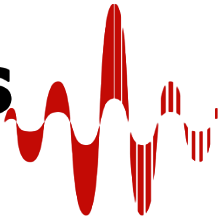


ATC Labs
Aideal Audio



Perceptual SoundMax™

Models Q24-4888 & Q24-4880

Multi-Functionality 8 Channel Audio De-Multiplexer, Decoder,
Broadcast Processor, Encoder, Multiplexer, and Streamer

*AI-Driven 24-band audio processing, high-quality encoding & Streaming and
Immersive Audio*

Operation Manual

Version 14.2, September 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

The Perceptual SoundMax™ Model Q24-4888 and Model Q24-4880 are multifunctionality audio processing equipment for 1-to-8 independent stereo channel processing. Along with audio processing these integrate highest grade stream de-multiplexing, decoding, encoding, streaming and multiplexing capabilities. These are designed for professional broadcasters, studios, and *large-scale* streamers who require reliable multi-channel hardware for their audio contribution, distribution, or streaming application. These are designed for continuous 24×7 operation in demanding broadcast environments.

There are two primary differences between Model Q24-4880 and Model Q24-4888, with the higher end Model Q24-4888 offering these 2 additional hardware capabilities: (1) Analog input (Model Q24-4880 is digital input only), (2) Hardware ASI input/output for MPEG2 Transport Stream (TS).

Multi-Function Integration with multiple Application Scenarios:

The Perceptual SoundMax™ Q24-488X series consolidates several broadcast processing, encoding, decoding, and streaming technologies into a single hardware platform, reducing cost, rack space, and power consumption. It may be used in any of the following *application scenarios*:

1. **High-Quality Streaming with audio processing of 1-to-8 audio feeds.** In this application scenario audio input may be received from a digital AES feed, or Analog L/R stereo feed (Model Q24-4888 only), or an Internet HTTP/UDP Stream, or a Multiplexed TS stream received via LAN, or a multiplexed (Transport Stream) TS stream received via a hardware ASI input on a coax connector (hardware ASI input in Model Q24-4888 only). High-quality stream encoder along with scalable multi-rate HLS chunk encoding and integrated distribution server and CDN push/pull functionalities are supported.
2. **Stream Decoding/De-Multiplexing:** In this application scenario the equipment may receive audio from independent HTTP/UDP Streams for each of the up-to-8 channels, perform standard compliant decoding, and playout decoded audio via the available digital AES independent outputs. Optionally audio input may be obtained by demultiplexing and input TS stream received via LAN, or a multiplexed (Transport Stream) TS stream received via a hardware ASI input on a coax connector (hardware ASI input in Model Q24-4888 only).
3. **Multiplexing of 1-8 processed and encoded audio feeds into a TS stream and distribution via Network or a Hardware ASI output connector** (hardware ASI connector in Model Q24-4888 only). In this application scenario also audio input may be received from a digital AES feed, or Analog L/R stereo feed (Model Q24-4888 only), or an Internet HTTP/UDP Stream, or a Multiplexed TS stream received via LAN, or a multiplexed (Transport Stream) TS stream received via a hardware ASI input on a coax connector (hardware ASI input in Model Q24-4888 only).
4. **Multiplexing of 1-8 processed and encoded audio feeds into a TS stream with embedding of additional channels from a de-multiplexed input transport stream (TS).** This scenario has all the functionality of the application scenario 3 above with the *additional functionality* that one or more of the Packetized Elementary Streams (PES) from an *input* TS maybe de-multiplexed and re-

multiplexed into the output TS. The input TS may be received as a broadcast/unicast stream through the Ethernet interface or in Model Q24-4888 via a hardware ASI input.

5. **Audio Processing for Video: Processing of one or more Audio Elementary Streams in a Multiplexed Video TS Stream.** In this application scenario the input video TS is demultiplexed and up to 8 audio streams may then be processed for audio level and quality and re-encoded and re-multiplexed **as an output TS stream**. The input TS may be received as a broadcast/unicast stream through the Ethernet interface or in Model Q24-4888 via a hardware ASI input. The output TS distribution may be performed via Network or via a hardware ASI output connector (hardware ASI connector in Model Q24-4888 only).
6. **General Purposed audio processor for 1-to-8 independent stereo channels.**

These models Q24-4888 and Q24-4880 incorporate several key enabling technologies which ensure highest performance in all the application scenarios above.

Key Enabling 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.
- **Standard Compliant Decoders:** fully standard complaint HE/LC-AAC (V/V2) and MP3 decoders are integrated into the products.
- **Low Latency MPEG2-Transport Stream (TS) multiplexers and de-multiplexers**
- **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.

Built on advanced ATC Labs signal-processing architecture, the Models Q24-4888 and Q24-4880 integrates sophisticated 24-band audio processing using an extremely detailed and accurate time-frequency representation of the audio signal. By combining modern filter-bank technologies, psychoacoustic modelling, harmonic analysis, and intelligent signal-conditioning techniques developed by ATC Labs, the system delivers enhanced loudness, clarity, spectral balance, and long-term listening comfort while maintaining codec efficiency and transparency.

The result is a detailed, powerful, and highly optimized broadcast sound that preserves natural tonal balance while improving perceived loudness and intelligibility. Broadcasters can achieve a clean, punchy, and consistent on-air sound without introducing listener fatigue, even at lower transmission bitrates.

The Perceptual SoundMax™ Models Q24-4888 and Q24-4880 feature an intuitive front-panel touch-screen interface for direct local operation and is complemented by advanced PC-based remote-control software that provides comprehensive monitoring, configuration, routing, streaming, and processing control over a network. The system includes multiple factory presets optimized for different applications while allowing users to create and store custom profiles as required.

Built using premium-grade broadcast components, the Perceptual SoundMax™ Q24-4888/Q24-4880 is engineered for uninterrupted 24×7 operation with high reliability, stability, and minimal maintenance requirements. The platform is optimized for contribution links, studio-transmitter links, internet streaming, satellite distribution, and multi-channel audio networking applications.

Technical Specifications for Models Q24-4880 and Q24-4888 - Highlights

Feature	Details
Digital I/O	8 x AES Digital I/O on XLR or TS connectors
Analog Input (Model Q24-4888 only)	8 x Balanced or Unbalanced L/R Stereo Input – 2 Configurations available (see section 2.3)
IP based Stream Input	HTTP/UDP Input. Independent configuration 1-to8 streams. Standard complaint MP3 and LC/HE-AAC Audio Decoding
TS Multiplexer/De-Multiplexer	1. IP based TS receive/transmit; multicast/unicast. 2. Physical Multiplexed ASI Input and Output connectors (Model Q24-4888 only)
Encoding/Decoding	HE/LC-AAC V1/V2: 32-256kbps, MP3, Scalable HLS, AAC+ with ADTS Encapsulation.
Streaming, AOIP, and Networking	Configurable Audio-Over-IP (AOIP) input/output – multicast/unicast. Distribution server & CDN connectivity Dual Ethernet (RJ-45): Control/Stream and AoIP
Compliance	Nielsen™ Watermarking
Processing	24-band high resolution Aideal Audio™ SoundMax Processing
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 Q24-4888 and Q24-4880, unless noted otherwise.

Section 2 Installation And Quick Start Guide

2.1 Supplied Accessories

Package Contents

Your Perceptual SoundMax™ Model Q24-4888/4880 shipment includes the following items:

1. Perceptual SoundMax™ Audio Processor cum Encoder/Decoder/Streamer/De-Multiplexer/Multiplexer (Model Q24-4888/Q24-4880) hardware unit
2. Power cord
3. Rack-mounting kit with 10-32 x ½" screws and washers
4. Ethernet cable
5. SoundMax™ Manager software installation CD-ROM
6. Operation manual
7. Warranty certificate
8. Test report

Important:

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

Save all original packing materials. Should you need to ship the Q24-4888 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 Q24-4888/Q24-4880 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 Panels for both Models Q24-4888/Q24-4880.

Model Q24-4888 is offered within 2 configurations as Listed below. **Configuration 1** with *digital-AES and unbalanced analog L/R inputs* and *digital-AES outputs* is attractive for general streaming, decoding, and processing applications where physical input and output connectors are both needed. **Configuration 2** is meant for *streaming or multiplexing output only applications*, where physical audio output connectors are not needed, so those are eliminated, instead the physical connectors are enhanced for the inputs such that *both balanced L/R stereo analog inputs and digital AES inputs* are supported.

Configuration 1: Input (Lower row TRS/XLR Combo connectors) – Digital AES (XLR), Unbalanced Analog (TRS). Output (Upper row XLR connectors) – Digital AES (XLR)



Configuration 2: Input Balance Analog & Digital AES. Analog L/R Input on XLR connectors – Left (upper row XLR)/Right (lower row XLR). Digital AES Input on TRS connectors (lower row). No physical output connectors.



Figure 2.3.1: Model Q24-4888 REAR/BACK Panel. Two Configurations Offered as above – match with your ordered product



Figure 2.3.2: Model Q24-4880 REAR/BACK Panel. Digital AES I/O only. No analog audio I/O

Connecting the Power Supply: Q24-4888/Q24-4880 box can operate at 110-250V, 50Hz/60Hz voltage rating. Verify your line settings before connecting to the unit. Use the supplied power cable or another cable of equivalent rating.

2.3.1 Back/Rear Panel details and Input and Output Connections of Q24-4888:

The rear panel of the Q24-4888 provides all audio input and output connections, network interfaces, and power supply connections. Connectors are arranged in clearly labelled groups from left to right: AES Output, Input (XLR-Digital/AES and TRS Stereo Unbalanced Analog Input), Ethernet, MUX/ASI Input/Output and Power.

Configuration 1

AES Output — Channels 1 to 8

The upper row of connectors provides eight independent AES/EBU digital audio outputs, numbered 1 through 8, on XLR-3M male connectors. Each output carries an independently processed stereo AES3 digital audio stream, allowing simultaneous distribution to multiple destinations such as FM exciters, HD Radio exciters, STL systems, or monitoring equipment.

Each output channel is independently assignable in software via the SoundMax™ Manager PC application, enabling flexible routing of processed audio streams to different destinations.

Note: All AES/EBU connections require 110-ohm balanced XLR cables to ensure correct signal integrity and AES3 standard compliance.

Input Connections

The lower row of connectors provides the audio inputs to the Q24-4888 and is divided into two sections:

XLR-AES — Digital Input Channels

The left portion of the input row provides AES/EBU digital audio inputs on XLR-3F female connectors. These accept stereo AES3 digital audio streams from external sources. Supported input sample rates are 32 kHz, 44.1 kHz, and 48 kHz. Input sample rate is auto-detected — no manual selection is required.

TRS — Stereo Analog unbalanced Input Channels

The right portion of the input row provides stereo unbalanced analog audio inputs on TRS (Tip-Ring-Sleeve) connectors. These accept unbalanced line-level stereo analog audio signals from external sources such as audio consoles, payout systems, or other analog programme sources.

Note: Ensure the correct input type is selected in the SoundMax™ Manager software to match the physical connection in use. Connecting to an input that does not correspond to the selected input type in software will result in no audio being received by the processor.

Network and Data Connectors

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

Multiplex ASI Input/Output Connectors:

- **Mux/ASI Input:** It takes in a complete, encoded **MPEG Transport Stream (MPEG-TS)** via a 75-ohm BNC coaxial cable. The processor internally demultiplexes (separates) the incoming stream based on PID (Process ID).
- **DeMux/ASI Output:** The Demultiplexed output stream is rendered out through this port.

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).

Configuration 2

Input Connections

Top Row

The Top row portion of the connectors provides Left-channel audio input on XLR-Female connectors.

XLR-Left Analog Input Channels

The Top row portion of the connectors provides Left-channel audio input on XLR-Female connectors. These accept mono Left Analog audio streams from external sources. Supported input sample rates are 32 kHz, 44.1 kHz, and 48 kHz. This accepts Right-channel analog audio signals from external sources such as audio consoles, playout systems, or other analog programme sources.

Bottom row

The bottom row of connectors provides the audio inputs to the Q24-4888 and is divided into two sections:

XLR-Right Analog Input Channels

The bottom portion of the input row provides Right-channel audio input on XLR-TRS Combo connectors. These accept mono Right Analog audio streams from external sources. Supported input sample rates are 32 kHz, 44.1 kHz, and 48 kHz. This accepts Right-channel analog audio signals from external sources such as audio consoles, playout systems, or other analog programme sources

TRS — Digital Input Channels

The bottom row of the combo input connector provides Digital AES/EBU audio inputs on TRS (Tip-Ring-Sleeve) connectors. These accept unbalanced line-level stereo.

Note: Ensure the correct input type is selected in the SoundMax™ Manager software to match the physical connection in use. Connecting to an input that does not correspond to the selected input type in software will result in no audio being received by the processor.

Network and Data Connectors

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

Multiplex ASI Input/Output Connectors:

- **Mux/ASI Input:** It takes in a complete, encoded **MPEG Transport Stream (MPEG-TS)** via a 75-ohm BNC coaxial cable. The processor internally demultiplexes (separates) the incoming stream based on PID (Process ID).
- **DeMux/ASI Output:** The Demultiplexed output stream is rendered out through this port.

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 Back/Rear Panel details and Input and Output Connections of Q24-4880:

The rear panel of the Q24-4880 provides Digital output connections, network interfaces, and power supply connections. Connectors are arranged in clearly labelled groups from left to right: AES Output, Ethernet (TS/AOIP, Control/Stream), and Power.

AES Output — Channels 1 to 8

The upper row of connectors provides eight independent AES/EBU digital audio outputs, numbered 1 through 8, on XLR-3M male connectors. Each output carries an independently processed stereo AES3 digital audio stream, allowing simultaneous distribution to multiple destinations such as FM exciters, HD Radio exciters, STL systems, or monitoring equipment.

Each output channel is independently assignable in software via the SoundMax™ Manager PC application, enabling flexible routing of processed audio streams to different destinations.

Note: All AES/EBU connections require 110-ohm balanced XLR cables to ensure correct signal integrity and AES3 standard compliance.

AES Input — Channels 1 to 8

The lower row of connectors provides eight independent AES/EBU digital audio inputs, numbered 1 through 8, on XLR female connectors. Each input carries stream from the source to 1 channel stereo of audio processor for processing.

Network and Data Connectors

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

Power Supply:

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

2.3.3 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 (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 +20 dBu. Level control of the analog inputs and outputs is accomplished via front-panel control through System Setup.

2.3.4 Connecting Digital Audio Equipment:

The digital input and outputs conform to the professional AES/EBU standard. They have sample rate converters supporting 32 kHz, 44.1 kHz, and 48 kHz. Input sample rate is auto sensed. For best peak control, operate at 44.1 kHz or higher. Digital input is via one XLR-type female connector; digital outputs via two XLR-type male connectors. Levels are front-panel configurable.

2.4 Front Panel Controls

Figures 2.4.1 below shows the Front Panel Controls for both Models Q24-4888/Q24-4880



Figure 2.4.1: Model Q24-4888/Q24-4880 FRONT PANEL

1. **Power Switch:** Use this switch to turn the unit ON or OFF. It remembers last power state.
2. **TFT Touch Screen:** The equipment is equipped with a front panel TFT touch screen for 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 particular 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.

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 (NMS) Software Installation (may be skipped during initial setup)

Model Q24-4888/ Q24-4880 hardware is supplied with SoundMax™ Manager software for remote computer interface. This software is provided with CD-ROM or can be downloaded from [ATC Labs weblink](#).

Features

SoundMax™ Manager allows for elaborate configuration, control, monitoring, and tuning of multiple Q24-4888/Q24-4880 processors connected to a network. The presence of one or more Perceptual SoundMax™ Model Q24-4888/Q24-4880 on the network is automatically sensed by the SoundMax™ Manager running on any computer connected to that network.

Functionality

By using this software, users have complete flexibility to configure and monitor the Q24-6111 processors. The supported functionality includes:

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

Monitoring Capabilities

Monitoring capabilities include:

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

2.5.1 Installing Perceptual SoundMax™ Manager

Minimum system requirements

- **Operating System:** SoundMax™ Manager 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 SoundMax™ Manager 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 SoundMax™ Manager setup on a PC and follow the instructions

- After the SoundMax™ Manager installation, the user interface screen (UI) shown in Figure 2.5.1 below will appear.

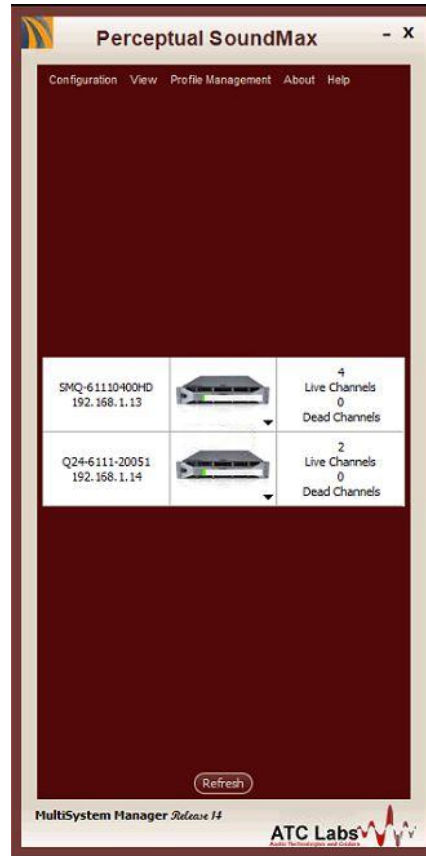


Figure 2.5.1: SoundMax™ Manager UI Screen

- SoundMax™ Manager will auto-detect the “Model Q24-4888/Q24-4880 rack mount encoder/decoder” as Live Station. By clicking on the station from this list, you can get a detailed view (as shown in Figures 2.5.2 and 2.5.3 below) and options to configure that particular station.

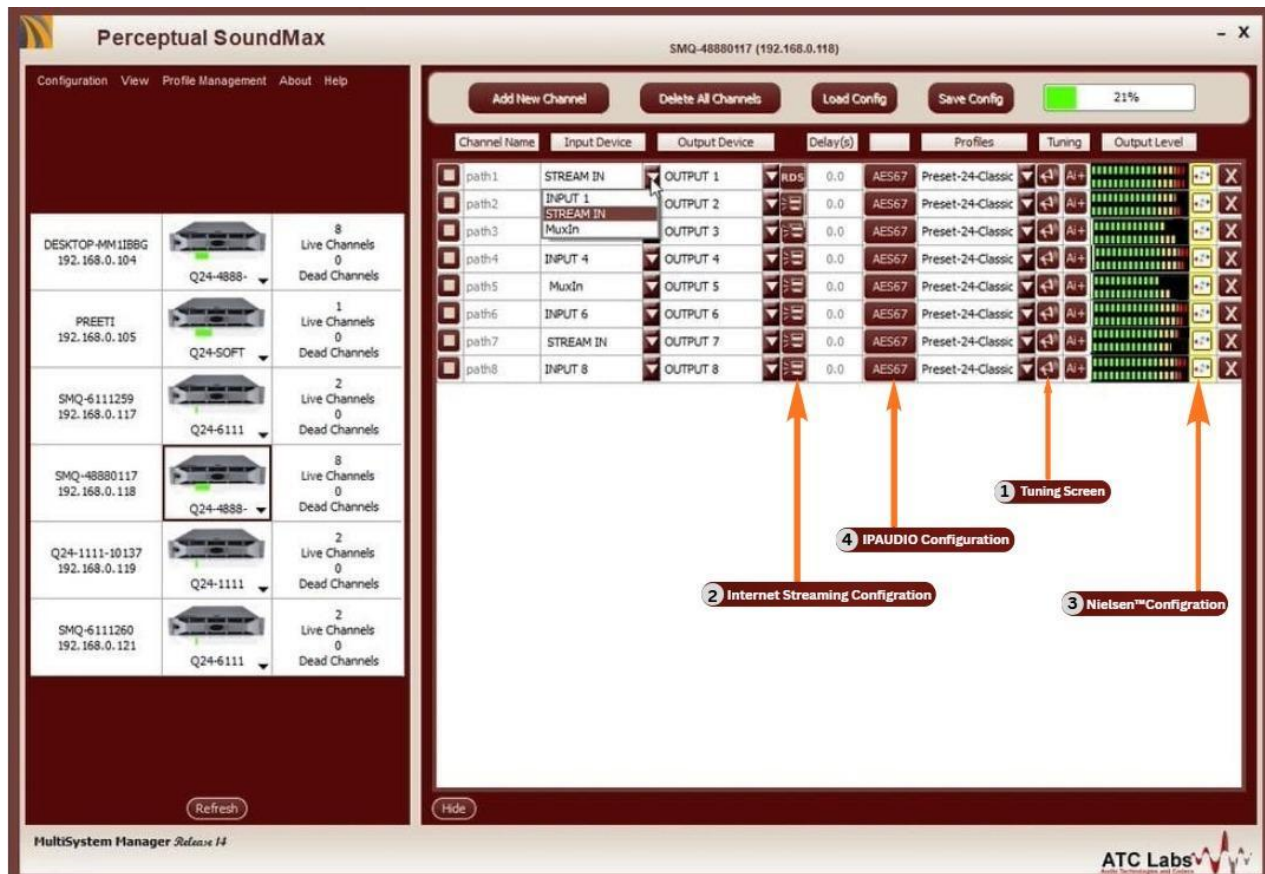


Figure 2.5.2: SoundMaxManager™ Detailed View (Model Q24-4888)

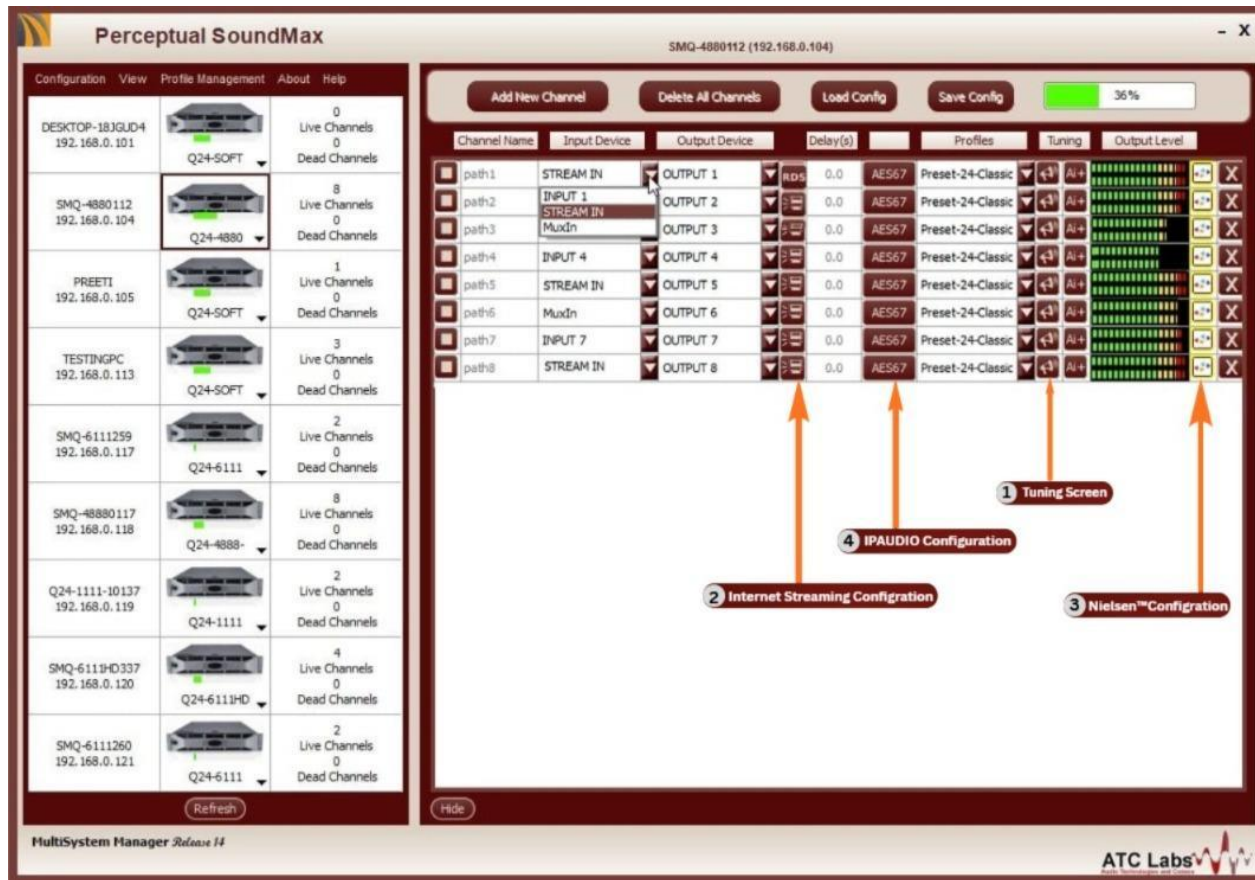


Figure 2.5.3: SoundMaxManager™ Detailed View (Model Q24-4880)

This view provides the ability to configure the following:

- For Model Q24-4888- It allows detailed configuration of encoding such as Input selection (**Analog, Digital, Stream**), bitrates selection, codec selection, and streaming parameters. Fine-tuning of encoding and processing parameters can also be performed independently for each channel along with the configuration for distribution.
- For Model Q24-4880- It allows detailed configuration of decoding such as Input selection (**Digital, Stream**), bitrates selection, codec selection, and streaming parameters. Fine-tuning of decoding and processing parameters can also be performed independently for each channel along with the configuration for distribution
- It provides an option to tune the audio quality of any channel based on the requirement. Clicking on the tune button will change your view to a tuning screen showing all the tunable modules and an option to save custom profiles and load saved profiles. For a more detailed explanation on TUNING through the manager refer to Section 5.

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

2.6 Multiplexer and Demultiplexer

The Multiplexer and the Demultiplexer are optional features of the Model Q24-4888 and Q24-4880, set up from **SoundMax Manager**. They are used when audio has to travel as an **MPEG transport stream (MPEG-TS)** instead of as separate audio feeds.

- The **Multiplexer** combines the audio of the unit's channels into one transport stream in which every channel is its own programme (service). The combined stream is sent over the IP network and, on a Q24-4888 fitted with an ASI interface, to the ASI output (MUX OUT).
- The **Demultiplexer** works in the opposite direction. It receives an incoming transport stream from the IP network or from the ASI card, shows what is inside it (video, audio and data streams), and lets you either pass selected streams on into the Multiplexer output or use an audio stream as the input of one of the unit's channels.

Note: The Multiplexer is available on the Q24-4888 and Q24-4880. The **Output To** selection (ASI output) and the Demultiplexer are available on the Q24-4888 only. Both are configured only from SoundMax Manager; they have no front-panel menu.

To open either panel, right-click the unit in the SoundMax Manager server list and choose **Multiplexer** or **Demultiplexer**.

