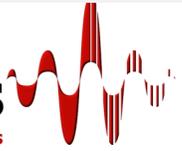


ATC Labs
Audio Technologies and Codecs



Perceptual SoundMax™

Model Q24

24-band Stereo Audio Processor for FM and Digital

<http://www.atc-labs.com/soundmax>

Operation Manual

Version 2.1, January 2012

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 Equipments of its Class.

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Section 1

Product Overview

Perceptual SoundMax™ is a new digital audio processor for FM & DAB employing powerful state-of-art signal processing techniques. It performs elaborate 24-band processing using an extremely detailed and accurate time-frequency representation of the audio signals. Using the latest advances in signal processing techniques such as filter banks and unique ATC Labs know how in Psychoacoustic Modeling and Harmonic Analysis we have developed a set of novel signal conditioning tools which maximize the performance the 24-band dynamics processing and augment it with novel signal conditioning tools. The result is a unique and pleasing sound that emphasizes the presence and perceived loudness while at the same time improving long term listenability. To put simply one can create breathtakingly detailed, consistent and balanced sound which is loud and punchy without introducing any element of listener fatigue.

ATC Labs has a strong sound processing technology pedigree. It was founded by researchers with long career in Bell Labs acoustics research. Technology developed by ATC Lab including the Perceptual SoundMax™ audio processor is being employed among others by leading US Satellite/Internet Radio Broadcasters' servicing over 20,000,000 listeners. Perceptual SoundMax™ is a new product coming out from our proven sound processing technology platform that has been in use for nearly 7 years. It allows conventional FM and AM broadcasters to achieve unparalleled sound quality and clarity. A lot of the technology embedded in Perceptual SoundMax™ is developed at our R&D center in Noida, India staffed with skillful research engineers and it is manufactured in our Noida facility.

Perceptual SoundMax™ Model Q24 is actually 3 products rolled into one - in addition to sophisticated 24-band processing it includes a built-in broadcast grade wide-band background noise removal product as well as a suite of audio conditioning/sweetening tools into one box eliminating the need for broadcasters to place 3 or more separate hardware products in their broadcast chain to achieve the similar impact. Model Q24

includes all standard inputs and outputs expected of a product of this class such as Digital AES I/O balanced analog I/O, baseband composite FM outputs, SCA & RDS inputs, etc. It offers an attractive touch-screen interface for user control and is supplemented by sophisticated software that allow unsurpassed monitoring and tuning options using a networked PC based remote control. The product comes pre-loaded with multiple processing presets matched to different genres and allows the user to create and save as many custom profiles as desired. Based on the highest quality components, the product is meant for uninterrupted 24x7 operations for years with minimum maintenance requirement.

Section 2

Installation And Quick Start Guide

2.1 Supplied Accessories

- ✓ Perceptual SoundMax Model Q24 box as shipped contains the following items:
 1. Perceptual SoundMax Audio Processor for Digital and FM (Model Q24) hardware unit
 2. Power cord
 3. TFT Stylus Pen
 4. AES Cables with XLR connectors (2X)
 5. Rack-mounting kit including rack-mounting screws, 10-32 x ½ — with washers, #10
 6. Ethernet Cable
 7. SoundMax Manager software installation CD-ROM
 8. Operation Manual
 9. Warranty & Registration Card
 10. Custom Profile Creation Worksheets (4X)

If any items are missing or if any damage is observed, please get in touch with your sales contact immediately.

- ✓ Save all packing materials! If you should ever have to ship the Model Q24 (e.g., for servicing), it is best to ship it in the original carton with its packing materials because both the carton and packing material have been carefully designed to protect the unit.
- ✓ Complete the Registration Card and return it to ATC Labs (address on the card). The Registration Card enables us to inform you of new applications, performance improvements, software updates, and service aids that may be developed, and it helps us respond promptly to claims under warranty without our having to request a copy of your bill of sale or other proof of purchase. Please fill in the Registration Card and send it to us today. We do not sell our customer's information to anyone.

