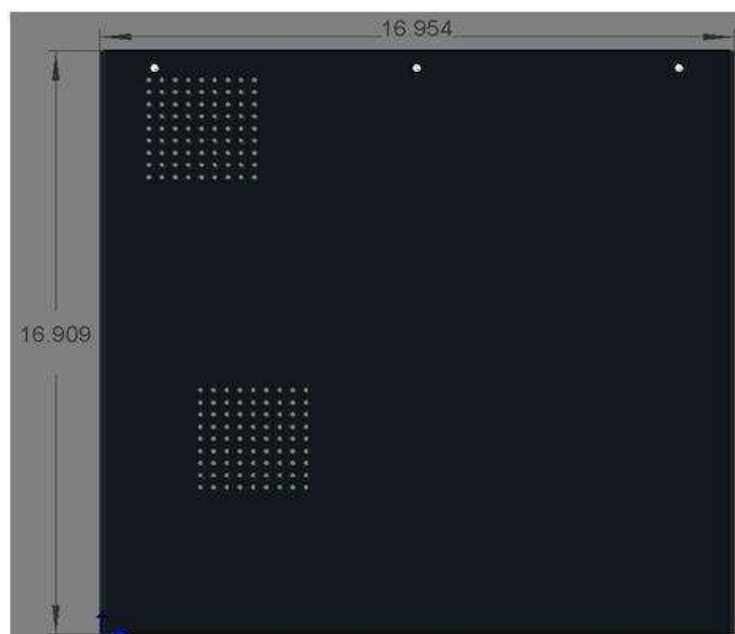
Front Panel View with Dimensions (in meters & inches)



Back Panel View of Model Q24-6111 (top image) and Q24-6111HD (bottom image) with Dimension



Top Cover View with Dimensions (inches)



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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Email: support@atc-labs.com

Website: <https://www.atc-labs.com>

Online Resources

- [Perceptual SoundMax™ Product Page](#)
- [SoundMaxManager™ Download Page](#)
- [Sales Enquiry](#)
- [Support Enquiry](#)

Perceptual SoundMax™ Model Q41-6111 and Model Q24-6111HD User Manual Version 8.0

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