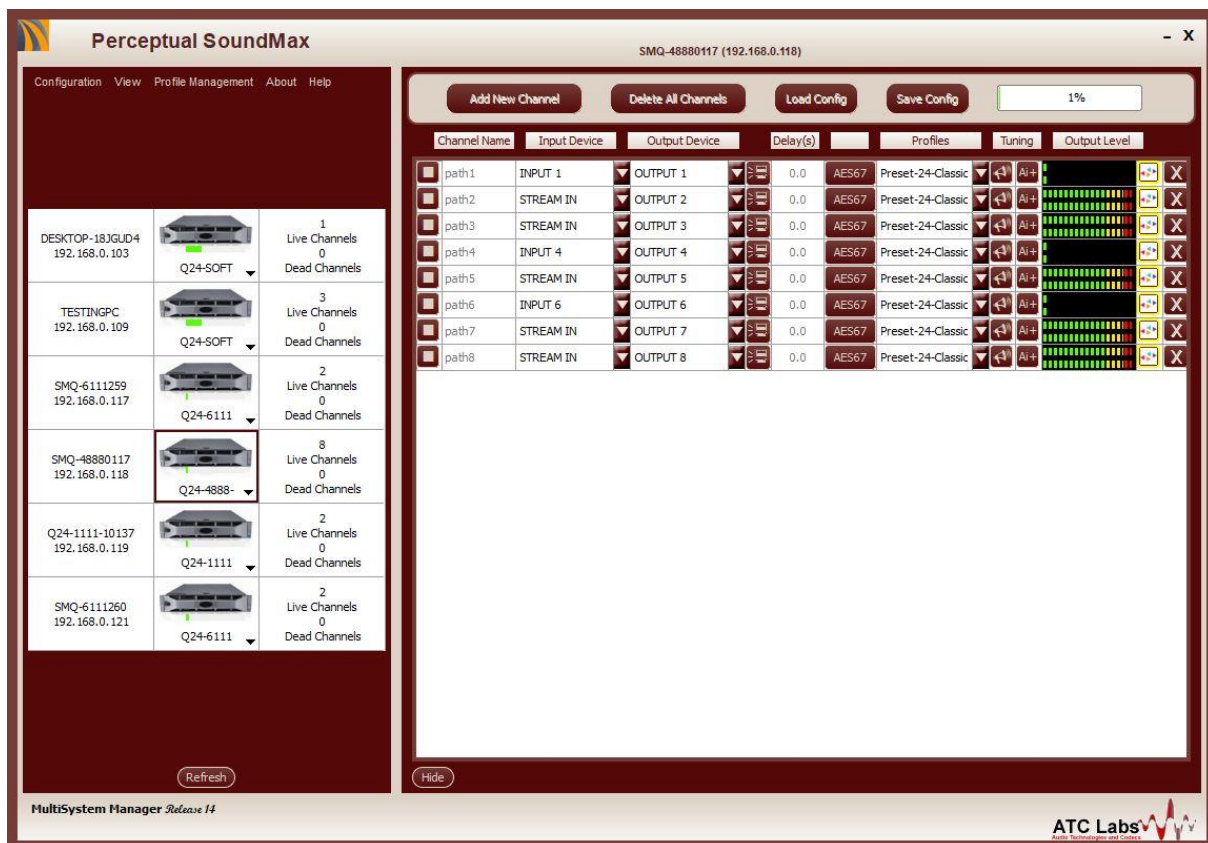


Figure 2.6.1: SoundMax Manager - Q24-4888 unit selected in the server list

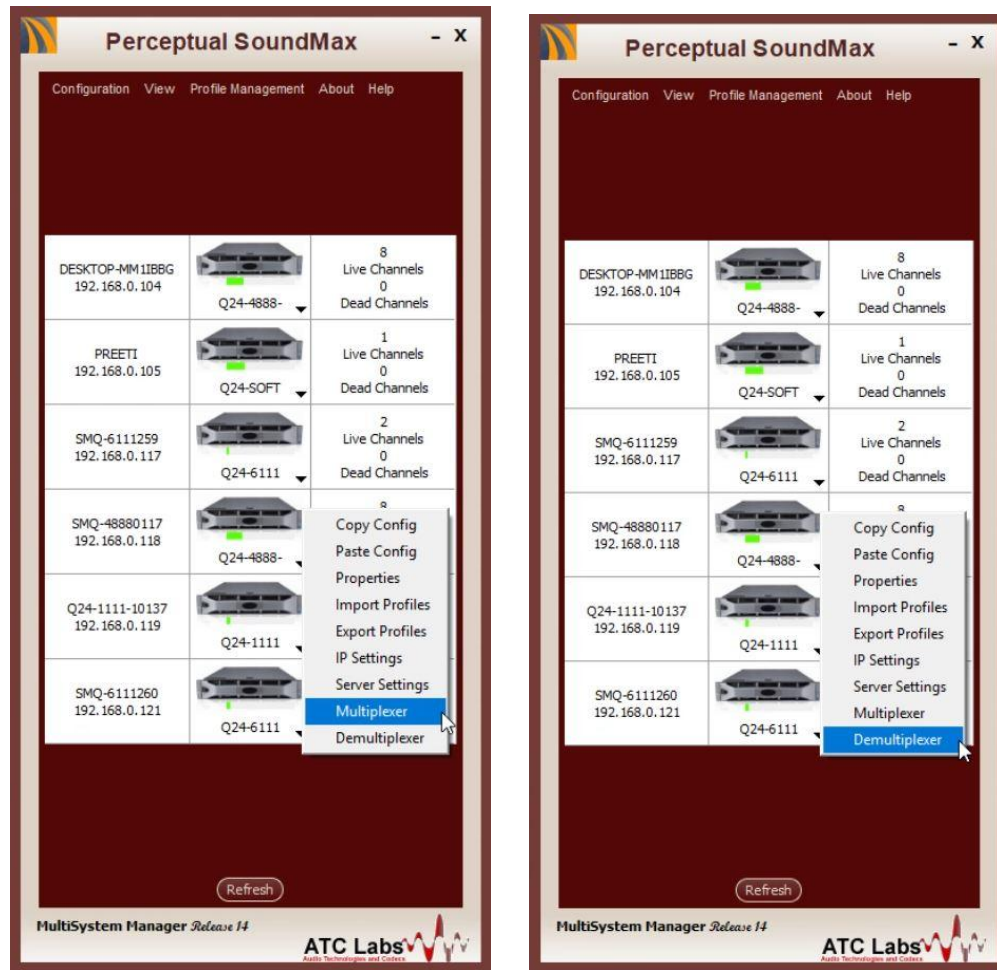


Figure 2.6.2: SoundMax Manager - right-click menu of a Q24-4888 unit showing "Multiplexer" (left) and "Demultiplexer" (right)

2.6.1 Multiplexer

How it operates

Each channel that is ticked in the Multiplexer panel encodes its processed audio (AAC or MP3) and hands it to the Multiplexer service that runs inside the unit. The service combines these streams into a single transport stream. Every ticked channel becomes one programme with its own **PID**, **Service ID**, **service name** and **service provider**, so a receiver or an encoder downstream can pick the channels apart again. The combined stream is then sent to the destination chosen in the panel.

Because the audio is taken after the channel's processing, everything set for the channel (profile, processing, delay, and so on) is already applied. The Multiplexer does not change the processing of a channel; it only packages the result.

Channel Number	Channel Name	Multiplexer Participation	PID	Service ID	Service Name	Codec	AAC Profile	AAC Codec Mode	Bitrate	Service Provider
1	path1	☒	011	124	MuxINSERVICE1	AAC	HE-v1	Stereo	32	ATC LABS
2	path2	☐	015	258	MuxINSERVICE2	MP3	LC	Stereo	64	ATC LABS
3	path3	☒	017	365	MuxINSERVICE3	MP3	LC	Stereo	96	ATC LABS
4	path4	☒	020	752	MuxINSERVICE4	AAC	HE-v2	Dual-Mono	128	ATC LABS
5	path5	☒	024	682	MuxINSERVICE5	MP3	LC	Stereo	192	ATC LABS
6	path6	☒	028	520	MuxINSERVICE6	AAC	HE-v1	Dual-Mono	256	ATC LABS
7	path7	☐	048	985	MuxINSERVICE7	MP3	LC	Stereo	320	ATC LABS
8	path8	☒	065	785	MuxINSERVICE8	AAC	HE-v2	Stereo	64	ATC LABS

Figure 2.6.3: Multiplexer panel of a Q24-4888 (settings on the left, per-channel table on the right)

Global settings (left side of the panel)

Setting	Description
Multiplexer Enable	Switches the Multiplexer on or off. With the box unticked, no combined stream is produced.
Output To (Q24-4888 only)	Where the combined stream goes: IP Network (UDP) - sent on the network only. ASI Card (TS Out) - sent to the ASI TS OUT connector only. IP Network + ASI Card - both at the same time. When "ASI Card" is chosen, the network fields below are greyed out because they are not used.
Unicast / Multicast	Unicast sends the stream to one receiver; Multicast sends it to a group address that any number of receivers can join.
TTL	Multicast only. The number of network hops the stream may cross (default 50).
Group IP / Destination	Multicast: the group address (224.0.0.0 - 239.255.255.255). Unicast: the IP address of the receiver.
Port	UDP port of the stream (default 8000).
Interface	The network connector the stream leaves from: CONTROL/STREAM or AOIP (as labelled on the rear panel).

Per-channel table (right side of the panel)

Column	Description
Channel Number / Name	The unit's channels (path1 to path8).
Multiplexer Participation	Tick to include the channel in the combined stream. At most 16 channels may take part; the 8 channels of the Q24-4888/Q24-4880 are always within this limit.

Column	Description
PID	Packet identifier of the channel's audio in the combined stream. Give every channel a different PID.
Service ID	Programme number of the channel in the combined stream. Give every channel a different Service ID.
Service Name / Service Provider	Names shown by receivers for this programme.
Codec	AAC or MP3. AAC is recommended; the AAC Profile and AAC Codec Mode columns apply to AAC only.
AAC Profile	LC, HE-v1 or HE-v2.
AAC Codec Mode	Stereo or Dual-Mono.
Bitrate	Audio bitrate in kbps: 32, 64, 96, 128, 192, 256 or 320.
Input Source	Read-only. Shows where the channel's own input comes from: Local input , STREAM IN or MuxIn (fed by the Demultiplexer, see below).

To set up the Multiplexer

1. Open the Multiplexer panel (right-click the unit, **Multiplexer**).
2. Tick **Multiplexer Enable**.
3. On a Q24-4888, choose **Output To**: IP Network (UDP), ASI Card (TS Out) or both.
4. For network output, choose **Unicast** or **Multicast**, then enter the Group IP (or the receiver's IP), the Port and the Interface.
5. Tick **Participation** for every channel that is to be in the combined stream, and set its PID, Service ID, service name, codec and bitrate.
6. Click **OK**. The Multiplexer service is restarted with the new settings.

Note: Applying changes restarts the Multiplexer service. The combined stream is interrupted for a few seconds while it restarts, but the unit's normal audio outputs are not affected. If nothing in the panel was changed, pressing OK does not restart anything.

Note: ASI output needs the DekTec DTA-2172 card and its driver installed in the unit. If the card cannot be opened, the ASI output is switched off for that run and the network output (if selected) keeps working.

Where the Multiplexer output is shown

- **Front-panel TFT:** on the Output tab of a channel, MUX output is ticked when the output of the channel is sent in the Multiplexer output, and the destination of the combined stream is shown next to it: the network address and port, plus ASI Card when the ASI output is also selected (Output To).

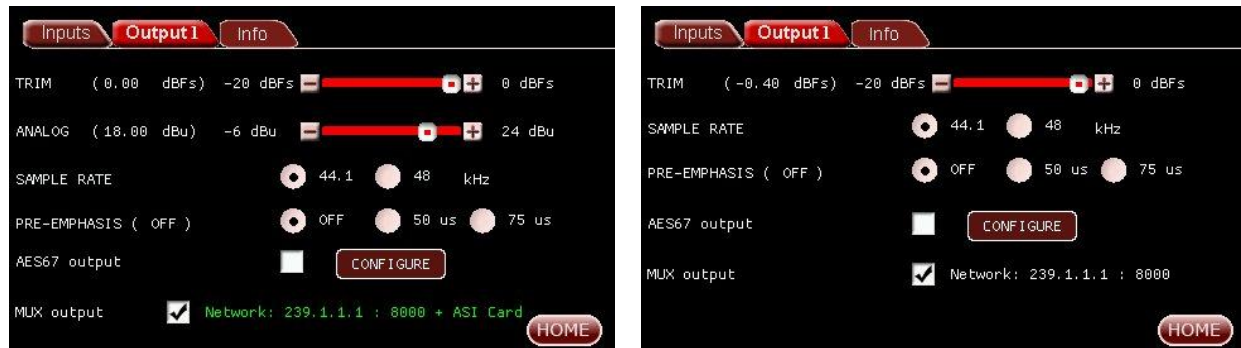


Figure 2.6.4: Front-panel Output screen with MUX output ticked - Q24-4888 (left) and Q24-4880 (right)

Note: For a detailed description of the TFT MUX output, please refer to Section 3.3.4 Output Configuration.

2.6.2 Demultiplexer (Q24-4888)

How it operates

The Demultiplexer listens to an incoming transport stream, **scans** it and reports its contents. You then decide, stream by stream, what the unit does with it. Everything the unit needs to know about the input is read from the stream itself.

Choose the **Input Source**:

- **IP Network** - choose Unicast or Multicast, enter the Group IP (multicast only) and the Port, and choose the Interface (CONTROL/STREAM or AOIP) the stream arrives on.
- **ASI Card** - the stream is read from the ASI input of the DekTec DTA-2172 card; no further settings are needed.

Press **Scan**. For up to about ten seconds the unit listens to the input, then the table lists the stream. The line above the table gives the service name, service ID, provider, PCR PID and total bitrate of the stream, and how many video, audio and data streams it carries. Each row of the table is one stream:

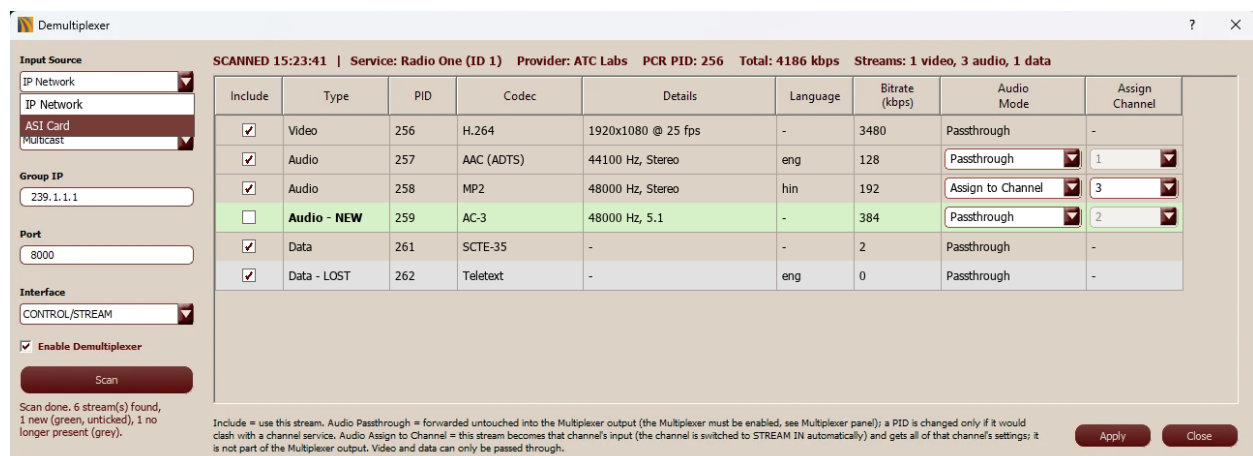


Figure 2.6.5: Demultiplexer panel after a Scan (input settings on the left, stream table on the right)

Column	Description
Include	Tick the streams the unit should use. Streams are shown unticked after a first scan; tick the ones you need.
Type	Video, Audio or Data.
PID	Packet identifier of the stream in the incoming transport stream.
Codec	For example H.264, MPEG-2 Video, AAC (ADTS), MP2, MP3, AC-3, SCTE-35, Teletext.
Details	Video: picture size and frame rate. Audio: sample rate and channel layout (Mono, Stereo, 5.1 ...).
Language	Language code signalled for the stream (for example eng, hin), if any.
Bitrate (kbps)	Measured bitrate of the stream.
Audio Mode	Audio streams only: Passthrough or Assign to Channel (see below). Video and data streams can only be passed through.
Assign Channel	With Assign to Channel: the channel (1 to 8) that takes this audio as its input.

What the two Audio Modes do

Audio Mode	Result
Passthrough	<p>The stream is copied, unchanged, into the Multiplexer output as an extra programme next to the channel programmes. Nothing is decoded or re-encoded. Its PID is changed only if it would clash with a PID already used by a channel programme. Video and data streams (for example SCTE-35) can be passed through in the same way.</p> <p>Passthrough only reaches the output while the Multiplexer is enabled and Output To includes the destination you need.</p>
Assign to Channel	<p>The audio stream becomes the input of the chosen channel. The channel's input is switched automatically to MuxIn, and the stream receives all the processing and settings of that channel exactly as any other input. The assigned audio is not part of the Multiplexer output by itself; if the channel is ticked in the Multiplexer panel, its processed audio is multiplexed like any other channel.</p> <p>One audio stream can be assigned to a channel; two streams cannot share a channel.</p>

Applying

Tick **Enable Demultiplexer** and press **Apply**. The selection is saved in the unit and the Multiplexer service is restarted if needed. If passthrough streams are selected but the Multiplexer is not enabled, Manager saves the selection and reminds you to enable the Multiplexer; nothing is sent until you do.

Note: Pressing Scan only looks at the input; it does not change the saved setup. Changes take effect only when you press Apply.

Live information

When the Demultiplexer has been applied, the panel shows **live** information while it is open. The line above the table starts with **LIVE** and the time of the last update, and the table keeps itself up to date every few seconds without disturbing the selection you are working on:

- A stream that appears in the input is added as a green row marked **NEW**, unticked, so that new streams are easy to spot. The tag stays until you press Apply.
- A stream that disappears from the input is kept in the table, shaded grey and marked **LOST**, together with its saved selection, so it is picked up again if it returns.
- If the input stops, the status line reports **NO DATA** and all streams are marked LOST until data returns.

If the Demultiplexer has not been applied, there is no live view. When the panel is opened with a saved setup, the unit scans the input once automatically; press **Scan** at any time to look again.

Where MuxIn is shown

- **Channel list in SoundMax Manager:** the Input drop-down of a channel offers INPUT n, STREAM IN and **MuxIn**. A channel fed by the Demultiplexer shows MuxIn. MuxIn cannot be chosen from the drop-down; it is set by assigning an audio stream in the Demultiplexer panel.
- **Multiplexer panel:** the Input Source column shows MuxIn for such a channel, and the streams passed through from the Demultiplexer are listed as additional shaded rows after the channels.
- **Front-panel TFT:** on the Input screen of a channel fed by the Demultiplexer, the MuxIn mark is selected instead of Stream Input, and the line below it shows the input (for example the network address and port, or ASI Card) and the assigned stream (PID, codec, sample rate and channel layout), just as the stream address is shown for Stream Input.

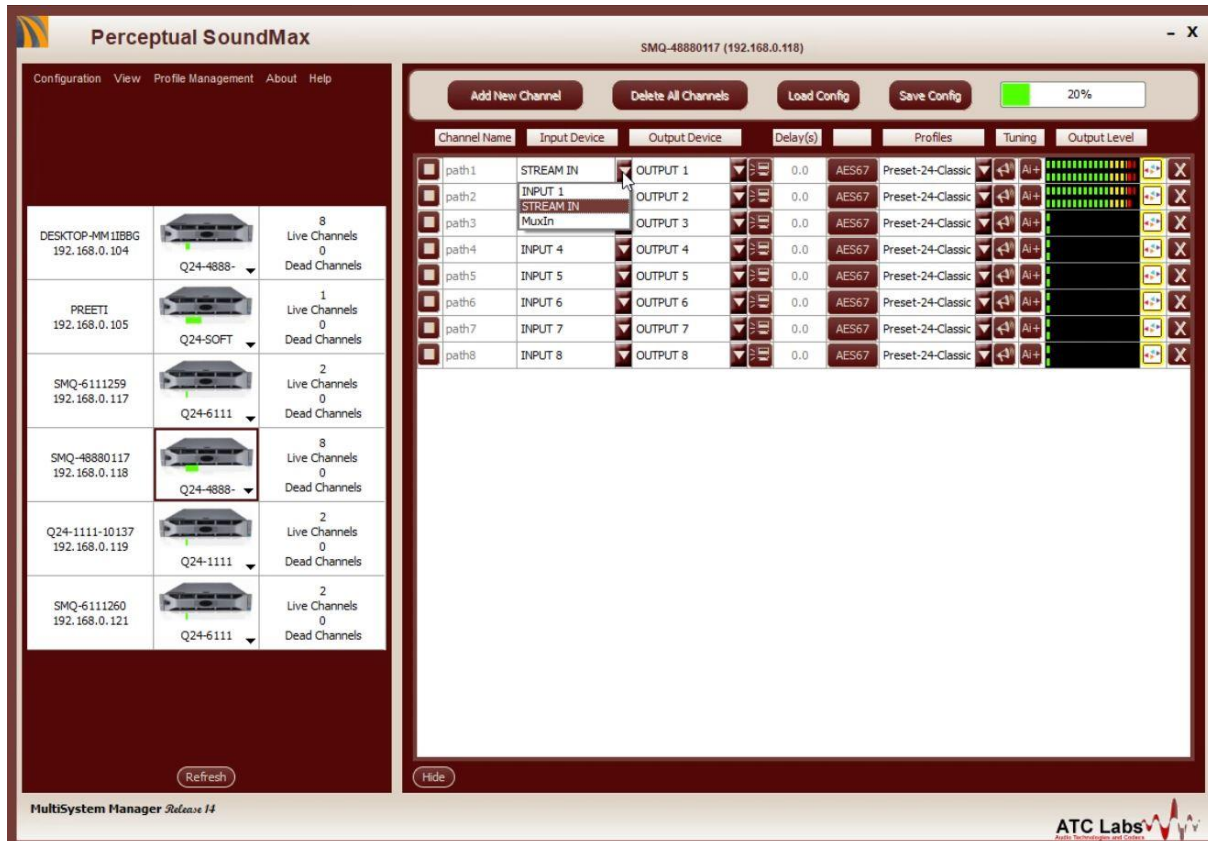


Figure 2.6.6: SoundMax Manager channel list - Input drop-down showing INPUT n, STREAM IN and MuxIn

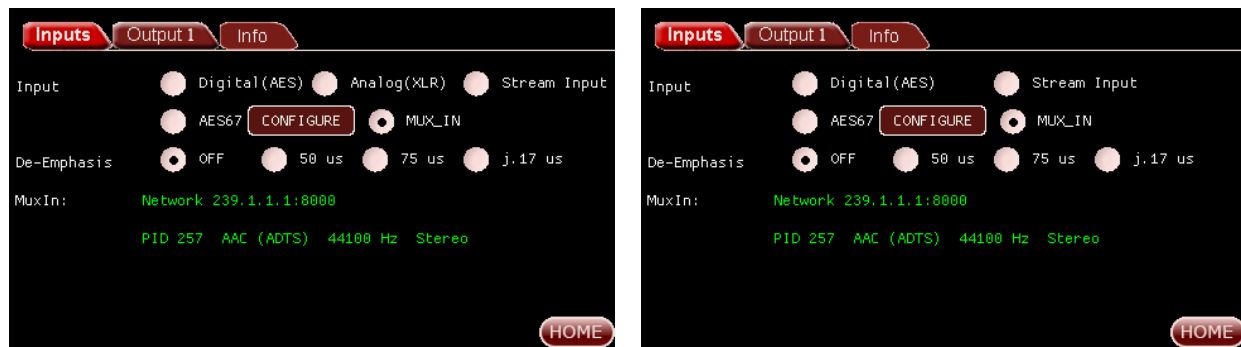


Figure 2.6.7: Front-panel Input screen with MUX_IN selected - Q24-4888 (left) and Q24-4880 (right)

Note: For a detailed description of the TFT MUX_IN input, please refer to Section 3.3.1 Input Configuration.

When a channel's input is changed by hand

If the input of a channel that is fed by the Demultiplexer is changed to something else (from SoundMax Manager, from the front panel, or by an AES67 setting), the unit releases that channel from the Demultiplexer automatically within a few seconds. The stream is un-ticked in the saved setup, so the Demultiplexer panel no longer shows it as assigned and a later Apply cannot silently take the channel back. To use the stream for the channel again, tick it, choose Assign to Channel and press Apply.

When the Multiplexer service runs

Situation	Multiplexer service
Multiplexer enabled	Runs normally: combines the channels, applies Output To, and carries any passthrough streams.
Multiplexer off, at least one channel assigned from the Demultiplexer	Runs only to receive the input and feed the assigned channels. It produces no combined stream and no network or ASI output.
Multiplexer off, nothing assigned	Not running. Passthrough selections are saved but have no effect until the Multiplexer is enabled.

Limits

- The input must be a transport stream with a **single programme**. If it carries several programmes, the first one in the stream is shown.
- Up to **32 streams** of the input are listed.
- For H.265 video, AAC (LATM), E-AC-3 and DTS, only the codec name is shown; picture size or sample rate are not read.
- Only **one audio stream per channel** can be assigned.
- Assigned channels use the unit's internal loopback address (127.0.0.1) with UDP ports 18101 to 18108 for an IP input, or 18201 to 18208 for an ASI input. These ports are reserved for this purpose and must not be used by anything else.

2.6.3 Typical Procedures

A. Send the unit's channels as one multicast transport stream

1. Open the **Multiplexer** panel and tick **Multiplexer Enable**.
2. Choose **Multicast**, enter a Group IP (for example 239.1.1.1), the Port and the Interface.
3. Tick **Participation** for each channel and give each a different PID, Service ID and service name.
4. Click **OK**, then open the stream on a receiver at the chosen group address and port; each channel appears as its own programme.

B. Use the audio of an incoming transport stream as the input of a channel

1. Open the **Demultiplexer** panel. Choose **IP Network** (or **ASI Card**) and enter the stream's address, port and interface.
2. Press **Scan** and wait for the stream list.
3. Tick **Include** for the audio stream you want, set **Audio Mode** to **Assign to Channel** and choose the channel.
4. Tick **Enable Demultiplexer** and press **Apply**.
5. The channel's input now shows **MuxIn** in the channel list and on the front panel. Adjust the channel's profile and processing as for any other input.

C. Pass a video or data stream through to the output

1. Make sure the Multiplexer is enabled and **Output To** is set (Multiplexer panel).
2. Open the **Demultiplexer** panel, enter the input and press **Scan**.
3. Tick **Include** for the video, data or audio streams to pass through (leave audio on **Passthrough**).

4. Tick **Enable Demultiplexer** and press **Apply**.
5. The streams appear in the Multiplexer output as an extra programme, and as shaded rows in the Multiplexer panel.