2.2 Rack Mounting the Unit

Use the supplied rack-mounting kit to mount the unit in a standard 19-inch equipment rack. Model Q24 requires standard rack unit space of 2.5 U (4.37inch). There should be a good ground connection between the rack and the chassis. It is recommended that you leave 1U of space both above and below the unit and 10 cm (4 inch) or more space in the rear.

2.2.1 Environmental Considerations

- **Ventilation:** The equipment should be situated so that its position or location does not interfere with its proper ventilation. For Example, the equipment should not be situated on bed, sofa, rug or any similar surface which may block the ventilation openings; or, placed in a built in installation such as a bookshelf or cabinets which obstruct the flow of air through the ventilation openings.
- **Moisture & Water:** The equipment should not be used near water (e.g near a washbowl, bathtub, kitchen sink, laundry tub etc, in a wet basement, or near a swimming pool etc)
- **Heat:** The equipment should be situated away from heat sources such as stoves, heat registers, radiators, or any other equipment (including amplifiers) which produces heat.
- **Power source:** The equipment should be connected to a power supply only of the type defined in the operating instructions or as marked on the equipment.
- **Grounding or Polarization:** Precautions should be taken so that the grounding or polarizations means of equipment is not defeated.

2.2.2 Use Precautions

- **Power-Cord Protection:** Power supply cords should be routed so that they are not likely to be walked on or pinched by items placed upon or against them, paying particular attention to the cords at plugs, convenience receptacles, and the point where they exit from the equipment.
- **Cleaning:** The equipment should be cleaned only as recommended by manufacturer.

- **Non Use periods:** The power cord of the equipment should be unplugged from the outlet when left unused for a long period of time.
- **Object and Liquid Entry:** Care should be taken so that objects do not fall and liquids are not spilled into the enclosure through openings.
- **General:** Model Q24 has a sturdy design and can be used in most places, but to prolong operating life the following environmental considerations are recommended.
 - Do not place the unit on unsteady surfaces experiencing frequent vibrations or resonance.
 - Ambient temperature should not exceed 50° C when equipment is powered on. Equipment life may be further extended by operating it in cooler places.

2.3 I/O and Power Connections

All the I/O and power connectors for Model Q24 are available at the rear panel (with the exception of headphone monitoring output which is available on the front panel as described below). Make the power and I/O and power connections as described below.

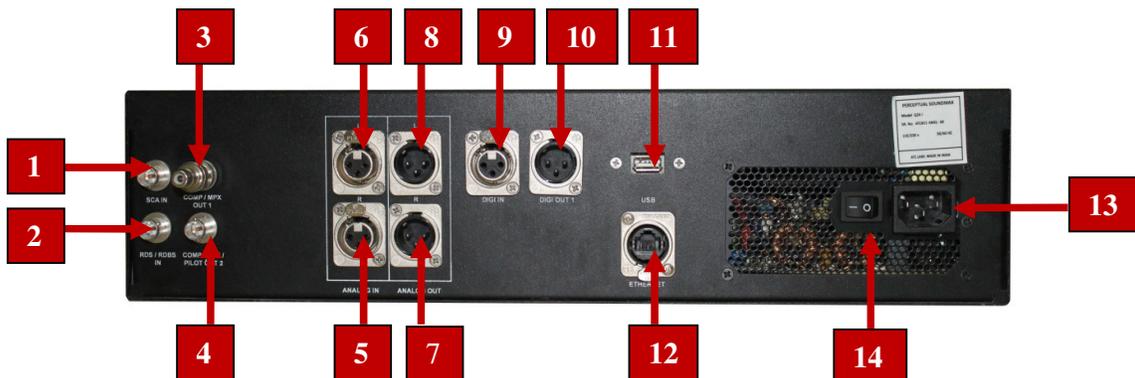


Figure 2.1: Model Q24 Rear Panel

- ✓ **Connecting the Power Supply:** Q24 box can operate at 110-250V, 50Hz/60Hz voltage rating. Verify your line settings before to connect to the unit. Use the supplied power cable or another cable of equivalent rating (Label 13 in the Rear Panel Schematic above).