2.6.4 Troubleshooting

What you see	What to check
"No data received"	Check the address, port and interface, that the source is actually sending, and (for multicast) that the network passes the multicast group to the unit's interface.
"Data is arriving, but no program information (PAT/PMT) was found"	The input is not an MPEG transport stream, or the programme tables are missing from it.
"The input could not be opened"	The port is already used by something else, the address is wrong, or (ASI) the DTA-2172 card or its driver is not available.
"No live data"	The Demultiplexer has not been applied yet, or the Multiplexer service is not running. Press Scan to look at the input once.
Passthrough streams are not in the output	The Multiplexer must be enabled; check Output To; check the streams are ticked, Enable Demultiplexer is on and Apply was pressed.
An assigned channel is silent	Check the stream is present (LIVE view, not LOST), that Apply was done, that the channel is running, and that its input still shows MuxIn.
No ASI output	Check the DTA-2172 card and driver are present, that Output To includes ASI Card, and that the Multiplexer is enabled.

Note: If a problem persists, the Multiplexer log in the SoundMax Station data folder records what the Multiplexer service did (output target, input started, streams passed, card or network errors) and helps support to diagnose it.

2.7 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 Q24-4888/Q24-4880 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 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 2.7.1 below.

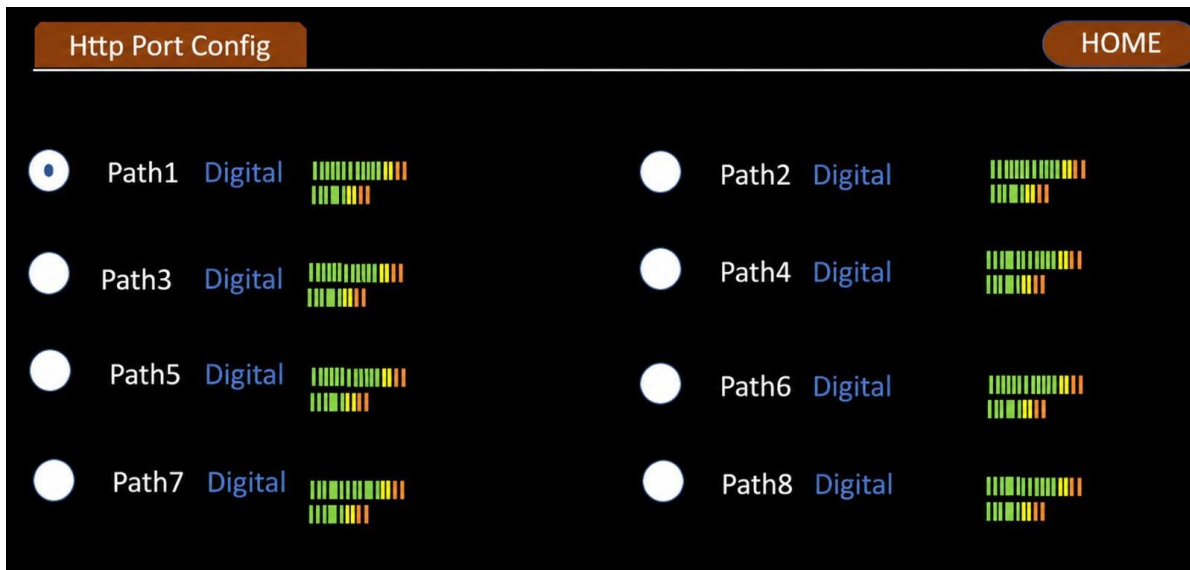


Figure 2.7.1: Selection Screen

- The Selection Screen lists the eight paths (Path1 to Path8). For each path it shows the input in use (for example Digital) and its level meters.
- Select the path (channel) to be operated. The Main Screen of the selected path then opens, as shown in Figure 2.7.2 below.

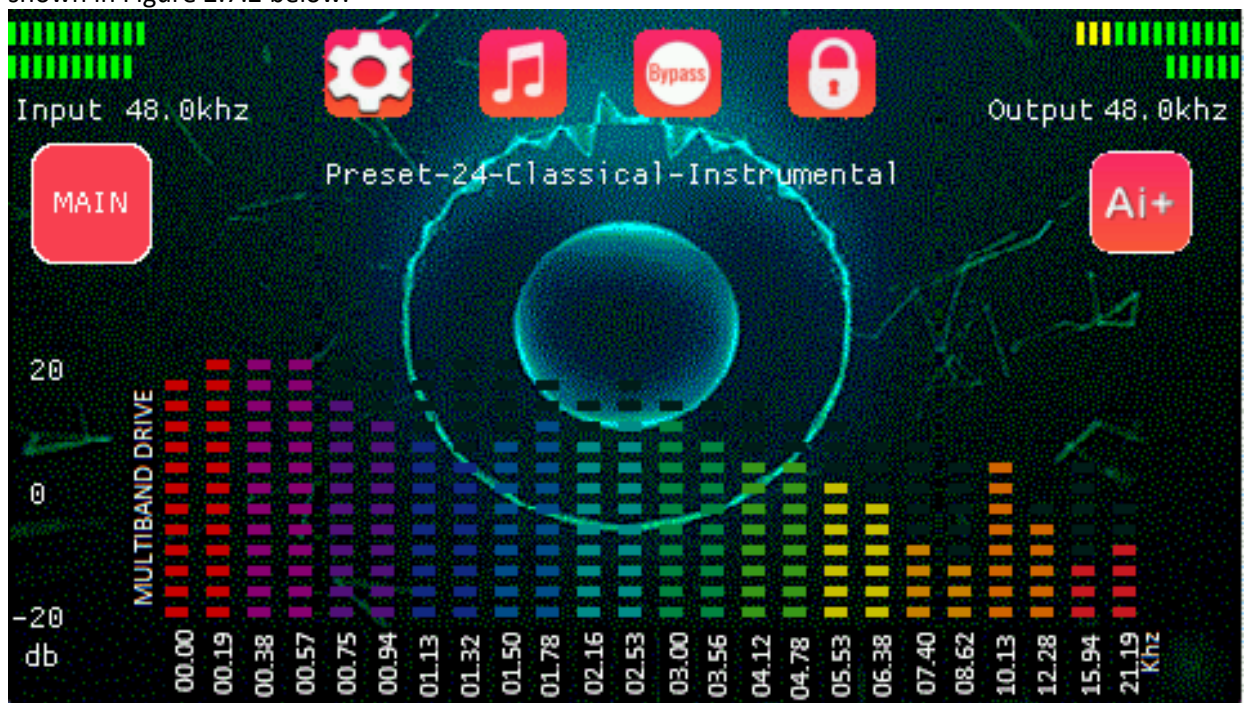


Figure 2.7.2: Main Screen of the Selected Path

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 Q24-4888 ships with the following default I/O configuration:

- **Input:** Stream Input, Analog (TRS unbalanced) and XLR-AES Digital Input
- **Output:** Digital (AES/EBU)

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

The Model Q24-4880 ships with the following default I/O configuration:

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

Step 5 — Channel Selection (Optional)

The user may select a channel number to access I/O settings, tuning, and monitoring controls. These are advanced settings described in **Section 3.2** and onwards.

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 6 — 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 Q24-4888/Q24-4880 Screen Controls**

Perceptual SoundMax™ Q24-4888/Q24-4880 features 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

At start-up, the front panel shows the Selection Screen (Figure 2.7.1), where the path (channel) is selected. The Main Screen of the selected path then opens.

Main Screen Display (Figure 2.7.2) labels the following items:

- **Soft keys:** Setting, Tuning, Bypass, Unlock/Lock, AI+, Channel Profile View
- Input and Output VU meter
- Output Spectrum in 24 Bandwidth central frequency display
- On-Air Preset indicating the currently active processing profile

Navigation:

- **AI+ Configuration**
 Opens the Auto Tuning & AI Genie configuration panel
- **Channel Profile View Key**
 Opens the Auto Tuning & AI Genie configuration panel
- **Setting**
 Displays the currently active processing profile for the selected channel.
- **Bypass Mode:**
 Click Setting to access Analog Input, Digital I/O, Composite, SCA/RBDS setup, and Info screens
 - 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 profile is shown (e.g., “OnAIR: Path1 – Preset-24 BASS-TREBLE”)
- **TUNING:**
 Click TUNING to view all factory presets and custom profiles & apply any one of them.
- **LOCK / UNLOCK:**
 Click the LOCK/UNLOCK key to lock or unlock the screen.
- **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, HD1, HD2, HD3 based on operational requirements.



Figure 3.2.1: Configure Screen

3.3 Input/Output Configuration

With the SETTING 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

For Q24-4888

The Model Q24-4888 supports Stream (IP based input), TRS stereo Unbalanced Analog input, XLR-Digital input through the AES/EBU standard and ASI based Multiplexed Input.

Input selection screen is fully software configurable, allowing flexible integration into different system setups. In addition, the unit provides software-selectable de-emphasis, ensuring precise control over audio signal processing and compatibility with various digital audio sources.

- **Word Length:** Supports 24-bit word length for high-resolution audio input.
- **Input Type:** The input source can be selected as Digital (AES), Analog (XLR), Stream Input, AES67 or MUX_IN. 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.
- **Reference Level:**
 - Analog Reference Level: Matches the analog input's maximum peak level to the Q24-4888/Q24-4880 ADC clipping point. Adjustable from –10 dBu to +12 dBu.
 - Digital Reference Level: Adjustable from –25 dBFS to 0 dBFS.
- **Peak Level:**
 - Analog Peak Level: Provides the required headroom for peaks above the reference level. Adjustable from 0 dBu to 12 dBu.
 - Digital Peak Level: Automatically managed by the AGC module, which detects peak levels and adjusts overall gain.
- **Prescale:** Adjustable from –10 dB to +20 dB.

Input Selection on the Inputs Tab (Q24-4888 and Q24-4880)

The Input row of the Inputs tab lists the inputs: **Digital (AES)**, **Analog (XLR)**, **Stream Input**, **AES67** and **MUX_IN**. The Inputs tab is the same on the Q24-4888 and the Q24-4880, except that the Analog (XLR) input is available on the Q24-4888 only. The selected input is shown by the filled button. The De-Emphasis row is the same for every input, while the rows below it change with the selected input, as shown in Figures 3.3.1 to 3.3.4; where both models are shown, the Q24-4888 screen is on the left and the Q24-4880 screen on the right. For each level control, the value currently set is shown in brackets at the right of the slider and is changed with the – and + keys or by moving the slider.

Digital (AES) Input

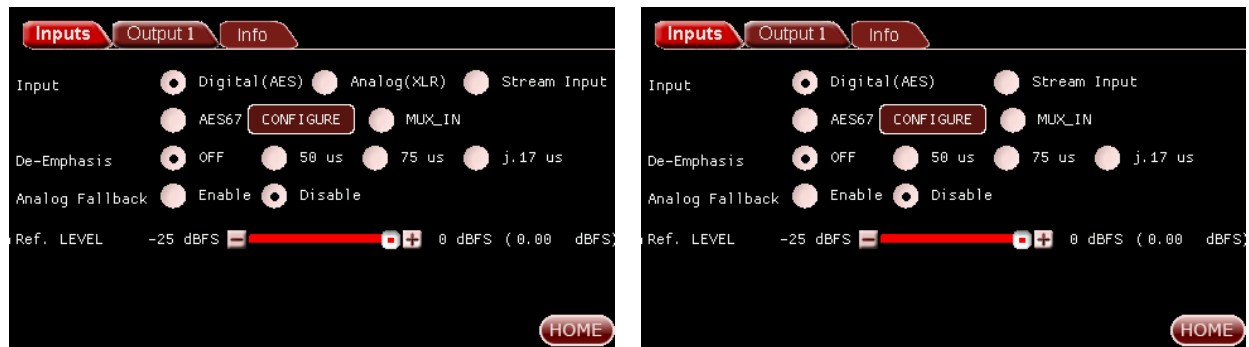


Figure 3.3.1: Input Configuration – Digital (AES) Input selected: Q24-4888 (left) and Q24-4880 (right)

- **Input – Digital (AES):** The channel processes the AES/EBU digital audio connected to its XLR digital input. The incoming sample rate is auto sensed.
- **De-Emphasis:** OFF, 50 μ s, 75 μ s or J.17. Select the de-emphasis that matches the pre-emphasis of the incoming audio; keep OFF for flat audio.
- **Analog Fallback:** Enable or Disable. When enabled, the channel falls back to the analog input automatically if the digital input becomes unavailable (used on the Q24-4888, which has analog inputs).
- **Ref. LEVEL:** Digital reference level, adjustable from –25 dBFS to 0 dBFS.

Analog (XLR) Input (Q24-4888 only)

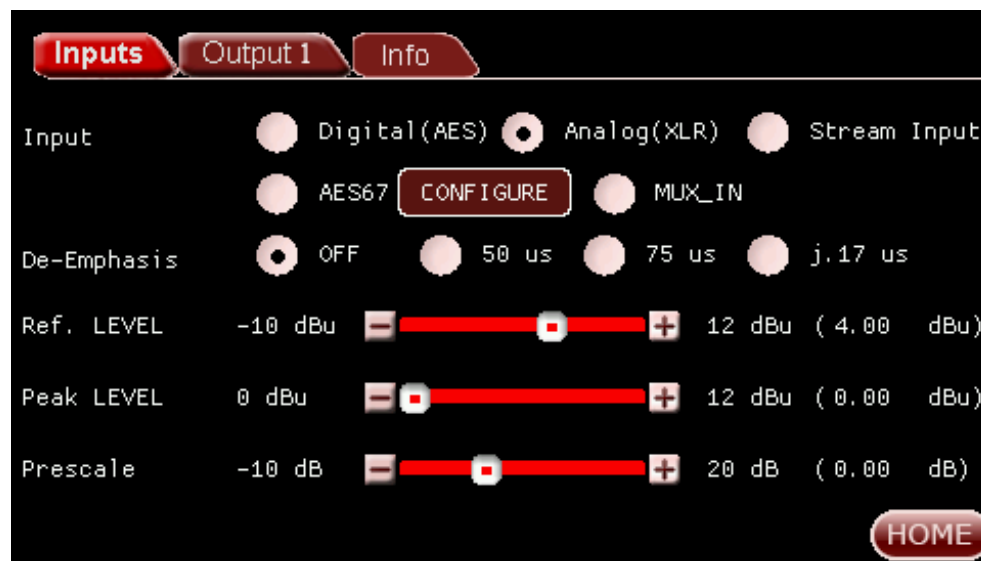


Figure 3.3.2: Input Configuration of Q24-4888 – Analog (XLR) Input selected

- **Input – Analog (XLR):** The channel processes the analog audio connected to its analog input.
- **De-Emphasis:** OFF, 50 μ s, 75 μ s or J.17, as for the digital input.
- **Ref. LEVEL:** Analog reference level, adjustable from –10 dBu to +12 dBu. It matches the maximum peak level of the analog input to the ADC clipping point.
- **Peak LEVEL:** Adjustable from 0 dBu to +12 dBu. It provides the headroom for peaks above the reference level.

- **Prescale:** Adjustable from –10 dB to +20 dB.

Stream Input

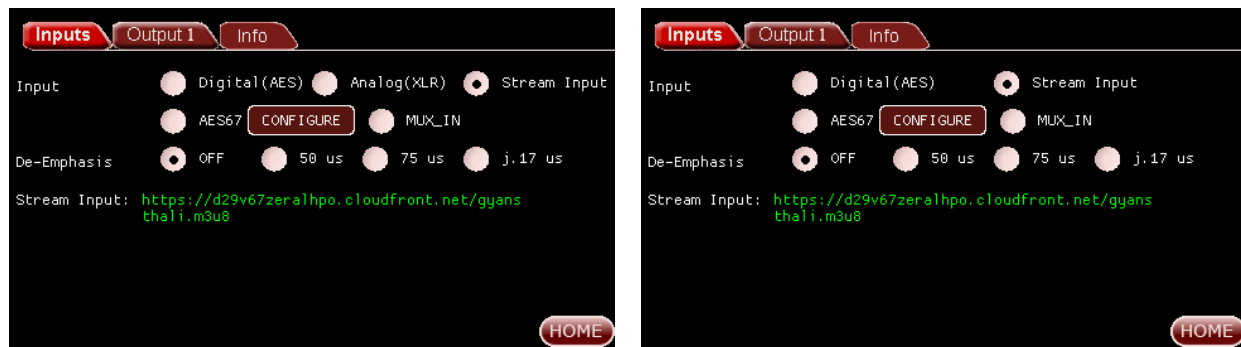


Figure 3.3.3: Input Configuration – Stream Input selected: Q24-4888 (left) and Q24-4880 (right)

- **Input – Stream Input:** The channel takes its audio from a network stream in compressed audio format (HTTP/UDP AAC-ADTS/TS, MP3-TS or HLS). The stream is configured from SoundMax™ Manager only (see Section 3.3.2).
- **Stream Input:** Shows the address (URL) of the stream being received, for example an HLS (.m3u8) link.
- **De-Emphasis:** Available as for the other inputs.

MUX_IN Input

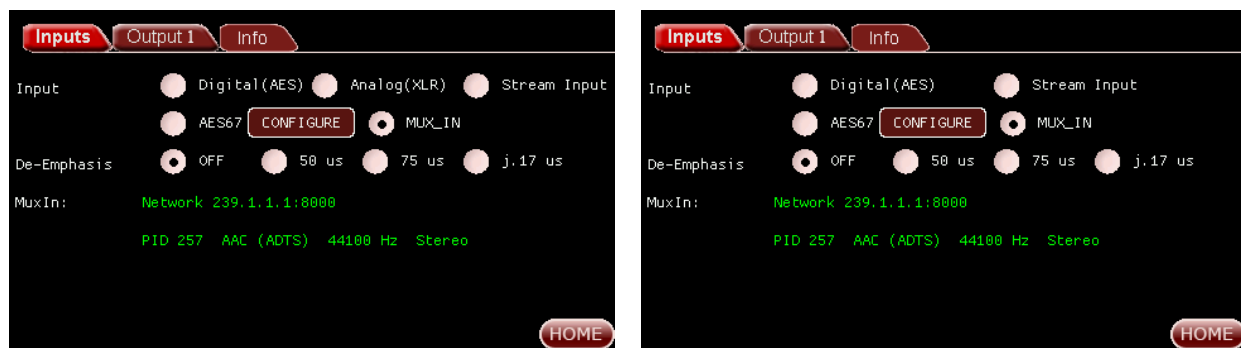


Figure 3.3.4: Input Configuration – MUX_IN Input selected: Q24-4888 (left) and Q24-4880 (right)

- **Input – MUX_IN:** The channel takes its audio from an audio stream of an incoming MPEG transport stream (MPEG-TS), received from the IP network or the ASI input and separated by the Demultiplexer. MUX_IN is set by assigning the audio stream to the channel in the Demultiplexer panel of SoundMax™ Manager (see Section 2.6).
- **MuxIn (first line):** The input of the Demultiplexer: the network address and port of the transport stream (for example Network 239.1.1.1:8000), or ASI Card.
- **MuxIn (second line):** The audio stream assigned to the channel: its PID, codec, sample rate and channel layout (for example PID 257 AAC (ADTS) 44100 Hz Stereo).
- **De-Emphasis:** Available as for the other inputs.

Note: If another input is selected for a channel that is fed by the Demultiplexer, the unit releases the channel from the Demultiplexer automatically (see Section 2.6).

AES67 Input

- **Input – AES67:** The channel takes its audio from an AES67 network audio stream. Press CONFIGURE to open the AES67 Input Settings screen described in Section 3.3.3.

3.3.2 Stream Audio Input (Configurable with SoundMaxManger™ Only)

For Q24-4888

The Model Q24-4888 audio processors support the audio Input 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 SoundMaxManger™. The Input Device drop down selection on the SoundMaxManger™ expanded screen allows for the input to be toggled between Digital, Analog and Stream; if the third option is selected, a panel as shown in Figure 3.3.7 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 3.3.5); reselecting Digital/Analog/AES67 input options directs the source back to the corresponding inputs.

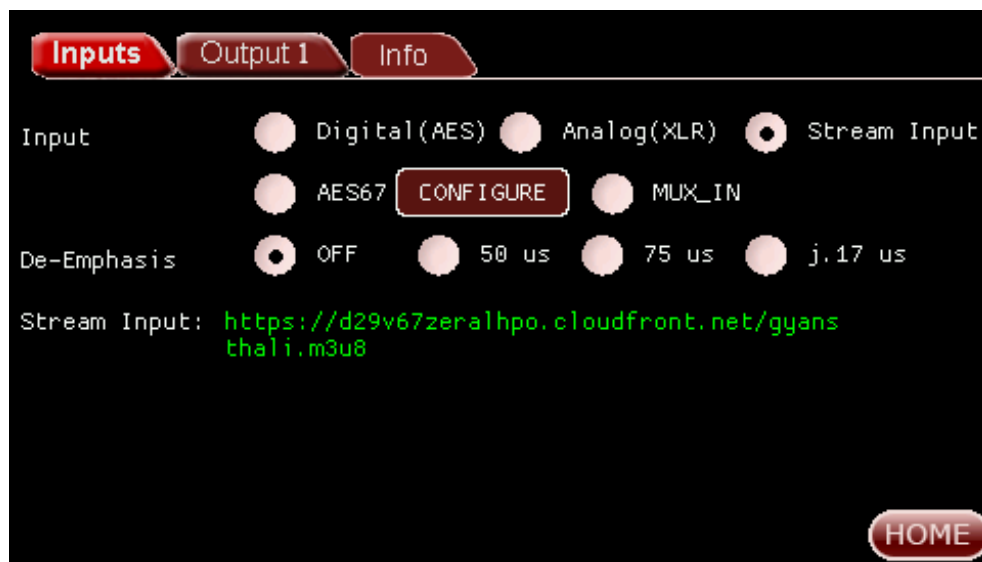


Figure 3.3.5: Input Configuration of Q24-4888 – Stream Input selected

For Q24-4880

The Model Q24-4880 audio processors support the audio Input 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 SoundMaxManger™. The Input Device drop down selection on the SoundMaxManger™ expanded screen allows for the input to be toggled between Digital and Stream; if the second option is selected, a panel as shown in Figure 3.3.7 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 3.3.6); reselecting Digital/Analog/AES67 input options directs the source back to the corresponding inputs.

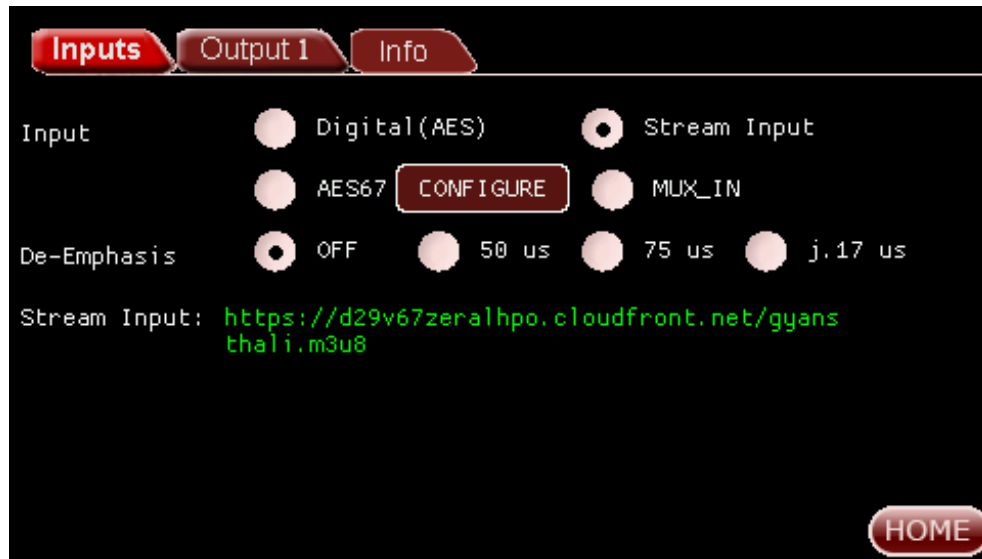


Figure 3.3.6: Input Configuration of Q24-4880 – Stream Input selected

X
Stream Audio Input

☒ **Enable HTTP/URL Input**

Stream Link

http://streamasia.atc-labs.com:8000/ATCTest.aac.m3u

Username

Password

Supports HTTP/HTTPS streaming URLs (Icecast, HLS)

☐ **Enable UDP / Multicast Input**

Protocol

MULTICAST

Stream Type

TS

IP Address

Port

0

Samplerate

44100

Format

L24

Channels

Supports TS/AAC/MP3/RAW UDP/Multicast streams

OK

Figure 3.3.7: 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 for 4888/4880

The AES67 Input Settings screen (as shown in Figure 3.3.8 below) enables setup of IP-based audio input streams compliant with AES67 standards. This allows the Q24-4888/Q24-4880 to receive high-quality uncompressed audio over an IP network for broadcast and studio applications.

AES67 INPUT SETTINGS

PORT

IP ADDRESS

SAMPLE RATE ☐ 44.1 ☒ 48 kHz

FORMAT ☐ L16 ☒ L24

CHANNELS ☒ STEREO ☐ MONO

Figure 3.3.8: AES67 Input Settings of Q24-4888/4880

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

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

For Q24-4888

The Model Q24-4888 provides AES digital outputs. The output configuration (as shown in Figures 3.3.9 and 3.3.10 below) is software-selectable, allowing it to be set either as flat or with pre-emphasis at 50 μ s or 75 μ s, depending on system requirements. The digital output can be configured as either a stereo main signal or as an MPX output operating at 192 kHz, providing flexibility for different broadcast and audio processing applications.

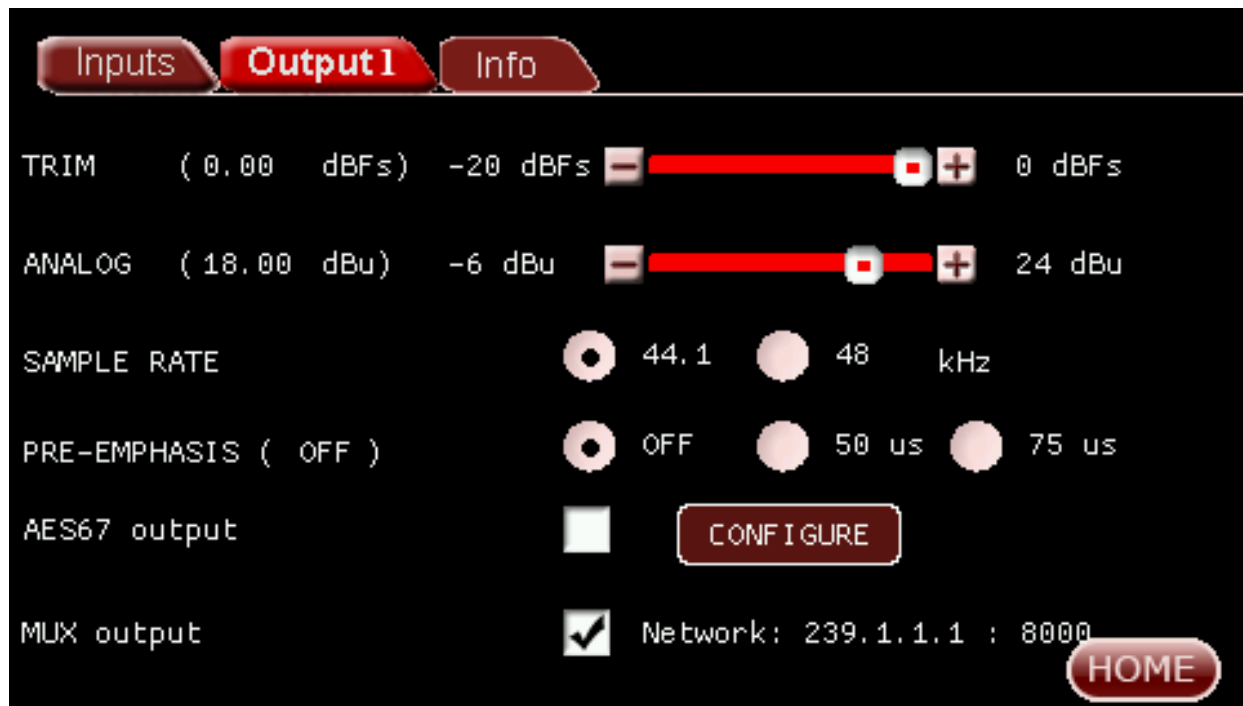


Figure 3.3.9: Output Configurations of Q24-4888

- **Output Level:**
 - Users should calibrate and set output levels by playing a full-scale tone as input.
 - Analog output levels: Range from -6 dBu to +24 dBu
 - Digital output levels: Range from -20 dBFS to 0 dBFS
- **Pre-emphasis:**
 - Provides pre-emphasized analog and digital outputs.
 - Types available: 50 μ s and 75 μ s
- **Output Sample Rate:**

- Software-selectable options: 44.1 kHz, and 48 kHz
- **Digital Output Selection:**
 - Can be configured as Stereo Main or Composite/MPX at 192 kHz
- **AES67 Output:**
 - Enables the AES67 network output of the channel. Press CONFIGURE to open the AES67 Output Settings screen (see Section 3.3.5).
- **MUX Output:**
 - When ticked, the output of the channel is sent in the Multiplexer output (MPEG transport stream, see Section 2.6).
 - The destination of the transport stream is shown next to it, according to the Output To setting of the Multiplexer panel (Section 2.6): the network address and port (for example Network: 239.1.1.1 : 8000), ASI Card, or both, as shown in Figure 3.3.10.

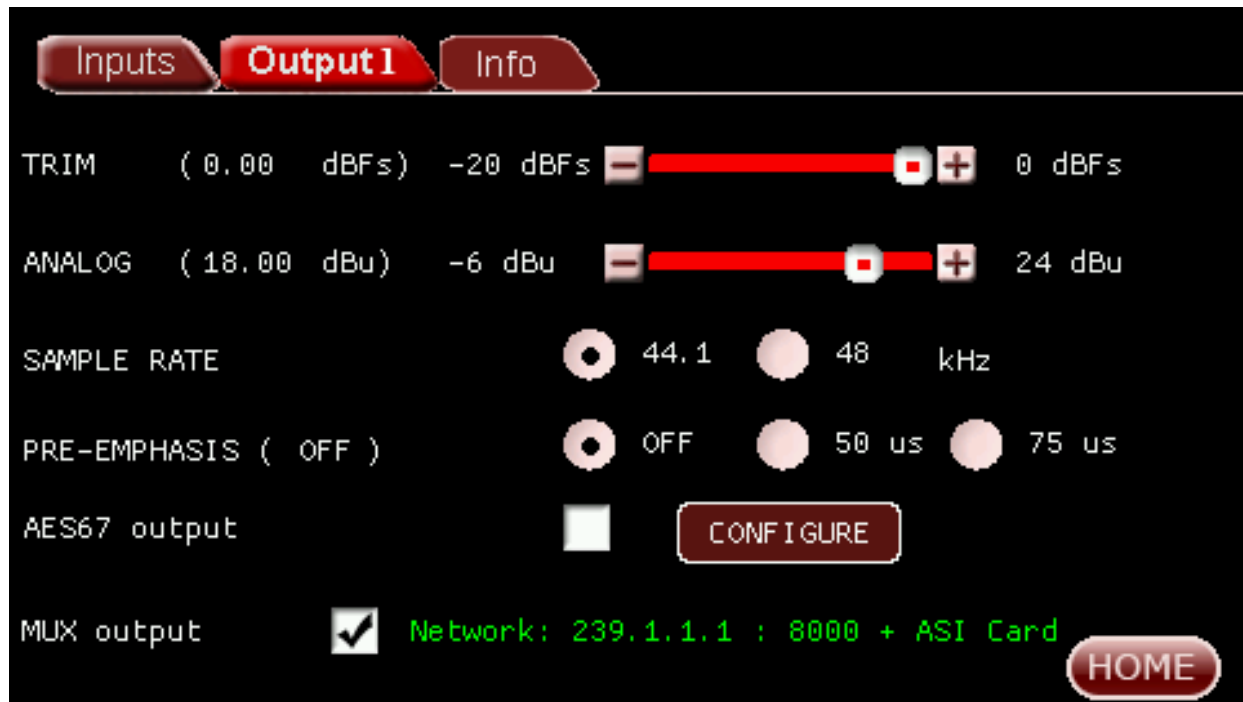


Figure 3.3.10: Output Configuration of Q24-4888 – MUX output to Network and ASI Card

For Q24-4880

The Model Q24-4880 provides AES digital output. The output configuration (as shown in Figures 3.3.11 and 3.3.12 below) is software-selectable, allowing it to be set either as flat or with pre-emphasis at 50 μ s or 75 μ s, depending on system requirements. The digital output can be configured as either a stereo main signal or as an MPX output operating at 192 kHz, providing flexibility for different broadcast and audio processing applications.

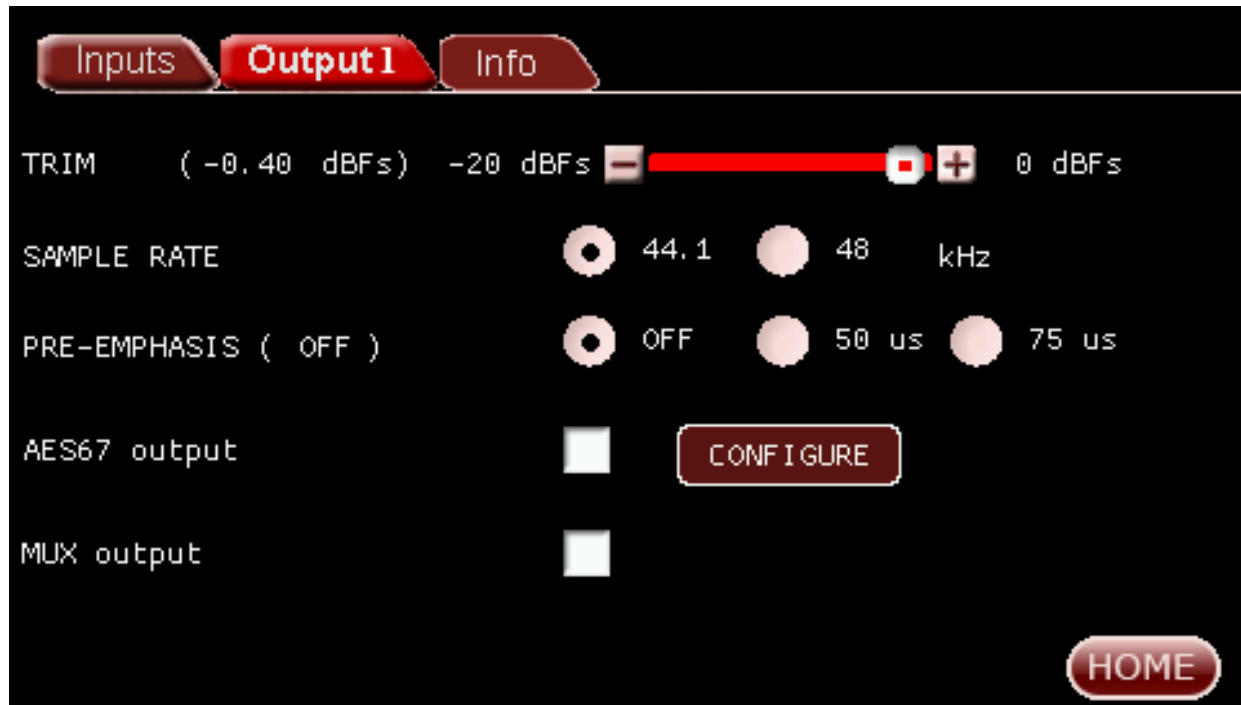


Figure 3.3.11: Output Configurations of Q24-4880

- **Output Level:**
 - Users should calibrate and set output levels by playing a full-scale tone as input.
 - output Trim: Range from -20 dBFS to 0 dBFS. For best results, we set it to -1dBFS.
- **Pre-emphasis:**
 - Provides pre-emphasized digital outputs.
 - Types available: 50 μ s and 75 μ s
- **Output Sample Rate:**
 - Software-selectable options: 44.1 kHz, and 48 kHz
- **AES67 Output:**
 - Enables the AES67 network output of the channel. Press CONFIGURE to open the AES67 Output Settings screen (see Section 3.3.5).
- **MUX Output:**
 - When ticked, the output of the channel is sent in the Multiplexer output (MPEG transport stream, see Section 2.6). The network address and port of the stream are shown next to it (for example Network: 239.1.1.1 : 8000), as shown in Figure 3.3.12.

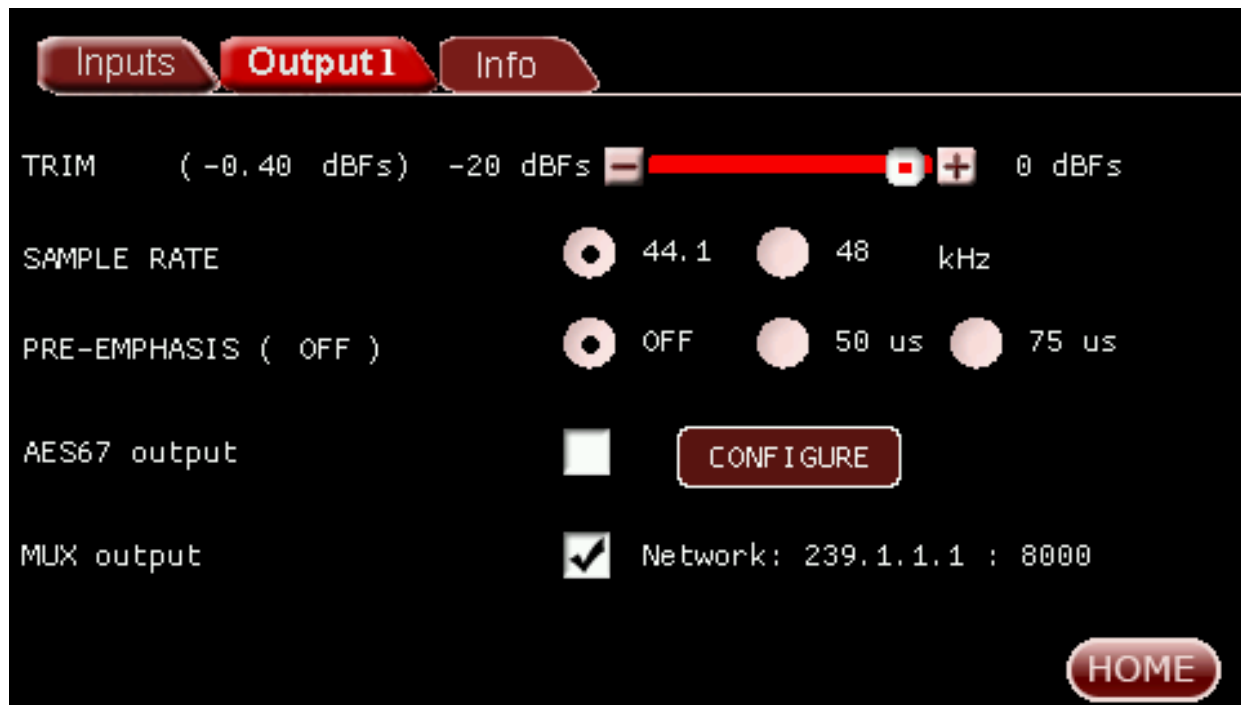


Figure 3.3.12: Output Configuration of Q24-4880 – MUX output ticked

3.3.5 AES67 Output Configuration Screen for Q24-4888/Q24-4880

The AES67 Output Settings screen (as shown in Figure 3.3.13 below) enables the user to configure IP-based audio output streams compliant with AES67 standards. This allows the Q24-4888/Q24-4880 to transmit high-quality uncompressed audio over an IP network for broadcast, studio distribution, and remote monitoring applications.

AES67 OUTPUT SETTINGS

PORT

IP ADDRESS

SAMPLE RATE ☐ 44.1 ☒ 48 kHz

FORMAT ☐ L16 ☒ L24

CHANNELS ☒ STEREO ☐ MONO

PACKETSize ms

BACK

CONFIGURE

SAVE

Figure 3.3.13: AES67 Output Settings

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**
- **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.

3.4 Aideal+ Audio AI Genie Controls

The **Aideal+Audio Processing Genie** is an additional Aideal Audio™ AI-driven tool available in Perceptual SoundMax™ Models Q24-4888 and Q24-4880. 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.

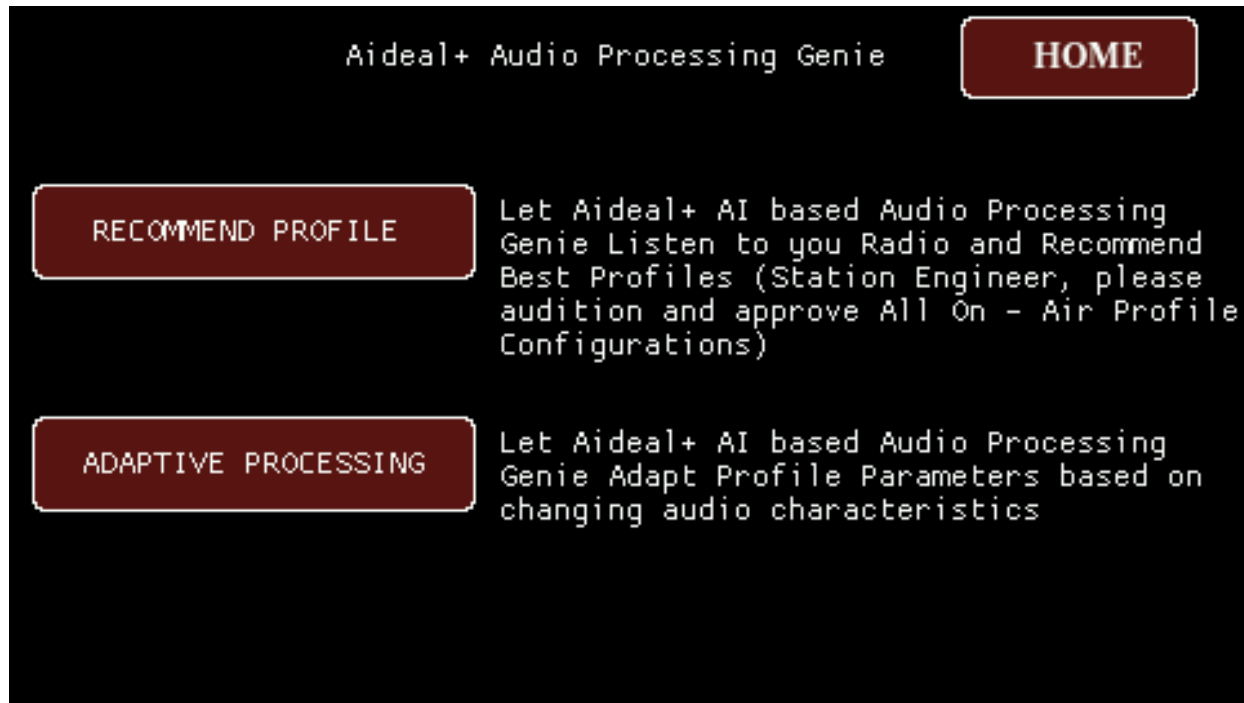


Figure 3.4.1: Aideal+ Audio Processing Genie Main Screen

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.

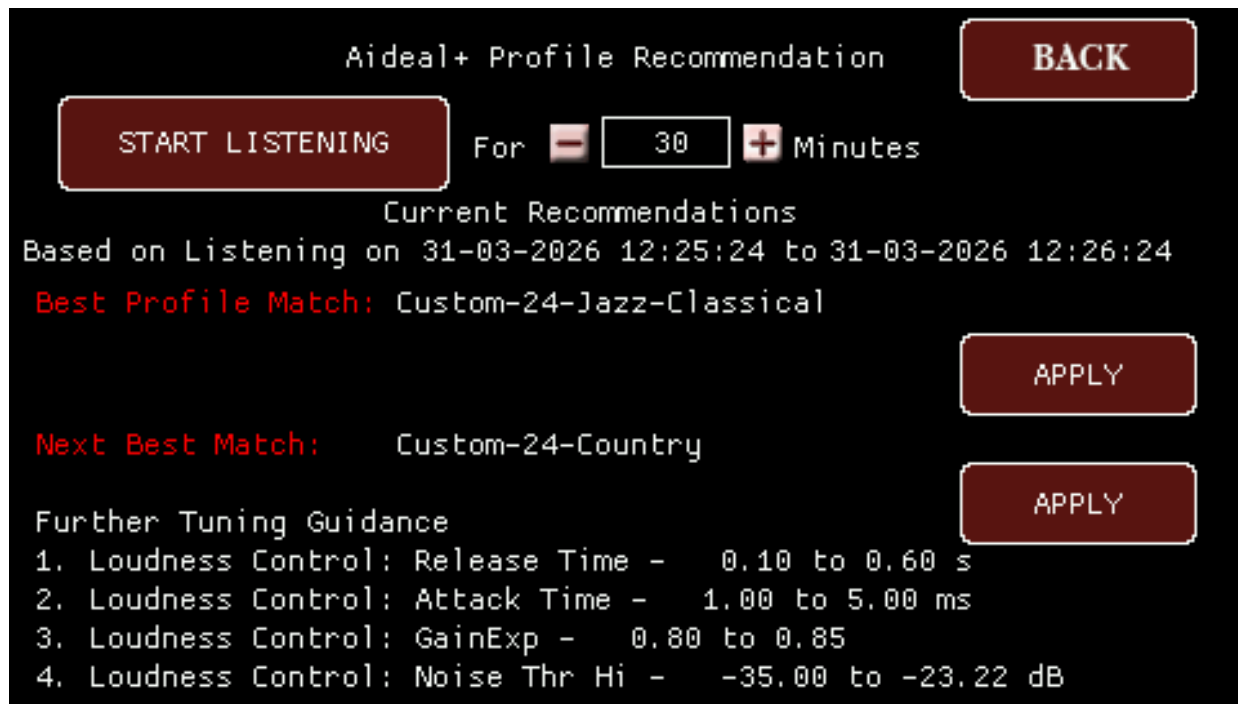


Figure 3.4.3: Aideal+ Profile Recommendation — Results Screen

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.

3.5 Information & Diagnostics Controls

The Info Screen (Figure 3.5.1 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.

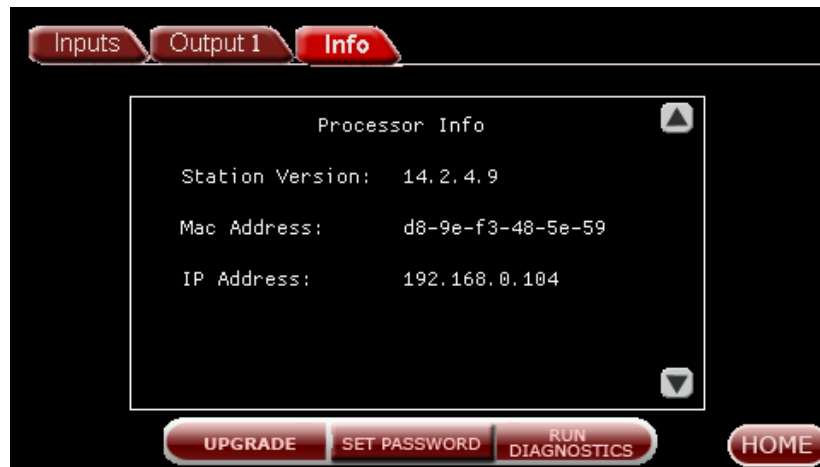


Figure 3.5.1: Audio Settings Info Screen of Q24-4888 and Q24- 4880

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 3.5.2 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 3.5.2: 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 3.5.3 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 3.5.3: Diagnostic Screen

3.6 Processing Parameters Touch-Up

The Tune screen 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.

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

Controls

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 3.6.1: 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.

This section presents a brief overview of the Tuning Screen elements. Detailed control and parameter descriptions are provided in Section 5.3.1.1 – Processing Parameter Touch-Up via TFT Screen.

Section 4 SoundMax™ Manager Controls

SoundMax™ Manager controls can be adjusted from the detailed view in Figures 4.1 and 4.2 below.

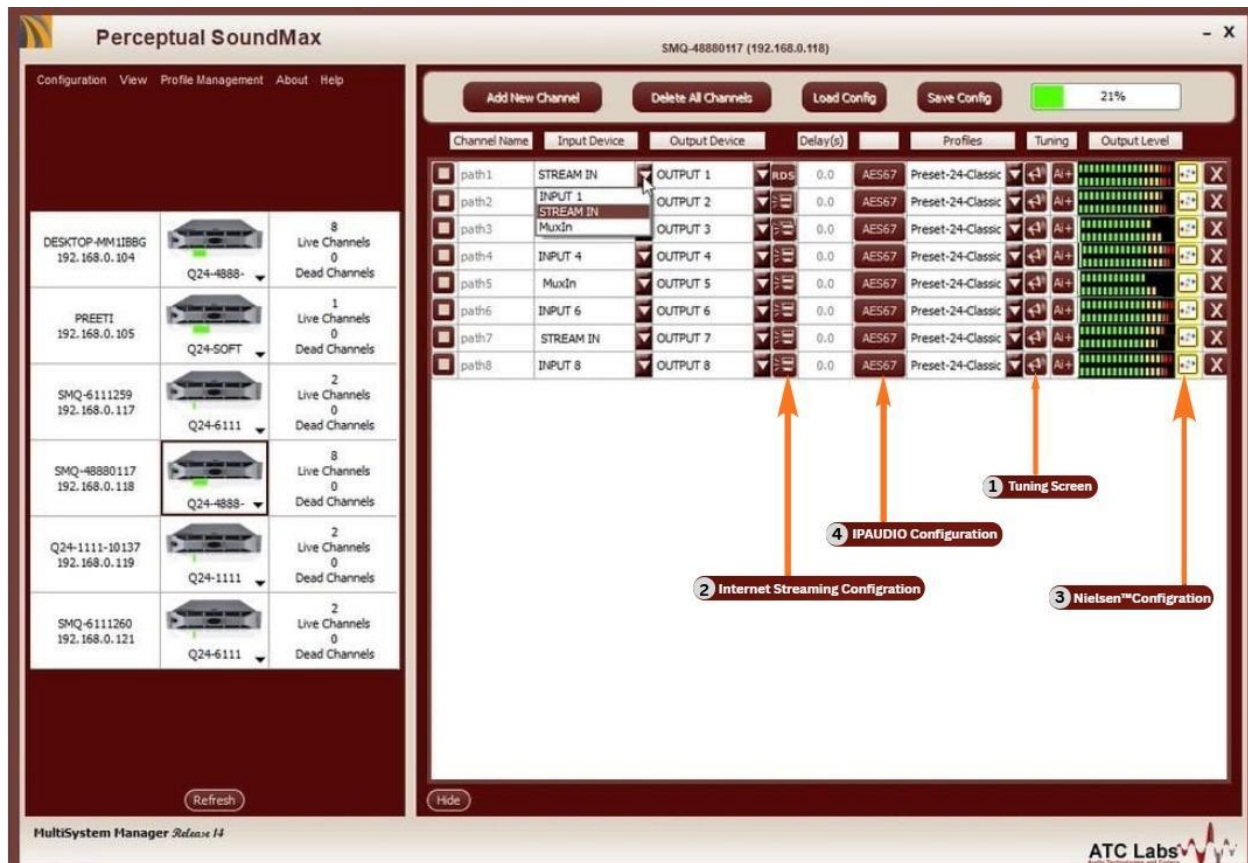


Figure 4.1: Manager Detailed View (Q24-4888) to adjust controls

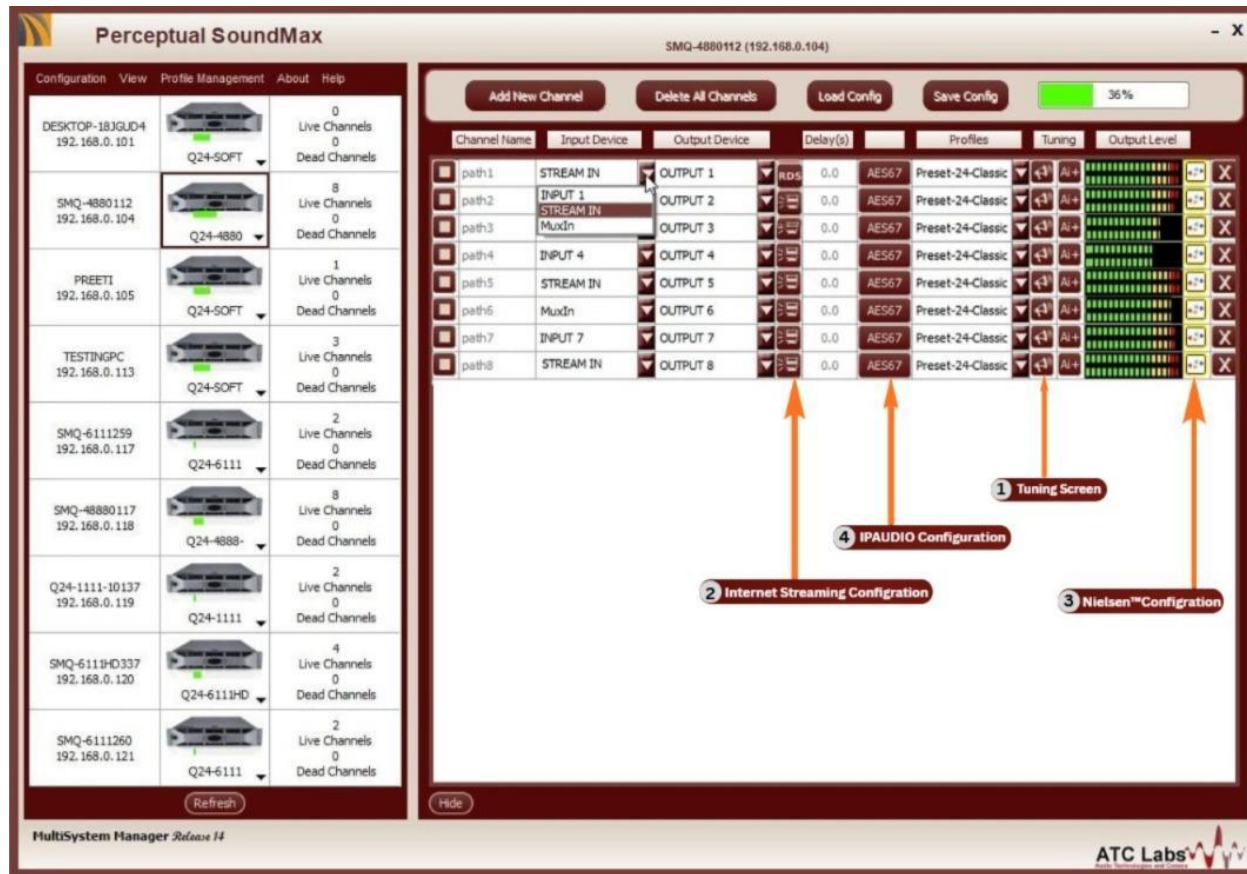


Figure 4.2: Manager Detailed View (Q24-4880) to adjust controls

4.1 Multiplexer and Demultiplexer

The Multiplexer and Demultiplexer of the unit are configured from SoundMax™ Manager: right-click the unit in the server list and choose Multiplexer or Demultiplexer (see Figure 2.6.2).

- **Multiplexer:** combines the processed audio of the selected channels into one MPEG transport stream (MPEG-TS), in which each channel is a programme (service) with its own PID, Service ID, service name and service provider. The stream is sent over the IP network (UDP unicast or multicast) and, on a Q24-4888 fitted with the DekTec DTA-2172 ASI card, to the ASI output.
- **Demultiplexer:** receives an incoming MPEG transport stream from the IP network or the ASI input, scans it and lists its video, audio and data streams. A stream can be passed through unchanged into the Multiplexer output (Passthrough), or an audio stream can be assigned to a channel as its input (Assign to Channel); the input of that channel then becomes MuxIn.

Note: For a detailed description of the Multiplexer and Demultiplexer, their settings and typical procedures, please refer to Section 2.6 Multiplexer and Demultiplexer.

4.2 Tuning Screen for Creating&Fine-TuningProcessing Profiles

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

Figure 4.2.1 below shows the tuning screen where various parameters can be controlled/set.

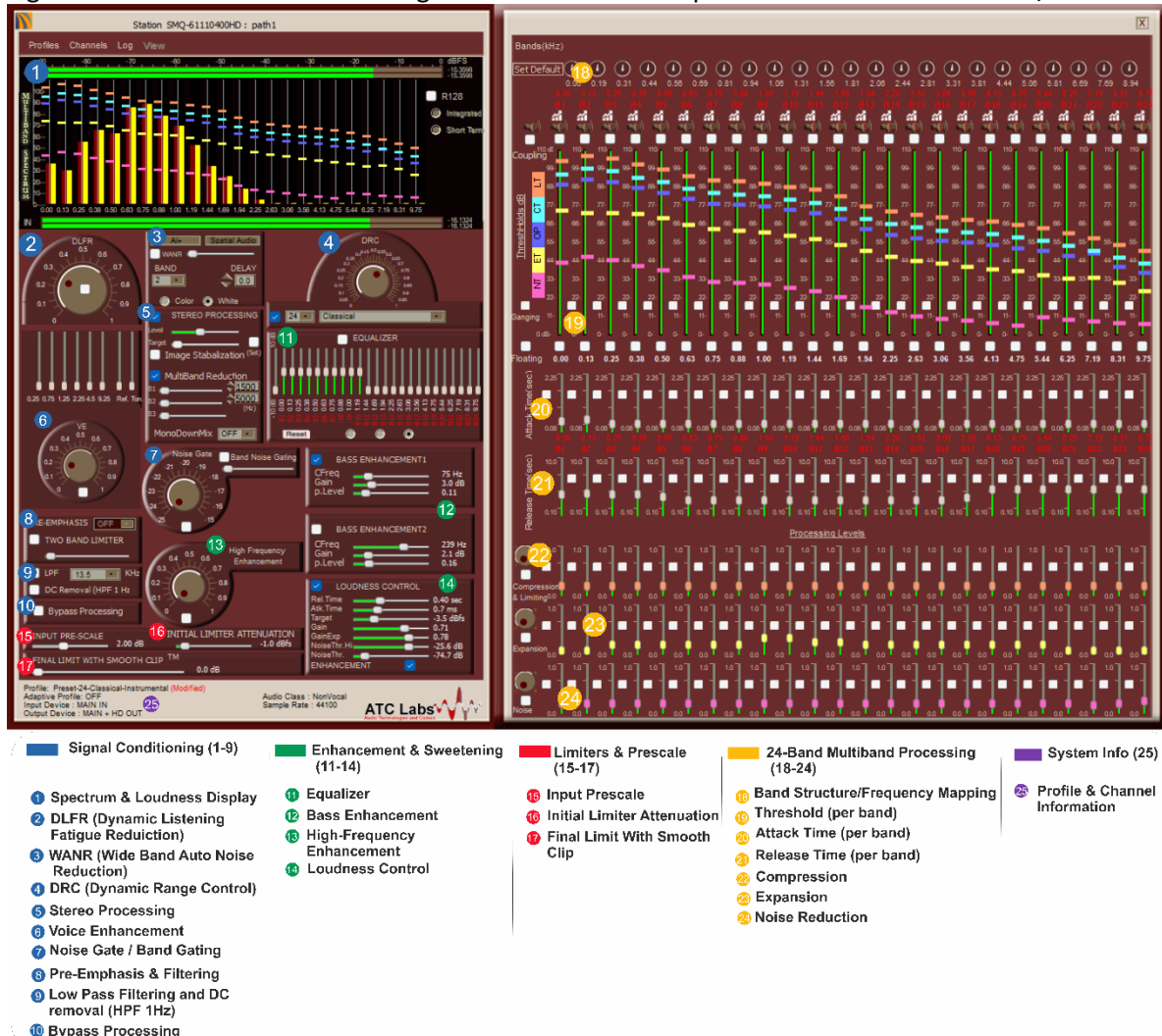


Figure 4.2.1: Tuning screen to control/set various parameters

4.2.1 Factory Preset Profiles

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 once preferences and saved in custom profiles or user defined profiles so that next time the user can directly access and tune into these user defined profiles.

- **CLASSICAL:** Very soft compression to ensure that the high frequencies do not become dense.
- **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.

4.2.2 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 an on-air sound signature. Alternatively, user may want to invoke a processing block which was previously unused. These tasks can be accomplished using either the LCD screen interface or Screen 3 or the SoundMax™ Manager Software as described in Section 5; with the SoundMax™ Manager providing a much higher level of control in terms of tuning the processing parameters as described above. Once a user is satisfied with the sound the resulting profile may then be saved as a custom profile either from the LCD or Controller and available as an option in future. Here is the list of custom and preset profiles:

4.3 Internet Streaming Configuration

Customized internet radio from SoundMax™ gives you a totally new experience in terms of simplicity and quality. Internet streamlining can be controlled from both SoundMax™ manager and the Q24-4888/Q24-4880. Following are the steps of configuration.. Internet streaming can be configured by pressing Button labels ② on the SoundMaxManager™ detailed view panel in Figure 4.1. The panel shown in Figure 4.3.1. 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 4.3.2)

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 4.3.3)
3. AAC/MP3 Distribution via an Icecast distribution server or UDP unicast/multicast over network (with optional TS encapsulation) (Section 4.3.4)

Following are the steps of Stream configuration from this panel.

4.3.1 Stream's Common Settings Panel

The Common Settings tab (Figure 4.3.1) 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.

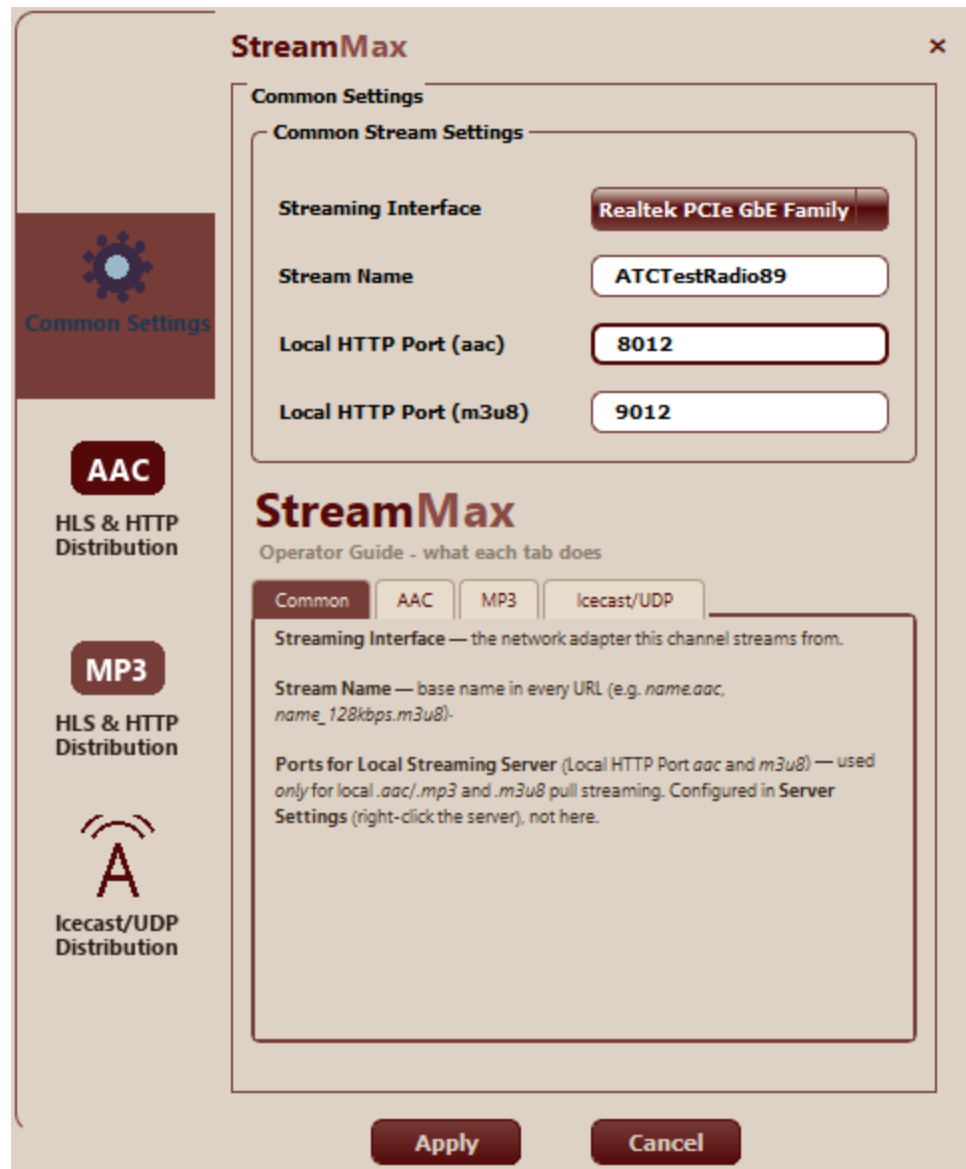


Figure 4.3.1: Common Settings Panel

4.3.2 AAC HLS & HTTP Distribution Settings

The **AAC HLS/HTTP Distribution** tab (Figure 4.3.2) 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, or 256 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).

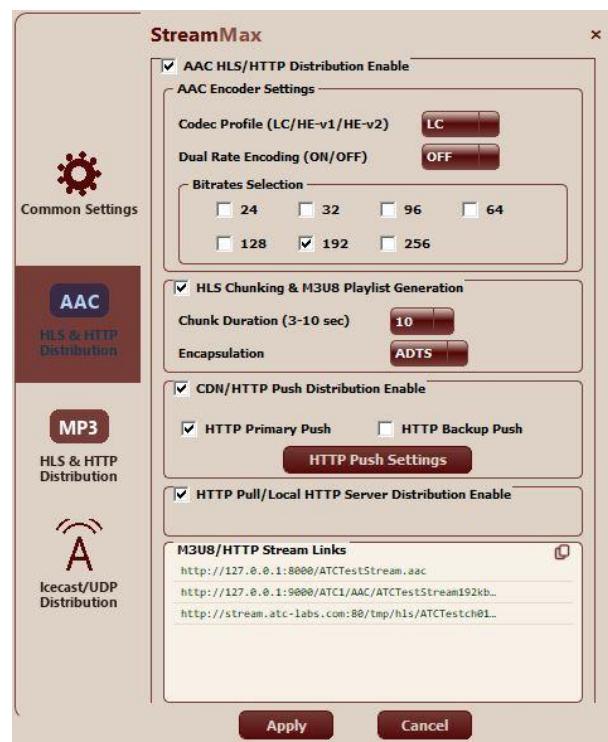


Figure 4.3.2: AAC HLS & HTTP Distribution Settings Panel

(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 Set HTTP Info panel (Figure 4.3.3), where the destination URLs and credentials are configured.

The **Set HTTP Info** panel allows users to configure the technical parameters required for pushing HLS data to remote servers or CDNs.

- **Primary & Backup Publishing Info:** To ensure stream redundancy, the system supports two independent publishing destinations:
 - **Host IP / URL:** The destination server address (e.g., stream.atc-labs.com).
 - **Remote Path:** The directory path on the server where the .m3u8 and audio segments are stored.
- **Common Info (Authentication & Protocol):** These settings apply to both the Primary and Backup connections:
 - **Username / Password:** Credentials required by the remote server for secure access.
 - **Port:** The network port used for the upload (typically 80 for HTTP or 443 for HTTPS).
 - **HTTP Push Type:** POST (standard for pushing data segments) or PUT (typically used for updating the manifest file).
 - **HTTP Auth:** The authentication method used by the server (e.g., BASIC or DIGEST).

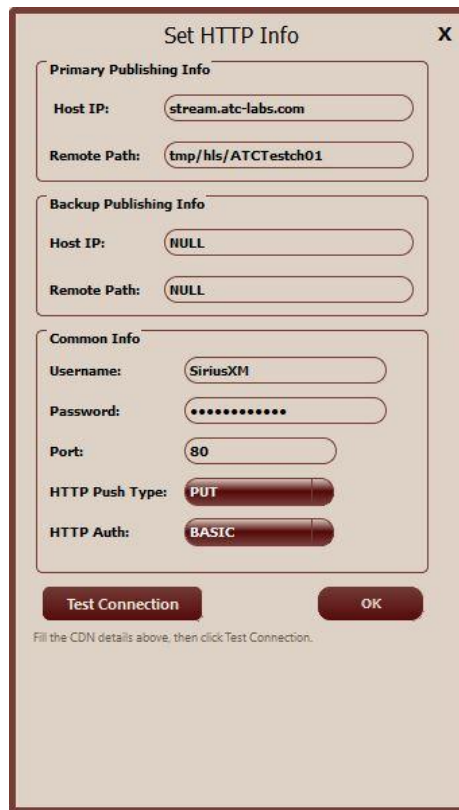


Figure 4.3.3: AAC HTTP Info Settings Panel

Note: If your destination does not require authentication, the Username field may be set to NULL. The Backup Remote Path should usually differ from the Primary path (e.g., add a -b suffix), and ensure the Port matches the destination server's security settings (Port 80 is often blocked by strict firewalls in favour of Port 443).

After entering the CDN details, click **Test Connection** to verify the destination, then **OK** to save the settings.

(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.

4.3.3 MP3 HLS & HTTP Distribution Settings

The **MP3 HLS/HTTP Distribution** tab (Figure 4.3.4) 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.

StreamMax

☒ **MP3 HLS/HTTP Distribution Enable**

MP3 Encoder Settings

Dual Rate Encoding (ON/OFF) **OFF**

Bitrates Selection

☐ 32 ☐ 64 ☐ 96 ☐ 128

☒ 192 ☐ 256 ☐ 320

☒ **HLS Chunking & M3U8 Playlist Generation**

Chunk Duration (3-10 sec) **10**

Encapsulation **MP3**

☒ **CDN/HTTP Push Distribution Enable**

☒ HTTP Primary Push ☐ HTTP Backup Push

HTTP Push Settings

☒ **HTTP Pull/Local HTTP Server Distribution Enable**

M3U8/HTTP Stream Links

http://127.0.0.1:8000/ATCTestStream.mp3

http://127.0.0.1:9000/ATC1/MP3/ATCTestStream192kb...

http://p-ep2035503.1.akamaientrypoint.net:80/2035...

Apply **Cancel**

Figure 4.3.4: 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, a higher-fidelity option than the standard AAC profile limits.

(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 4.3.3).

(D) HTTP Pull / Local HTTP Server Distribution Enable

Distributes the MP3 HLS playlist and stream from the Q24Soft server'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.

4.3.4 Icecast / UDP Distribution Settings

The **Icecast/UDP Distribution** tab (Figure 4.3.5) 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.

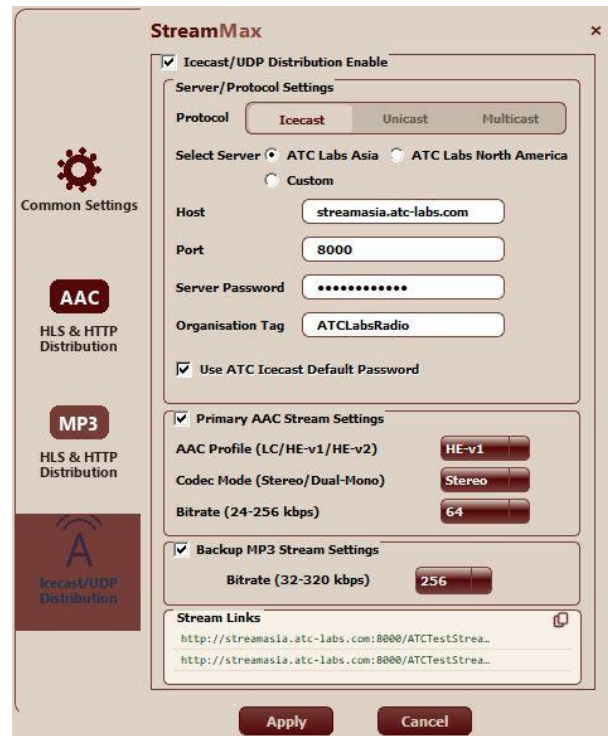


Figure 4.3.5: Icecast / UDP Distribution Panel

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 (24–256 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.

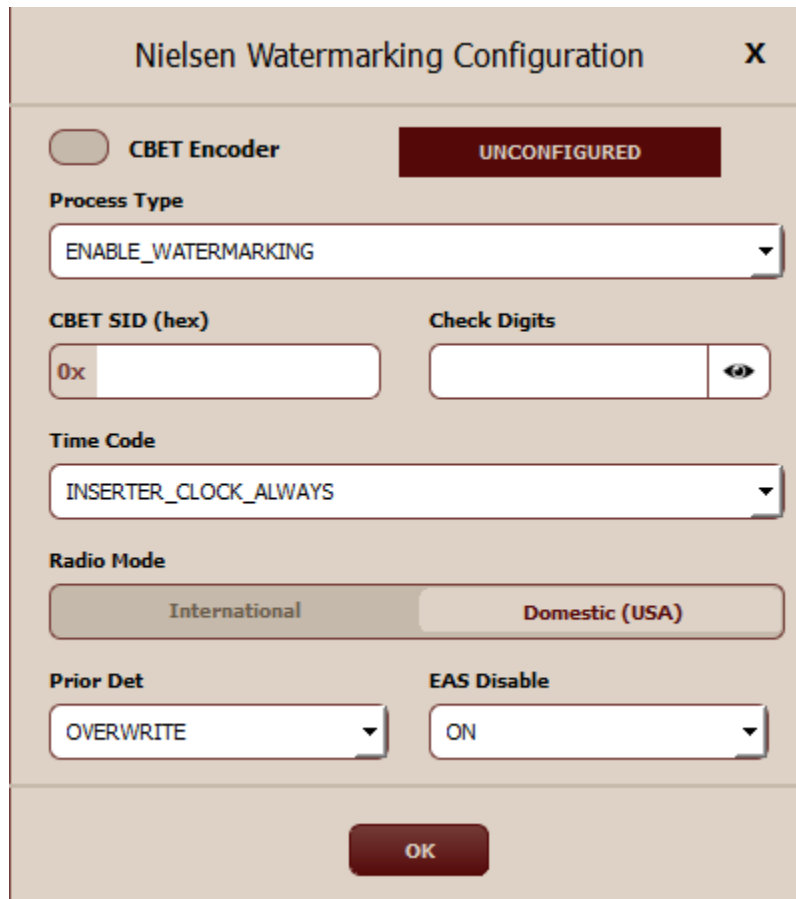
Stream Links

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

4.4 Nielsen™ Configuration

Perceptual SoundMax™ supports fully certified in-box Nielsen™ watermarking. The Nielsen™ Configuration Panel (Figure 4.4.1) is accessed via the SoundMaxManager™ detailed view (button labelled as ❸ 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 Watermarking Configuration' dialog box. At the top, there's a title bar with the text 'Nielsen Watermarking Configuration' and a close button 'X'. Below the title bar, there's a section for 'CBET Encoder' with a toggle switch and a status label 'UNCONFIGURED'. The 'Process Type' is set to 'ENABLE_WATERMARKING' in a dropdown menu. The 'CBET SID (hex)' field contains '0x' followed by an empty input box. The 'Check Digits' field is empty with a toggle icon. The 'Time Code' is set to 'INSERTER_CLOCK_ALWAYS' in a dropdown menu. The 'Radio Mode' has two buttons: 'International' and 'Domestic (USA)'. The 'Prior Det' is set to 'OVERWRITE' in a dropdown menu. The 'EAS Disable' is set to 'ON' in a dropdown menu. At the bottom, there is an 'OK' button.

Figure 4.4.1: 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.

4.5 RDS Configuration for Q24-4888

The Internal RDS Encoder Panel (Figure 4.5.1) is used to configure the Radio Data System (RDS) information. This allows you to transmit digital data such as station names and song titles alongside your audio, which is displayed on compatible FM receivers and digital players.

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

- Programme Service Configuration:**
 - PS Type: A dropdown menu set to "STREET_NAME".
 - Program Service (PS): A text input field containing "ATCRadio".
 - Send Rate (secs): A dropdown menu set to ".5".
- Radio Text Configuration:**
 - Static Radio Text: A text input field containing "TCRadio: FM Stereo and RDS encoder".
 - Dynamic Radio Text: Three radio buttons labeled "Overwrite", "Append", and "Disabled". The "Overwrite" button is selected.
- 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 window.

Figure 4.5.1: Internal RDS Encoder Panel

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

Programme Service (PS) Configuration

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

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

Radio Text (RT) Configuration

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

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

RDS Data Input Configuration

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

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

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

4.6 AES67 IP AUDIO Configuration

The AES67 IP Audio functionality is also fully configurable via the SoundMaxManager™ detailed view (button labelled as AES in Figure 4.1). The AES67 Configuration Panel (Figure 4.6.1) 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 4.6.1) enables high-performance Audio-over-IP (AoIP) interoperability. This allows the system to send and receive uncompressed, low-latency audio across a local network.

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) and a tab labeled "AOIP". The window is divided into two main sections: "AES67 Input Settings" and "AES67 Output Settings", both of which are checked with checkboxes.

AES67 Input Settings:

- IPAddress:** 192.168.1.56
- Port:** 5004
- SampleRate:** 48000
- Format:** L24
- Channels:** Stereo
- Capture:** Socket (IGMP)
- Source IP:** (empty field)

AES67 Output Settings:

- IPAddress:** 192.168.5.25
- Port:** 5004
- SampleRate:** 48000
- Format:** L24
- Channels:** Stereo
- PacketSize:** 1.0 ms

At the bottom, there is a label "AoIP NIC (in+out):" followed by a dropdown menu set to "Auto (streaming NIC)". An "OK" button is located at the very bottom center.

Figure 4.6.1: AES67 Configuration Panel

4.7 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  in Figure 4.1), which opens the Aideal+ Audio Processing Genie window (Figures 4.7.1 and 4.7.2).

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 4.7.1). 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.

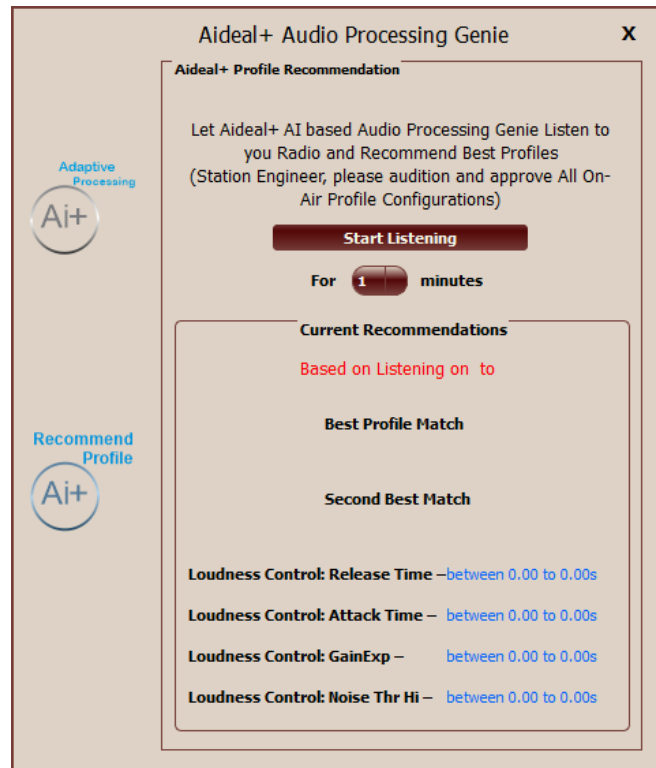


Figure 4.7.1: Aideal+ Profile Recommendation

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 4.7.2). 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."

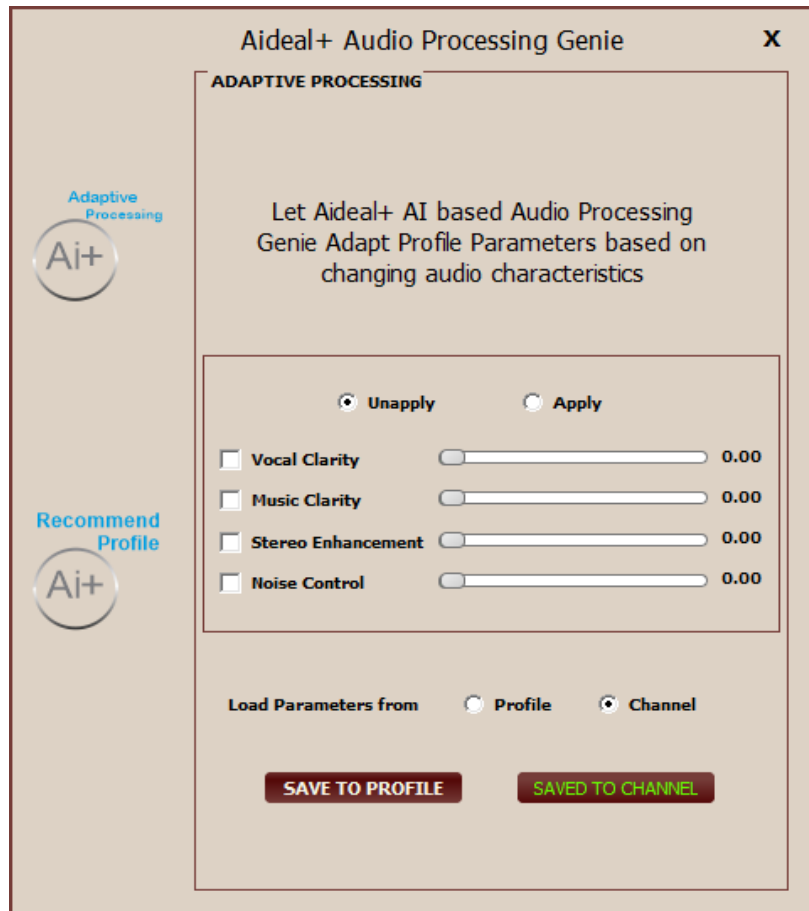


Figure 4.7.2: Aideal+ Adaptive Processing

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.

Section 5 Algorithmic Insights & Processing Optimizations

5.1 Audio Processing Architecture & Operation

This section discusses the audio processing architecture and operation.

5.1.1 Processing Architecture Highlights

Perceptual SoundMax™ release V12 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. Parameters in all 24 bands 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. in digital broadcasting the processing may be matched to the unique codec characteristics/impairments in multiple frequency bands. In general, more bands allow for *lighter processing* to achieve a target impact level since each band affects a smaller portion of audio.

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

Integrated Compression/Limiting/Expansion/Noise Gating: Perceptual SoundMax™ implements all envelope processing (i.e. expansion/AGC and compression/limiting) in each of the 24 bands in a single stage. This avoids a situation where different processing stages are fighting and/or negating each other's impact - a situation common with many other processing technologies, which typically implement multi-band compression and multi-band AGC as two separate modules. The *single-stage* envelope non-linearity for each band in Perceptual SoundMax™ is illustrated in Figure 5.1.1 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 band are fully configurable. Moreover, the level of processing in each of the regions is also independently controllable.

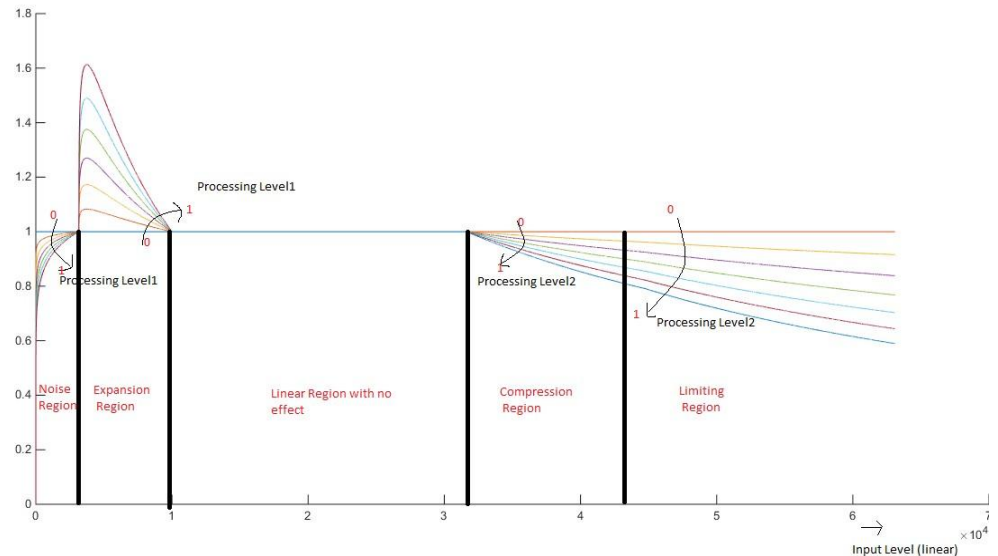


Figure 5.1.1: The configurable shape of Perceptual SoundMax™ Envelope Processing non-linearity in multiple regions

Advanced Final Look Ahead Limiting with SmoothClip™:proprietary smooth windowing technology is used to minimize spectral growth (increase in high frequency energy when processing leads to saturation). This is particularly helpful for digital broadcasting. Figure 5.1.2 (a) below shows the audio spectrogram for an AGC with conventional look ahead limiting, high frequency growth is readily obvious at the top. Figure 5.1.2 (b) shows the impact of technique used in Perceptual SoundMax™ (with much less spectral growth).

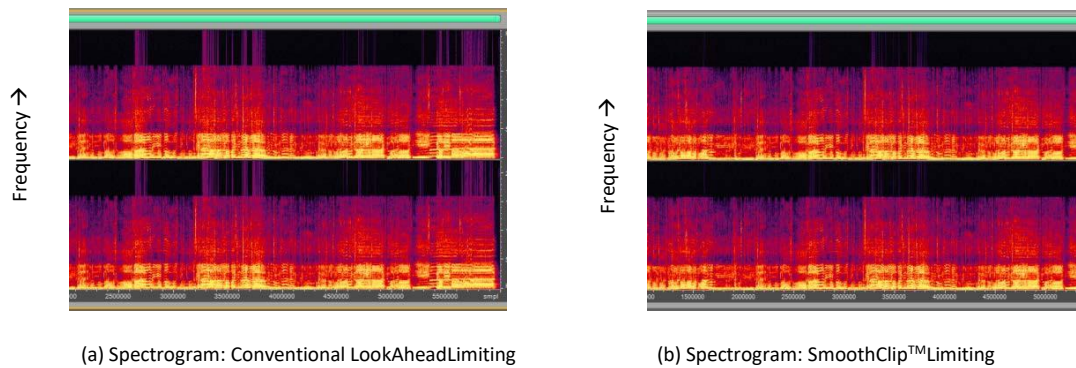


Figure 5.1.2: SmoothClip™ technology in Perceptual SoundMax™ substantially limits increase in high frequency energy during look ahead limiting

Extensive use of AI Based Audio Processing Algorithms: PerceptualSoundMax™ utilizes AI/ML based techniques throughout its processing platform. Techniques included sophisticated models for multiband noise analysis, signal tilt analysis, and our patent pending *Real-Time Multi-Class Hierarchical Audio Classification Deep ML model* that can be used to adaptively adjust processing parameters based on changing audio characteristics.

5.1.2 Processing Architecture: *SignalFlow*

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

The Processing Architecture Signal Flow is shown in Figure 5.1.3 below.

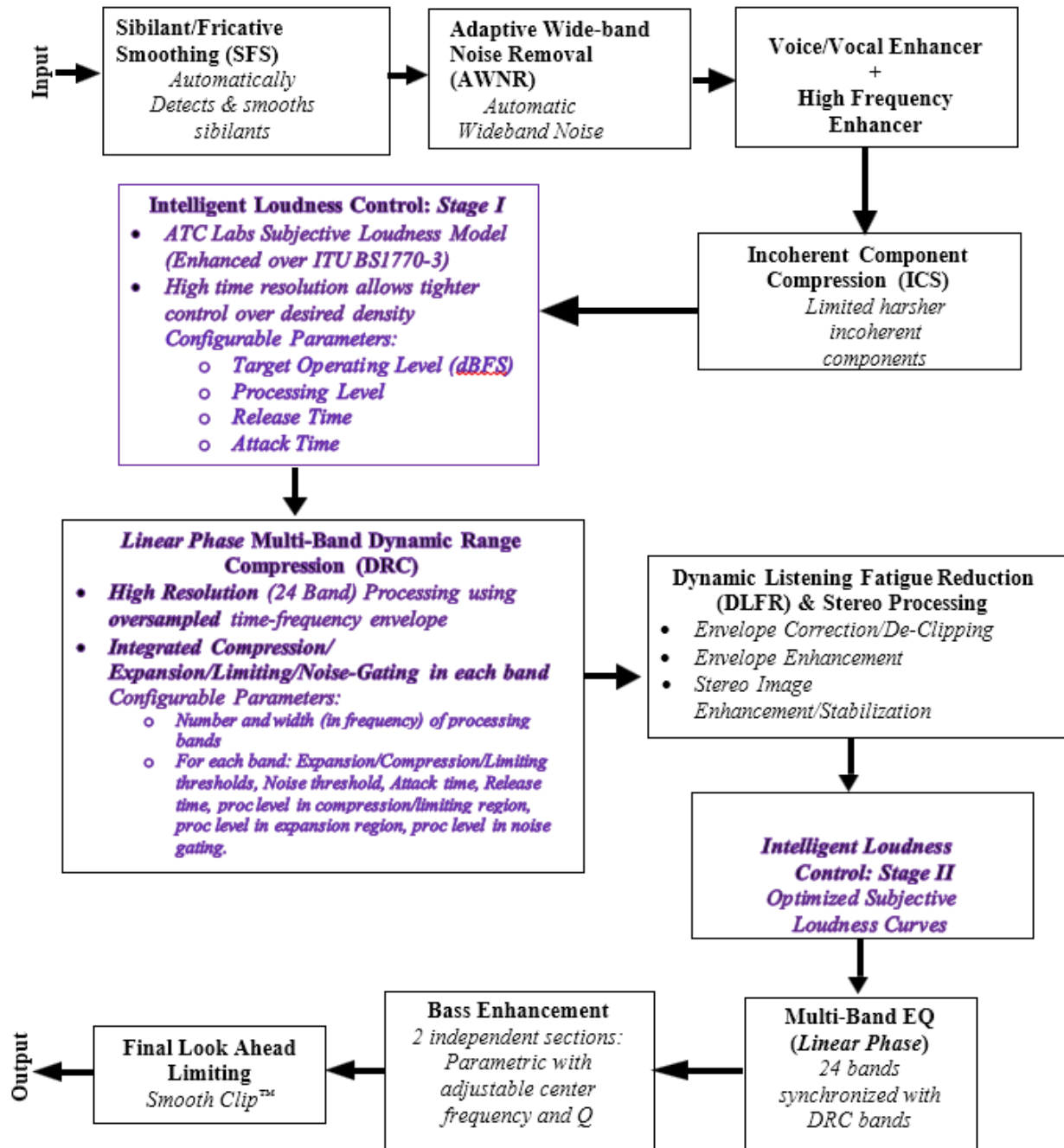


Figure 5.1.3: Processing Architecture – Signal Flow

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

(Note: These fine tuning are best performed using SoundMax™ Manger Tuning Screen Controls; although limited tuning is also possible using the TFT screen as noted in the description below)

We offers two levels of tuning controls, namely Basic and Advanced, each described in detail below.

5.2.1 Basic Tuning Guide

This section outlines the specialized controls available for shaping the characteristics and dynamic impact of audio. Each slider serves as a macro-control, intelligently adjusting multiple internal parameters—such as attack and release times, thresholds, and gain—to provide cohesive results with a single movement.

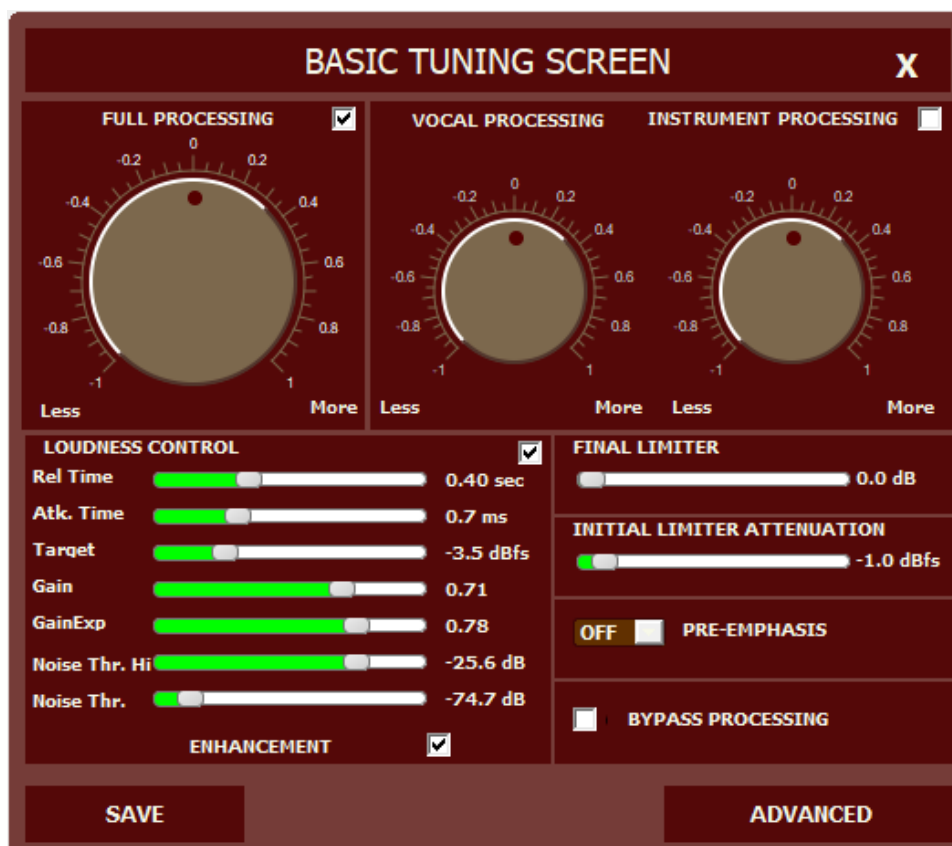


Fig 5.2.1: Basic Tuning screen

Full Processing

This master slider tunes the entire audio spectrum simultaneously. It is designed to normalize and enhance audio quality across every frequency band in a synchronized manner.

- **Increasing the Slider (Energetic/Dense)**

- **Behaviour:** Reduces attack and release times while increasing thresholds and gain across all bands.
- **Result:** Produces a "densely normalized" sound with controlled attacks and a significant boost in signal density.
- **Recommended For:** High-energy, fast-tempo genres such as Club, Hip-Hop, Rap, and Disco.
- **Decreasing the Slider (Relaxed/Natural)**
 - **Behaviour:** 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.

Vocal Processing

This control specifically targets the vocal frequency range within the audio spectrum, allowing you to manage the presence and loudness of vocals independently of the background music.

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

Instrumental Processing

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

- **Increase:** Enhances the loudness and definition of the music. This makes the instrumental elements feel more "forward" and powerful compared to the vocals or dialogue.
- **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. 5.2.1, refer to Section 5.2.2 (Advanced Tuning Guide). These sliders and checkboxes function in the same way as those in the Advanced Tuning screen.

5.2.2 Advanced Tuning Guide

This section provides an overview of the various audio processing tools available in the system and guidance on how to use them effectively to achieve high-quality output audio. The Perceptual SoundMax™ Manager Tuning Screen is shown in Figure 5.2.2. The system offers multiple processing modules which can be adjusted individually or in combination to create a desired sound signature.

As described above following different processing modules are available:

- AI based processing control.
- Intelligent Loudness Control
- Multi-Band Dynamic Range Compression (DRC) – 24 Bands
- Base Enhancement – 2 Sections
- Multi-Band Graphic Equalizer (24 bands)
- Vocal Enhancement (VE)
- Stereo Processing

- Incoherent Component Suppression (ICS)
- Sibilant/Fricative Smoothing (SFS)
- Wideband Automatic Noise Removal (WANR)
- Dynamic Listening Fatigue Reduction (DLFR)
- Pre-Emphasis
- Low Pass Filter
- Initial & Final Look Ahead Limiting
- Input Pre-scale

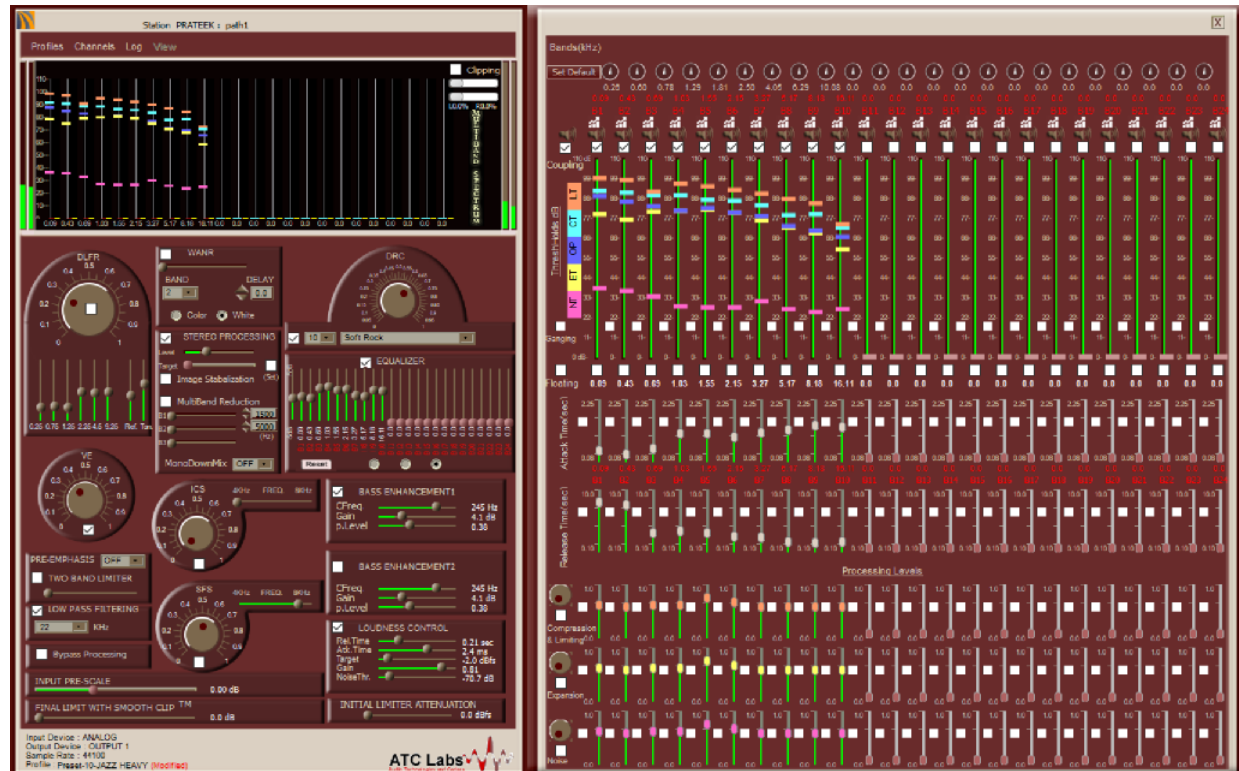


Figure 5.2.2: Perceptual SoundMax™ Manager Tuning Screen

These different tools when tuned judiciously can provide consistent, clear, pleasant, high impact audio and offer the broadcast engineer flexibility for creating a compelling signature sound for each service channel and bit rates.

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

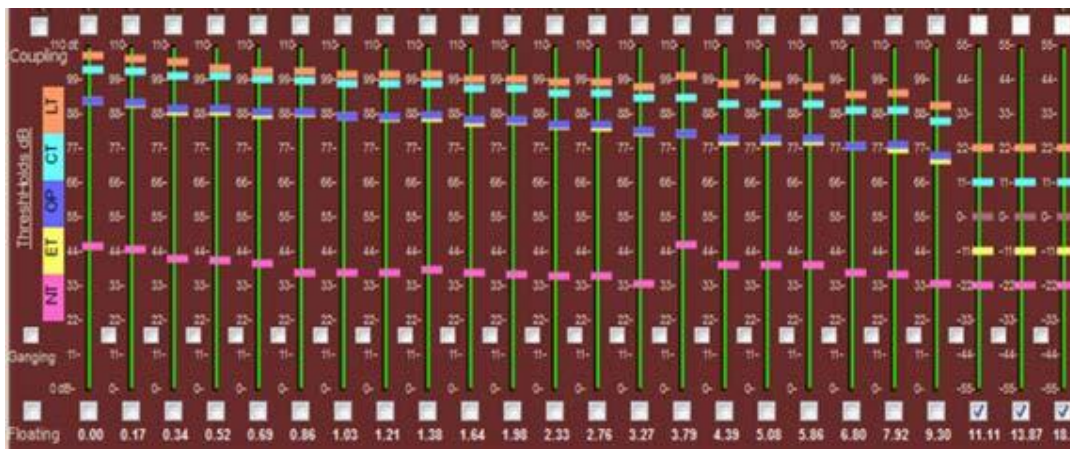


Multiband Dynamic Range Compression (DRC) is an integrated 24-band Dynamics Processor allowing for compression, limiting, expansion, and noise thresholding in 24 distinct frequency bands. Driven by the principles of Psychoacoustics, it employs completely linear phase signal processing. Intelligent control 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

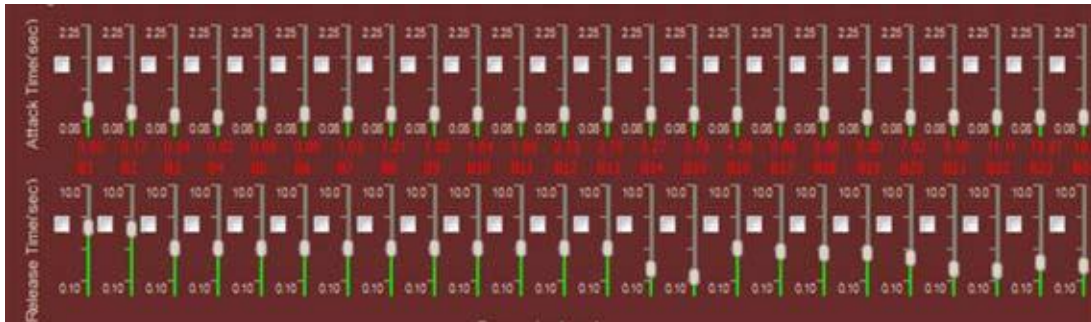
bank for generating an extremely detailed time-frequency representation of the signal. Available controls in DRC include:

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



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

- **Release Time and Attack Time in each of the bands:** as in the ILC module, Release Time is key parameters 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 make the audio more controlled and dense 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 control the level of dynamics in a band (a higher value results in more dynamics). As with the thresholds, the interface allows the ganging of multiple bands for convenient tuning.



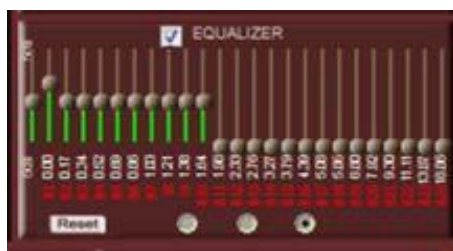
- **Processing Levels:** The new V11 interface allows for independent control of processing levels of DRC for each band, in three regions of non-linearity (Figure 5.1.1): (i) compression/limiting, (ii) expansion, (iii) noise-gating. Correspondingly there are 24 sliders for compression/limiting processing level, 24 sliders expansion processing level, and 24 sliders for noise gating processing levels. The interface allows for ganging of bands and also offers a master dial for each of the 3 set of sliders. Also, changing the master DRC dial impact all processing levels equally (with saturation of individual values for maximum/minimum).



- **Center Frequencies of the processing bands:** The new V11 interface allows for the center frequency of each band to be increased or decreased using a set of dial (with the constraint that center frequency of each band is upper/lower bounded by respectively the adjacent higher/lower band).

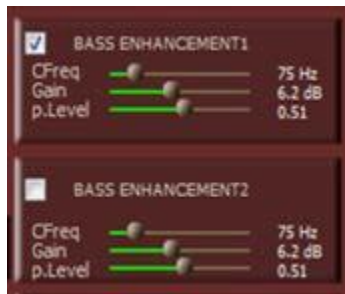


• Multi-Band Graphic Equalizer:



- This module equalizes the audio in 24 bands according to the 24 slider setting. If a slider is set to the middle of the range the equalizer is effectively flat in that band. Each slider can be moved between -5dB and 5dB. There are three options in the form of Soft keys, namely Tight Coupling, Loose Coupling and No Coupling, supported for coupling the sliders. The center frequency of the equalizer bands is aligned with the center frequency of the DRC bands B1-B24.

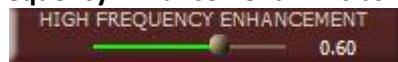
- **Bass Enhancement:**



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

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

- **High Frequency Enhancement:** This control enhances the higher frequencies and provides the 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:

- **Image Stabilization:** if this box is checked, stereo image is also expanded else the stereo image processing is purely compressive. Therefore, if this feature is invoked the stereo image may at time be wider than the original audio and remains more stable, losing some of its dynamics.
- **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:** Set any limit between 4-20kHz



- **Incoherent Component Suppression (ICS):**



This module detects the incoherent components of the input audio and smoothens the corresponding audio component resulting in a sound that is easier to process by the human auditory perception system, thus increasing the listenability of the sound. The slider controls for this module include a Start Frequency, which specifies the frequency threshold above which the processing is done; this can be varied from 4 kHz to 8 kHz. The dial controls the intensity of processing done by the algorithm. A new user can safely set the dial at 0.2 and the start frequency at about 6-8 kHz. **THIS MODULE IS NOW USED AS A MASTER GATING MODULE.**

- **Pre-Emphasis:** Use Pre-Emphasis along with the available 2-band limiter to maximize your modulation index and loudness for FM broadcasts.



- **Final Limiting Volume:** This sets final level for the look ahead limiter. For digital broadcasts 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 set to control the 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. However, in digital broadcast application scenarios where codecs exhibit extreme sensitivity to increase in high frequencies the algorithm may be aided by the application of an initial limiter attenuation of -1 to -2 dB using this available additional control.



- **Bypass Processing:**
By checking the bypass processing button the processing done in the Q24-1111 Soft 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):**



Model Q24-1111 comes bundled with a world-class real-time automatic noise control block. 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 two different structures for the noise model:

- **White Model:** For general noise types.
- **Color Model:** For heavy noise reduction of a particular environment such as machine and instruments noise type.

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

This module enhances the speech/vocal content of the audio. It uses an accurate harmonic analysis of the input audio and effectively enhances the vocal content. Using this information input audio sounds



cleaner with improved presence. This module is also useful in sports/news broadcast to make the commentators voice more intelligible, where high background audience noise exist.

- **Envelope 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.



This tool helps in maintaining a consistent stereo image which would make audio sound pleasant. A new user can safely set the dial at 0.3.

- **Sibilant Fricative Smoothing (SFS):**

This module dynamically detects the sibilant and fricative sections of the input audio and smoothes



out these sections. The slider controls include the start Frequency, which specifies the frequency threshold above which the processing is done; this can be varied from 4 kHz to 8 kHz. The dial controls the intensity of processing done by the algorithm. A new user can safely set the dial at 0.2

and the start frequency at about 6-8 kHz. Sibilants tend to add to some of the unwanted harshnesses in the broadcast audio since the perceived loudness of sibilants is quite high due to their high-frequency content.

- **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.

- **Incoherent Component Suppression (ICS):**

This module detects the incoherent components of the input audio and smooths the corresponding



audio component resulting in a sound that is easier to process by the human auditory perception system, thus increasing the listening ability of the sound. The slider controls for this module include a Start Frequency, which specifies the frequency threshold above which the processing is done; this can be varied from 4 kHz to 8 kHz. The dial

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.

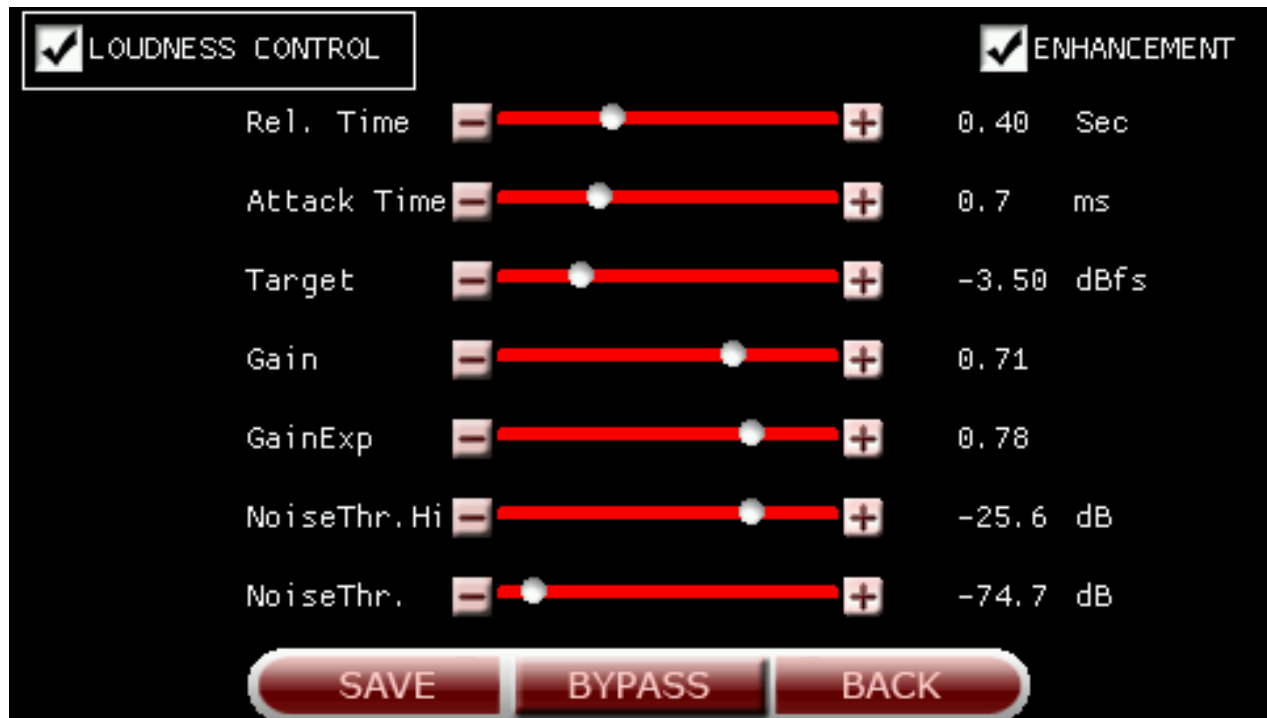


Figure 5.3.3: Loudness Control Screen

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 (e.g., 0.71).
- **GainExp** — Gain exponent that shapes the loudness curve response (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 (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.

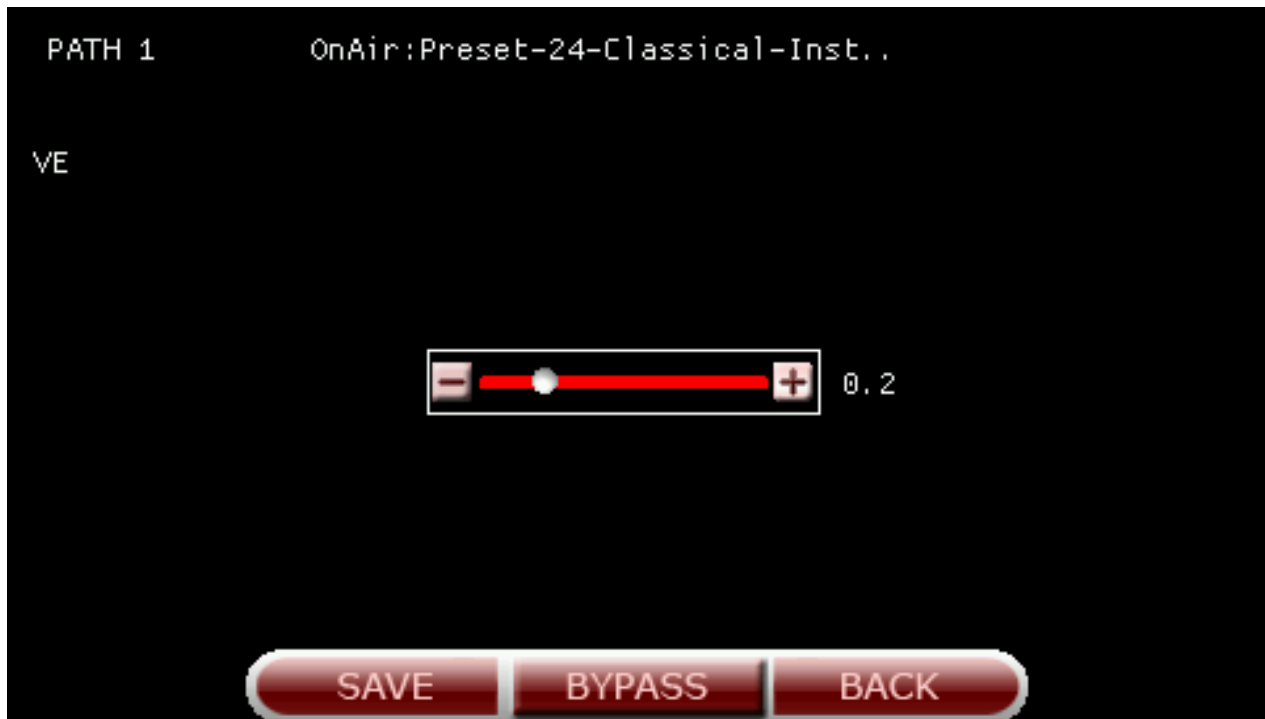


Figure 5.3.4: Voice Enhancement Screen

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.

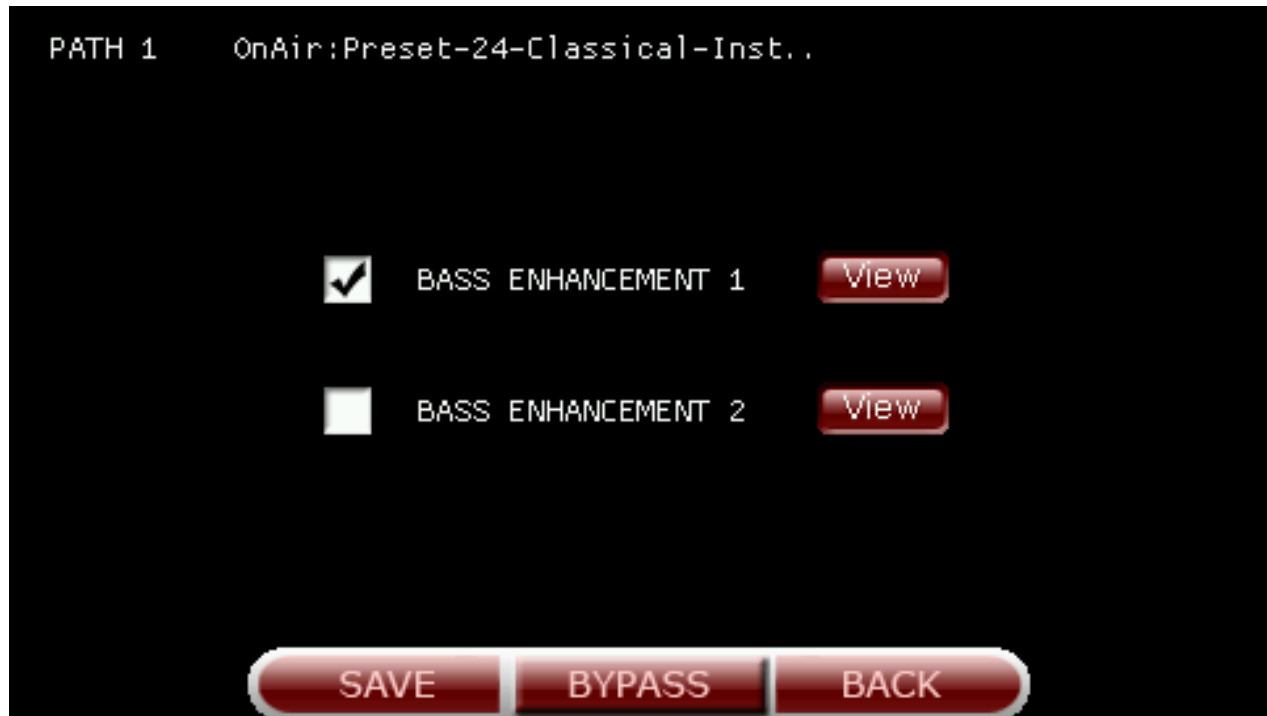


Figure 5.3.5: Bass Enhancement Screen

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.

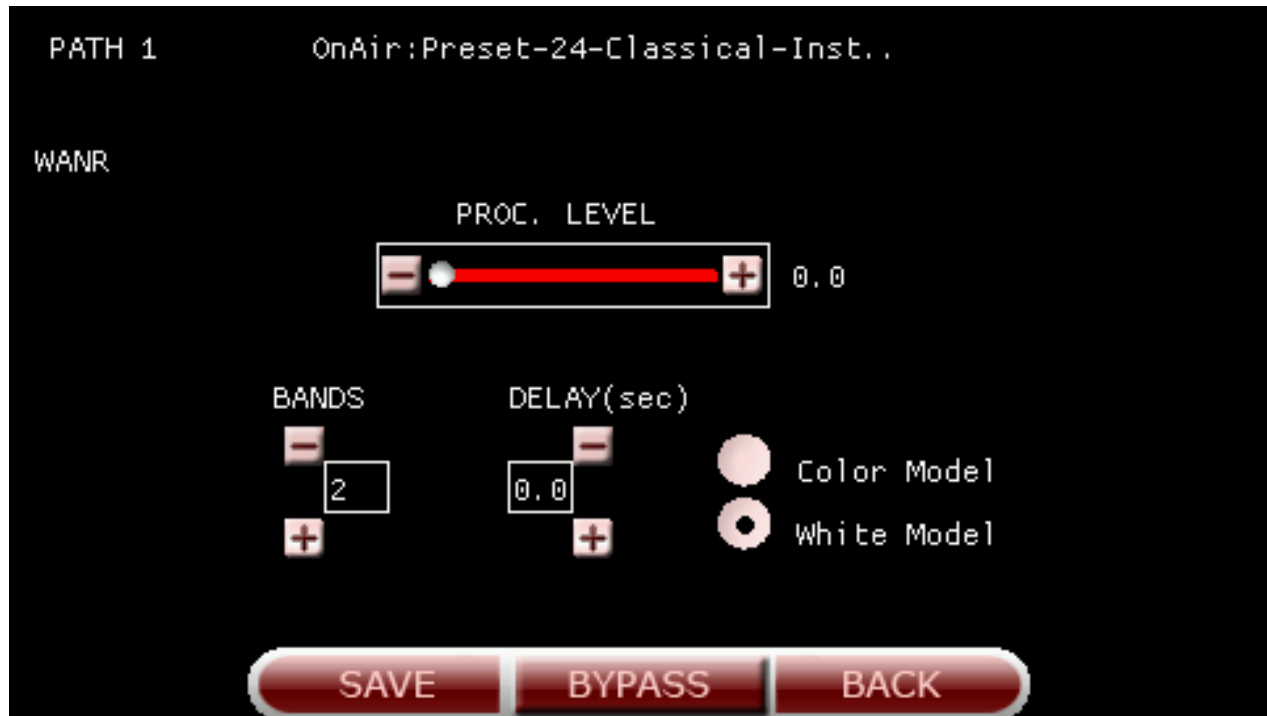


Figure 5.3.6: Wideband Auto Noise Reduction Screen

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.

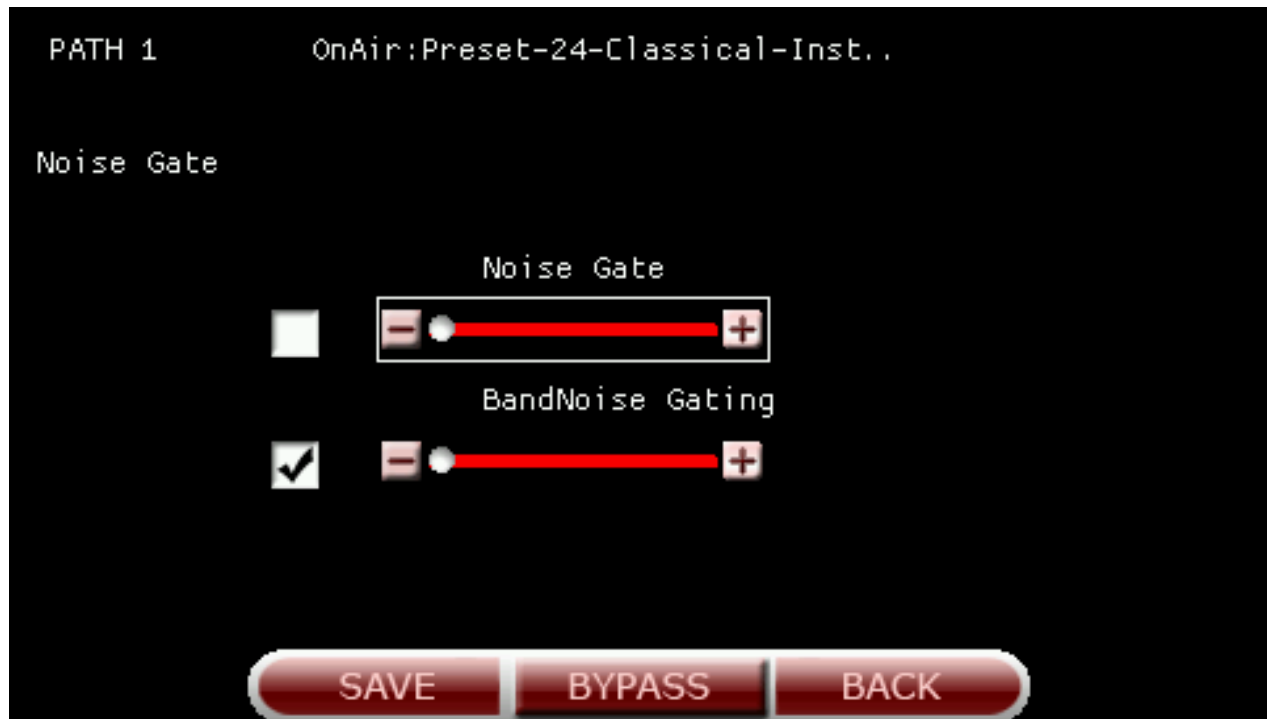


Figure 5.3.7: Noise Gate Screen

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 attenuates the output to suppress noise during silent passages.
- **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 first-stage limiter, which provides initial peak control before the audio enters the main multiband processing engine.

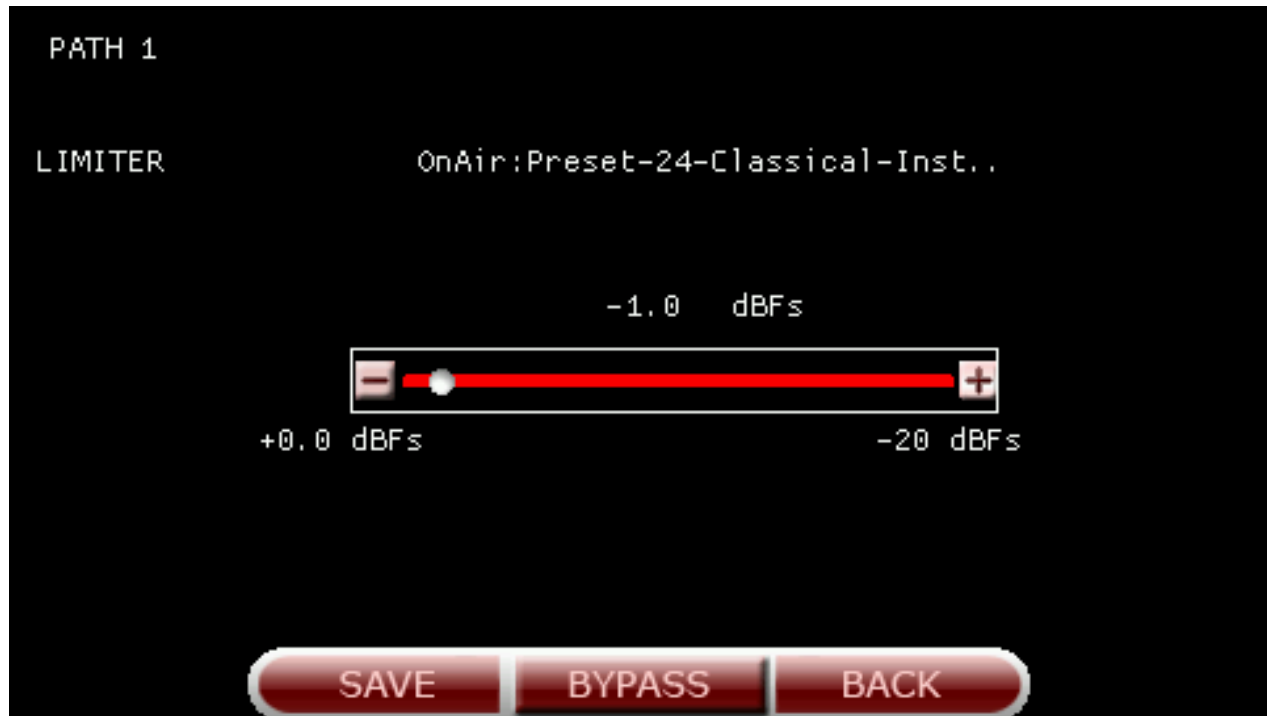


Figure 5.3.8: Initial Limiter Screen

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.



Figure 5.3.9: Two Band Limiter Screen

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

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

Not controlled by TFT Screen. Use SoundMax Manager to control DRC.

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.

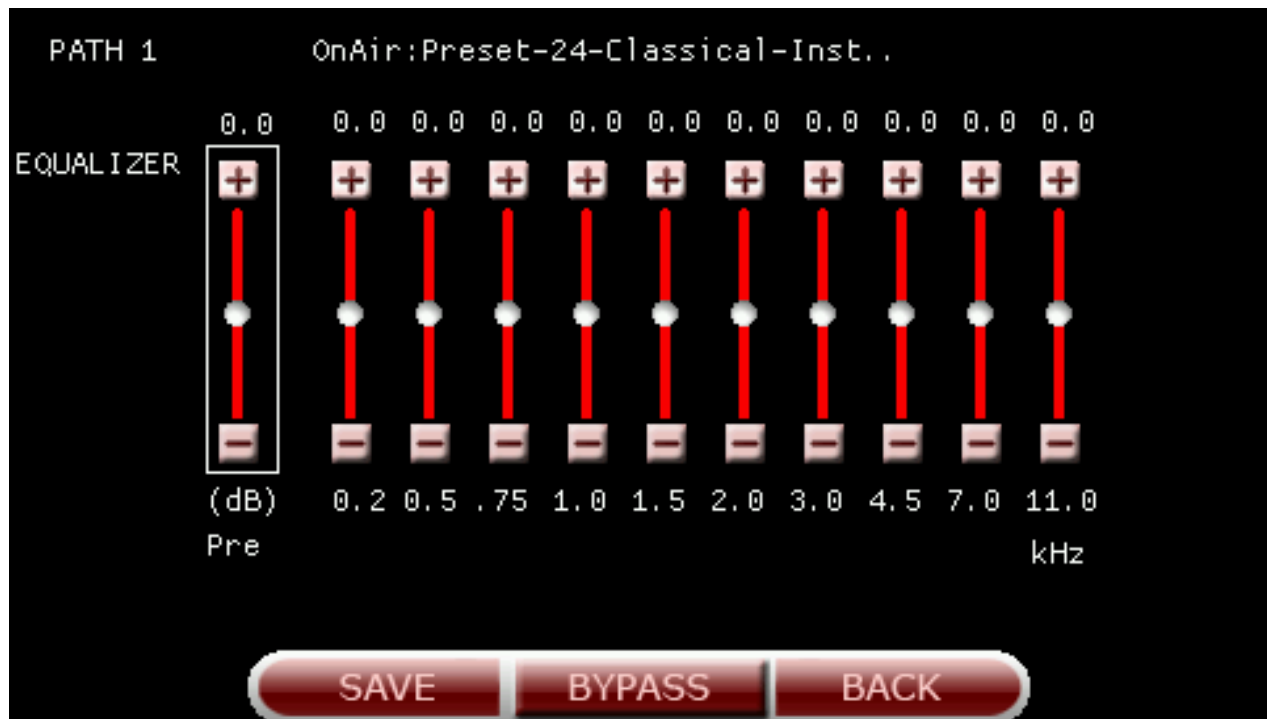


Figure 5.3.10: Equalizer Screen

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

Band Frequency

- 10.2 kHz 20.5 kHz 30.75 kHz 41.0 kHz 51.5 kHz 62.0 kHz 73.0 kHz 84.5 kHz 97.0 kHz 101.0 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.

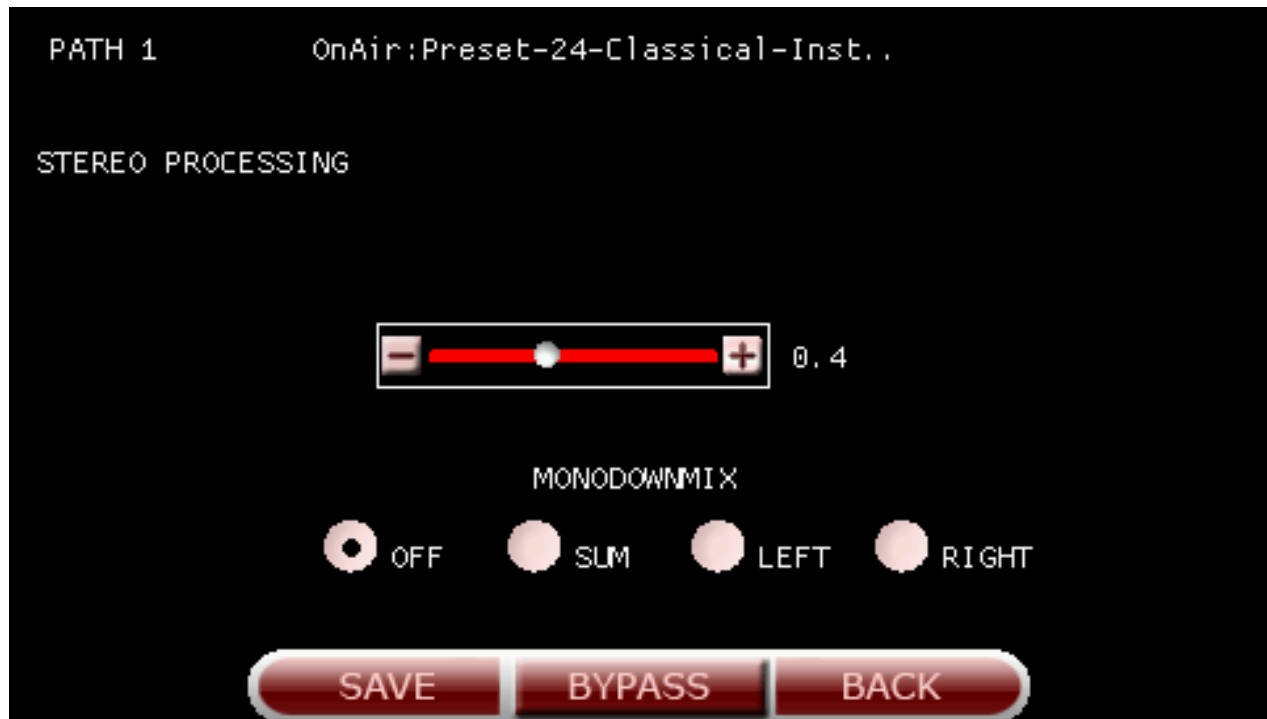


Figure 5.3.11: Stereo Processing Screen

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.

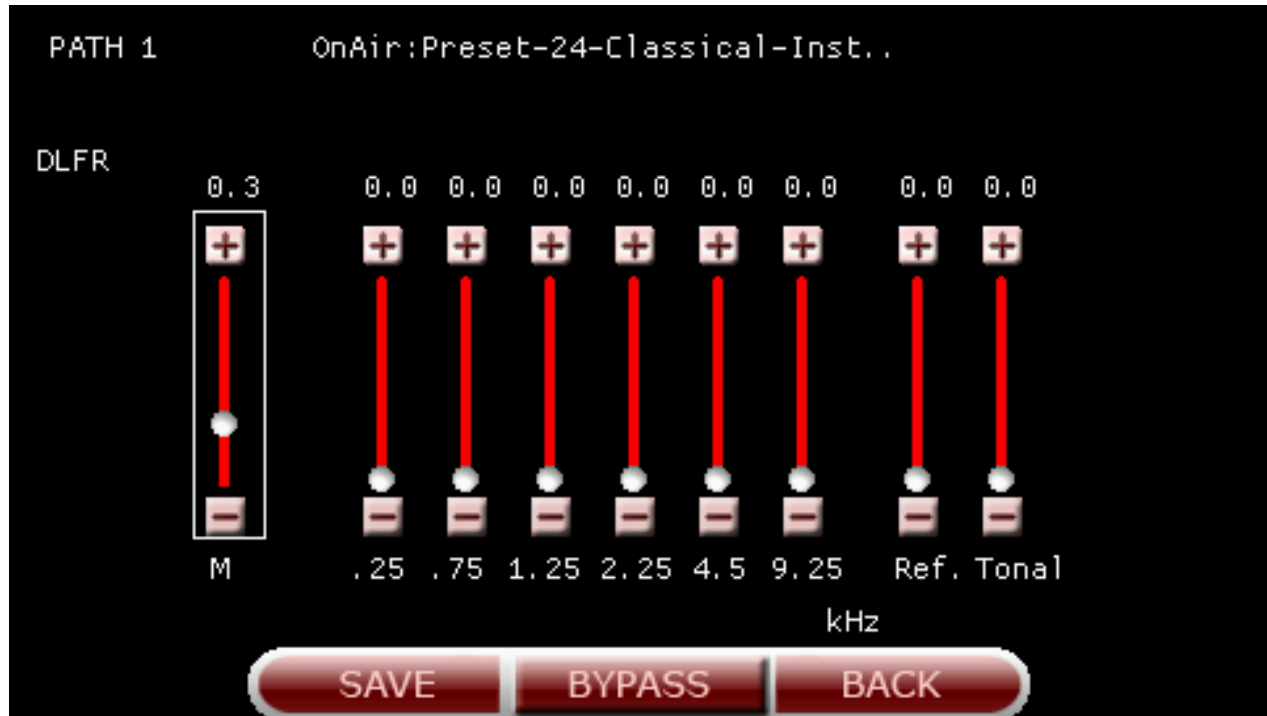


Figure 5.3.12: Dynamic Listening Fatigue Reduction Screen

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.

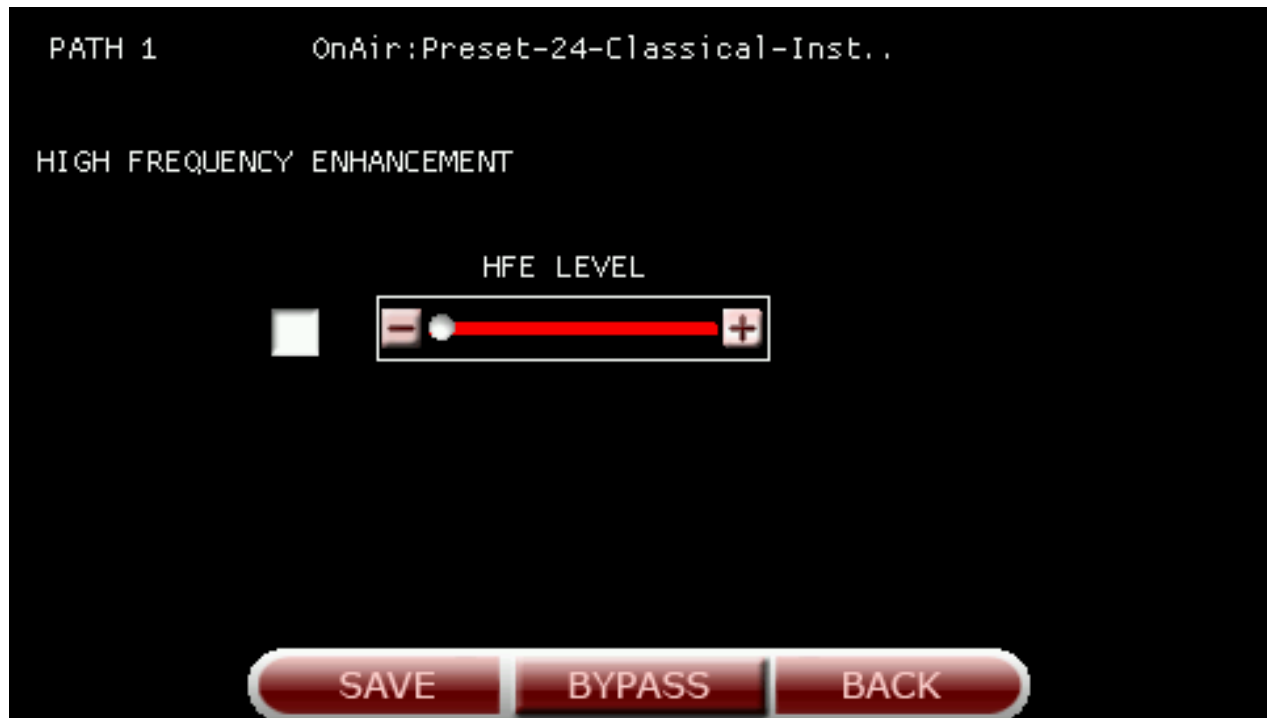


Figure 5.3.13: High Frequency Enhancement Screen

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.

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.

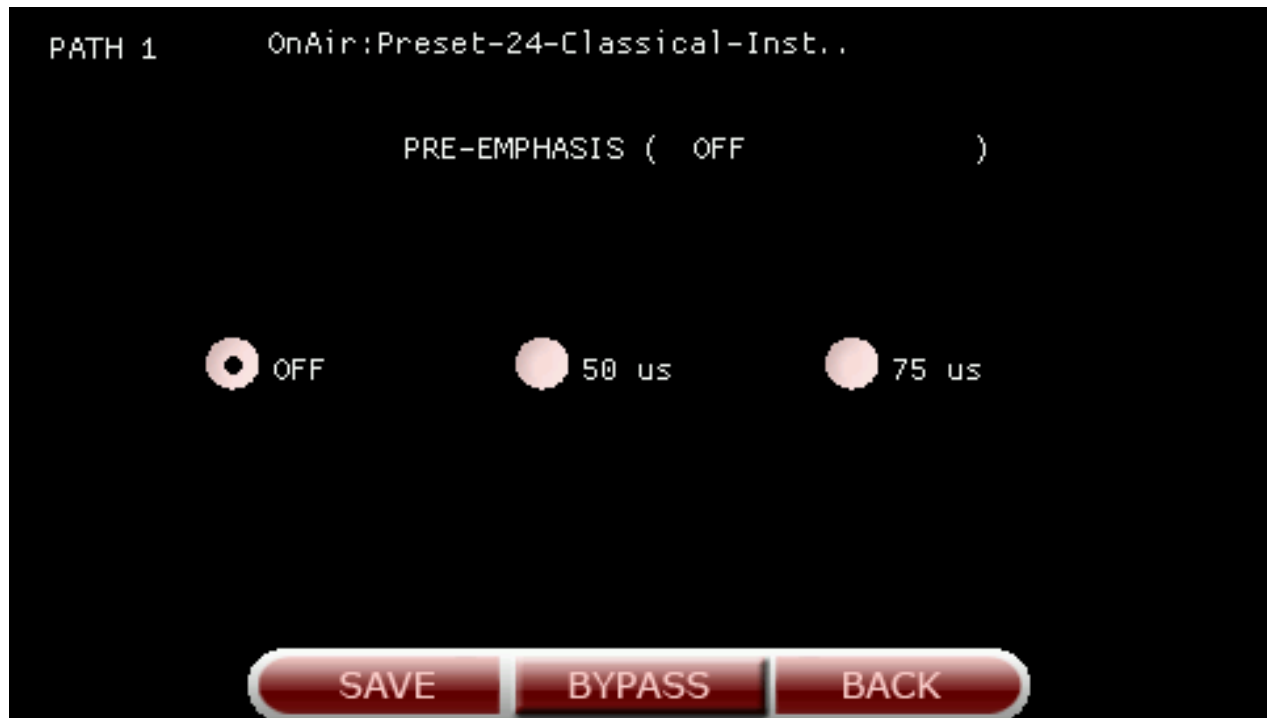


Figure 5.3.14: Pre-Emphasis Screen

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 most 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.

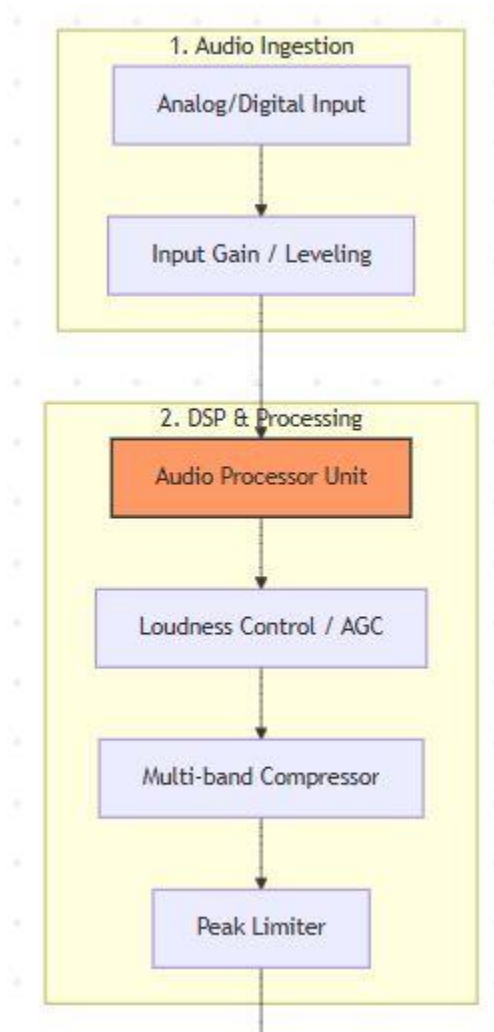
Only one option can be selected at a time. The selected value must match the corresponding de-emphasis setting at the receiver end.

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 demands of internet streaming of processed audio, optimized, robust and high perceptual quality compression technology is essential. Q24-4888/Q24-4880 processed Audio is transmitted on Internet using quality standard codecs and protocols. AAC and MP3 are well known best standard audio compression technology available in the market which is optimized and compatible with all online audio players running on target devices chipset. Different type of bitrates are supported in the Q24-4888/Q24-4880 product based on user specification and targets. This section describes the Codecs technology and streaming protocols used in the product. **Figure 5.4.1 shows the various steps in the internet streaming process.**



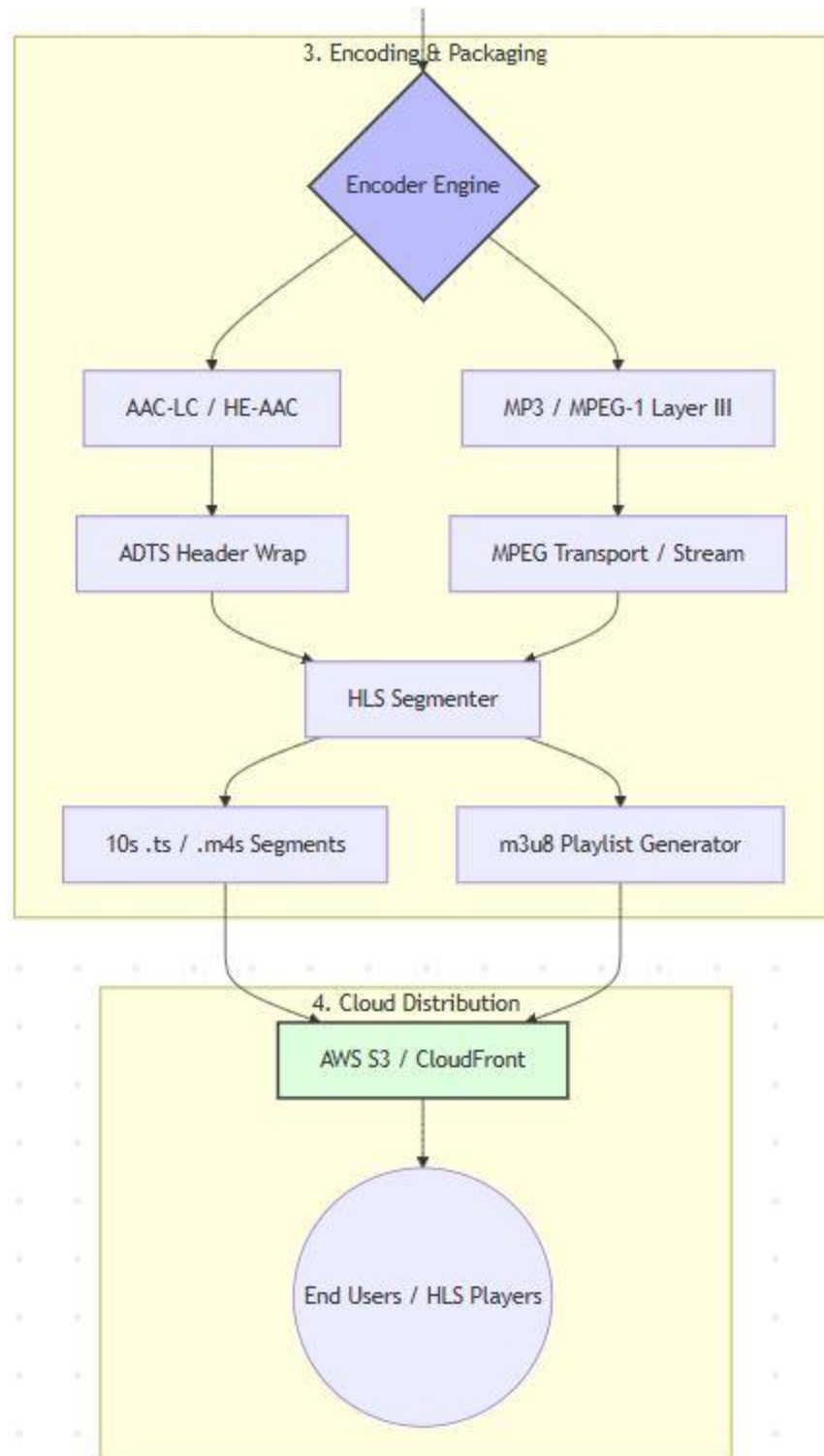


Figure 5.4.1: Internet Streaming Process

5.4.1 AAC LC and HE-AAC

AAC is the latest standard codec used in audio compression which provides transparent perceptual quality at low bitrates.

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

1. Time-to-Frequency Transformation: The MDCT

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

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

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

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

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

3. Advanced Quantization & Bit Reservoir

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

- **The Logarithmic Advantage:** Humans are more sensitive to changes in quiet sounds than loud ones. The 3/4 power-law allocates more precision to low-amplitude signals and less to high-amplitude ones, mimicking the logarithmic nature of human hearing.
- **Bit Reservoir Logic:** HLS usually requires **CBR (Constant Bitrate)** for predictable AWS bandwidth. However, audio complexity varies.

➤ *Simple frames (silence/pure tones):* Use fewer bits than the target.

- *Complex frames (applause/orchestra)*: These pull the "saved" bits from the reservoir.
- This ensures that a 10-second HLS segment maintains a steady data rate while allowing "bursts" of quality where the music demands it.

4. TNS (Temporal Noise Shaping)

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

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

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

Detailed Algorithmic Analysis of SBR (Spectral Band Replication)

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

1. The Split-Band Architecture

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

2. The Reconstruction Algorithm

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

- **Transposition:** It takes the decoded low-band harmonics and "transposes" (copies) them up into the high-frequency range.
- **Envelope Adjustment:** It applies the side-chain metadata to the transposed high-band. This ensures the energy levels of the high frequencies match the original source.
- **Noise/Inverse Filtering:** SBR can inject "synthetic noise" or "sine addition" into the high-band to reconstruct non-harmonic sounds like atmospheric air or percussive brushes that aren't present in the low-band harmonics.

3. QMF (Quadrature Mirror Filter) Bank

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

- The signal is passed through a 64-channel QMF bank.

- This allows the algorithm to have high frequency resolution for the replication process while maintaining low computational latency—critical for live HLS streaming where the processor must maintain real-time performance.

ADTS Encapsulation & Header Structure

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

ADTS Fixed Header Structure (7 Bytes):

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

5.4.2 MP3

While AAC is the modern standard for HLS, the Q24-4888/Q24-4880 provides robust **MP3 support** for legacy hardware and broad compatibility. The MP3 algorithm is a masterpiece of early digital signal processing, balancing computational limits with acoustic transparency.

1. The Hybrid Filter Bank (The Analysis Stage)

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

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

2. Psychoacoustic Modelling & Huffman Coding

This stage determines which data is "expendable":

- **Masking Thresholds:** The model identifies loud frequencies that "hide" quieter, nearby frequencies. The encoder then allocates zero bits to those hidden signals.

- **Huffman Coding (The Final Crunch):** After quantization, the data is compressed without loss. It analyses the bit patterns; a pattern that appears 1,000 times gets a very short code (e.g., 01), while a rare pattern gets a long code. This is why MP3 files have varying levels of internal efficiency.

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

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

- **Mid (M):** $S_L + S_R$ (The mono center information).
- **Side (S):** $S_L - S_R$ (The directional/spatial information).
- **Efficiency:** In most broadcasts, the "Side" signal is very quiet. The Q24-4888/Q24-4880 allocates the majority of the bit reservoir to the "Mid" channel, ensuring the core audio remains crystal clear while using minimal data for the stereo width.

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

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

A. PES (Packetized Elementary Stream)

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

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

B. MPEG-TS Muxing (The Container)

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

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

Section 6 General Purpose Inputs/Outputs (GPIO)

Optional – Available Only on Products Ordered with this Option

Perceptual SoundMax™ audio processor offers 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 reliable input as a zero. If the input pins are not connected to electrical circuitry the input to the processor is a logical zero. For the outputs a logical 1 outputs a voltage of 5 V between the +ive and ground pins of the corresponding inputs. The role of inputs and outputs can be configured using the SoundMax™ Manager panel.

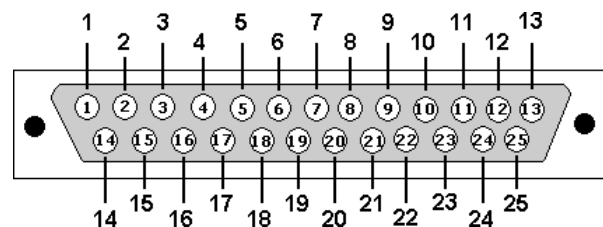


Figure 6.1: Connector Pin Configuration

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

GPIO Input Configuration

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



Figure 6.2: GPIO Settings Panel

The Options will be interpreted as shown below:

For Bits b1 and b0:

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

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

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

GPIO Output Configuration

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

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

Section 7 Hardware Specifications

Analog Audio Input:

- Configuration: 8-channel Unbalanced TRS Analog Stereo Input
- Maximum Input Level/Impedance: +24 dBu / >10 kΩ
- Nominal Input Level: Adjustable -10.0 to +13.0 dBu
- A/D Convertor Resolution: 24-bit
- EMI Suppressed: Yes
- Connectors: Balanced XLR Female

Analog Audio Output:

- **Configuration:** 8-channel Left and Right / Stereo, Flat or Pre-Emphasized (50μs or 75μs software selectable)
- **Output level:** Adjustable, -6.0 to +24 dBu
- **Frequency Response:** ±0.3 dB (20 Hz – 20 kHz),
- **SNR:** ≥ 90dB (20 Hz – 20 kHz), unweighted
- **Crosstalk:** ≤ -90 dB (20 Hz to 20 kHz)
- **Distortion:** THD+Noise ≤ 0.01% (20 Hz to 20 kHz)
- **D/A Convertor Resolution:** 24-bit
- **EMI Suppressed:** Yes
- **Connectors:** Balanced XLR Male

Digital Audio Input (AES/EBU):

- **Configuration:** 8-channel XLR Stereo
- **Sampling Rate:** 32, 44.1 or 48 kHz (Auto Sense)
- **Wordlength:** 16 or 24 bit
- **Input Reference level:** -25 to 0 dBFS
- **EMI Suppressed:** Yes
- **Connector:** AES – XLR Female, 110Ω

Digital Audio Output (AES/EBU):

- **Configuration:** 8-channel Stereo output with individual software level control.
- **Output level:** -20 to 0.00 dBFS
- **Frequency Response:** ± 0.3dB (5 Hz to 20 kHz) Sampling Rate: 32, 44.1, 48 kHz (User selectable) EMI Suppressed: Yes
- **Connector:** Two AES – XLR Male
- **De-Emphasized:** 50μs or 75μs software selectable

Audio Over IP Stream Input/Output:

- **Configuration:** 8 live audio channels multiplexed over MPEG-TS streams on UDP multicast. 8-channel Stereo, Livewire/Ravenna/Wheatnet input support, AES 6.

- Low to medium Latency
- Connection type UDP & TCP
- Container type MPEG2-TS
- **Connector:** RJ-45 , EMI -suppressed.

Composite Baseband Output:

- **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

Streaming Audio Input/Output

- **Configuration:** : 8 independent streams Input/Output
- **Encoding format:** Enhanced AAC+ with ADTS encapsulation
- **Bit rates:** 32 kbps – 256 kbps
- **Player compatibility:** VLC, Windows media player, Real Player
- **Browser compatibility:** IE, Chrome, Safari, Mozilla
- **Protocols:** Unicast, Multicast, TCP, UDP, HLS, MPEG2, MPEG-TS
- **Distribution servers:** Icecast, Shoutcast

Management Control:

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

USB Connection:

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

Ethernet & Remote PC Control:

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

- **Protocols:** TCP/IP , UDP , Multicast

GPIO Interface:

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

Power

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

Powerfactor: A/Input	Input line current	FullloadP.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.

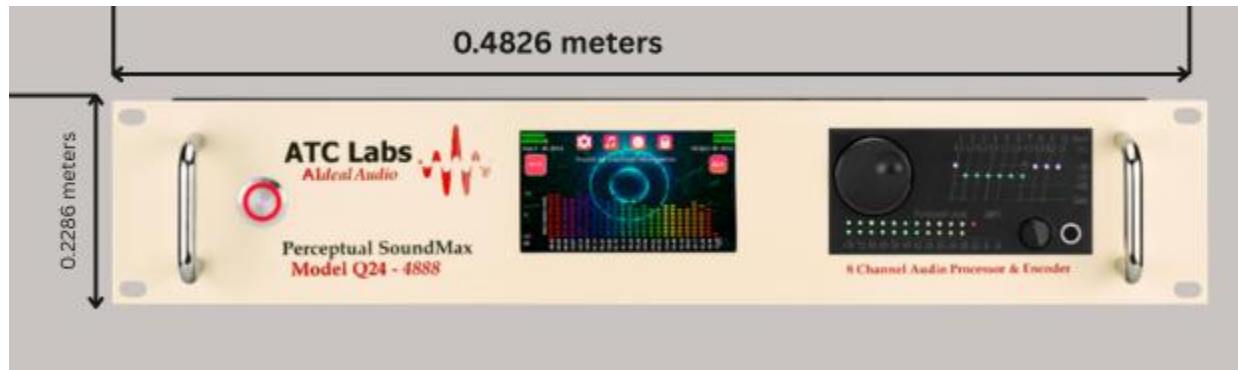
Dimensions:

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

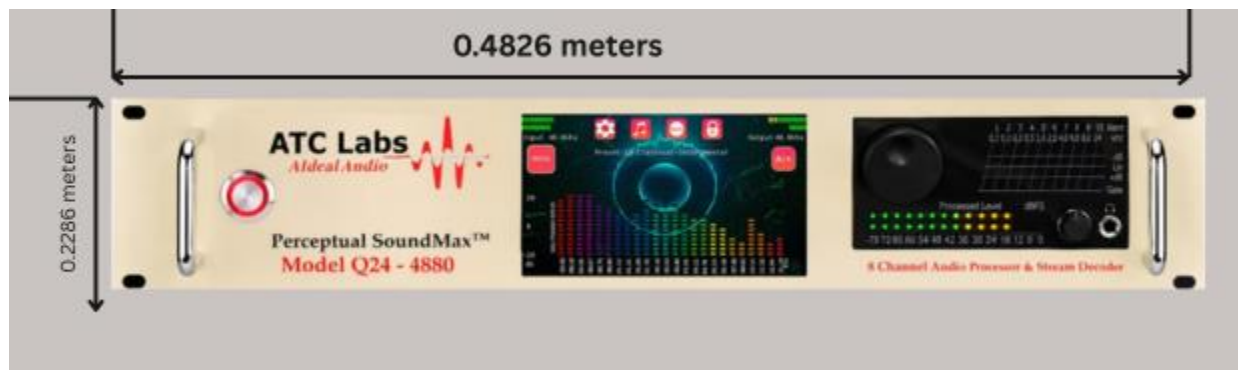
Section 8 Dimensioned Product View

Isometric View with Dimensions (meters)

Front Panel View with Dimensions (in meters & inches)



Front panel View of 4888



Front panel View of 4880

Back Panel View with Dimensions (in meters & inches)

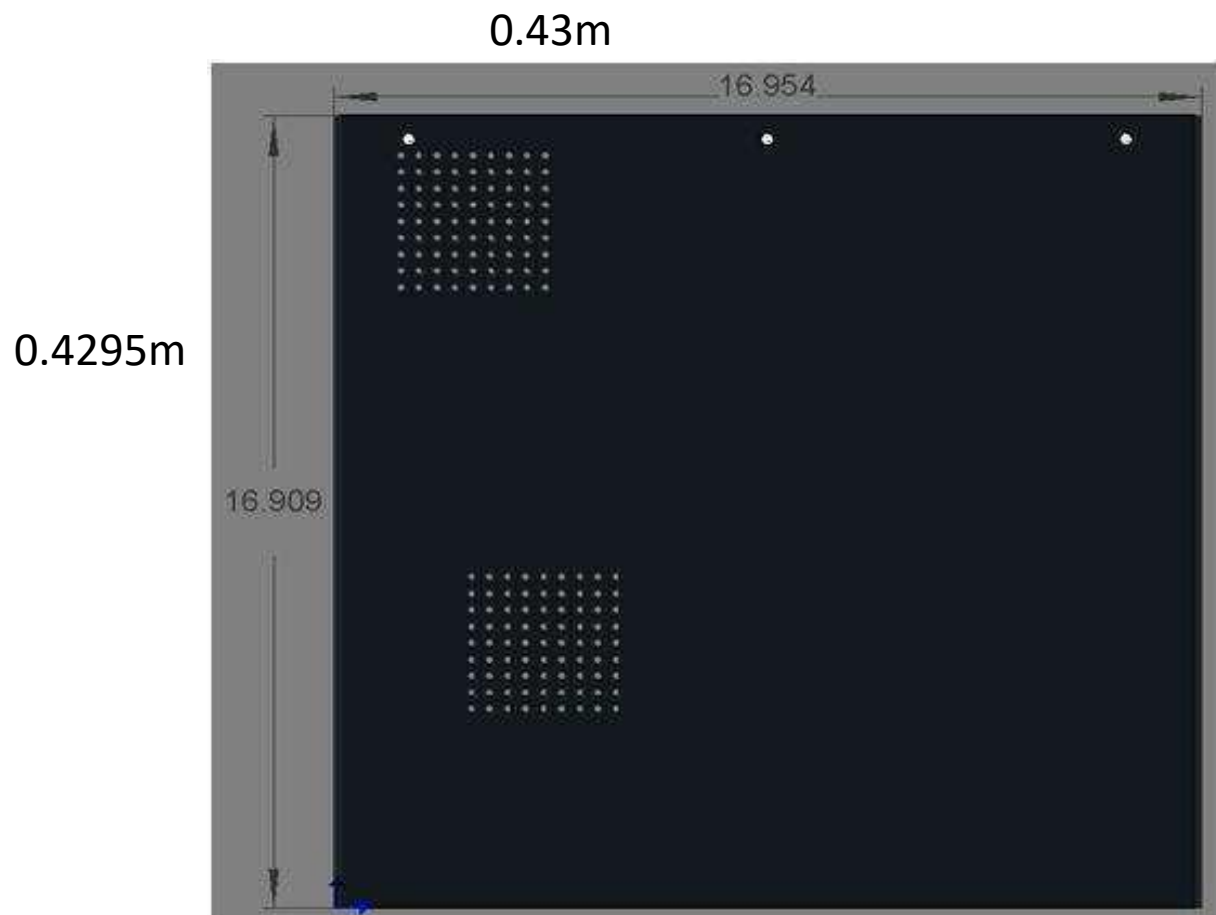


Back panel View of 4888



Back panel View of 4880

TopCoverViewwithDimensions(inmeters& 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

- [SoundMax™ Product Page](#)
- **License Purchase and Admin Dashboard:**
<https://dwywczovty6aj.cloudfront.net/server/webphp/new-en.php>
- **Sales Enquiry:** <https://www.atc-labs.com/contact-sales.php>
- **Support Enquiry:** <https://www.atc-labs.com/support.php>

Perceptual SoundMax™ Version 4.1

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