✓ **Input and Output Connections:**

1. Baseband Input SCA – BNC Connector
2. Baseband Input RDS/RBDS– BNC Connector
3. Composite Baseband Output 1 (COMP/MPX) – BNC Connector
4. Composite Baseband Output 2 (COMP/MPX/PILOT)– BNC Connector
5. Balanced Analog Input (Right) - XLR Connector
6. Balanced Analog Input (Left) – XLR Connector
7. Balanced Analog Out (Right) – XLR Connector
8. Balanced Analog Out (Left) – XLR Connector
9. AES/EBU Digital Input – XLR Connector
10. AES/EBU Digital Output – XLR Connector
11. USB Connector
12. Ethernet Connector – RJ 45
13. 3-bin Power Chord Connector (with earthing)
14. Master ON/OFF Switch

- ✓ **Connecting analog audio equipment:** The left and right analog inputs are on EMI-suppressed XLR-type female connectors on the rear panel. Input impedance is greater than 10k Ω ; balanced and floating. Inputs can accommodate up to +24 dBu (0 dBu = 0.775V_{rms}).

The left and right analog outputs are on EMI-suppressed XLR-type male connectors on the rear panel. Output impedance is 50 Ω ; balanced and floating. They can drive 600 Ω or higher impedances, balanced or unbalanced. The peak output level is adjustable from –6 dBu to +24 dBu. Level control of the analog inputs and outputs is accomplished via front-panel control through System Setup (Section 3.2.2).

- ✓ **Connecting digital audio equipment:** The digital input and outputs conform to the professional AES/EBU standard. They all have sample rate converters to allow operation at 32 kHz, 44.1 kHz, 48 kHz sample frequency. On the input the sample rate is auto sensed. For best peak control, operate at 44.1 kHz or higher. The left/right digital input is on one XLR-type female connector on the rear panel; the

left/right digital outputs are on two XLR-type male connectors on the rear panel. Level controls are front-panel configurable (Section 3.2.2).

- ✓ **Connecting composite/pilot outputs:** The stereo encoder has two unbalanced analog baseband outputs on two BNC connectors on the rear panel. Each output can be strapped for 0 or 75 Ω source impedance and can drive up to 8V peak-to-peak into 75 Ω in parallel with up to 0.047 μ F (100ft / 30m of RG-59 / U cable) before any significant audible performance degradation occurs. Independent level control of each output is via front-panel in the SETTING > COMPOSITE screen. (Section 3.2.4)
- ✓ **Connecting SCA/RDS/RBDS inputs:** The stereo encoder has two unbalanced 600 Ω subcarrier (SCA) inputs with rear panel BNC connectors to accept any subcarrier at or above 23 kHz. The subcarriers are mixed into each composite output. Their levels can be changed through front-panel (gains applied internally) before mixing with composite out. (Section 3.2.4)
- ✓ **Turning the Unit into Standby-Power-On Mode:** The unit is equipped with a master power-on/off switch on the rear panel (Label 14 in the schematic above), moving this switch in the on position puts the unit in a standby mode and it can subsequently be turned on using the front-panel ON/OFF switch (see below). Once the power and I/O cables are tightly connected, place the unit in standby mode by moving the master power ON/OFF switch to the ON position.

2.4 Remote Computer Monitoring Software Installation (*This Step may be Skipped During Initial Setup*)

Model Q24 hardware is supplied with SoundMax Manager™ software for remote computer interface. This software is provided with CD-ROM or user can download from <http://www.atc-labs.com/soundmax>. SoundMax Manager™ which allows for elaborate Configuration, Control, Monitoring and Tuning of multiple Q24 processors connected to a network. The presence of one or more Perceptual SoundMax™ Model Q24 on the network is automatically sensed by the SoundMax Manager™ running on any computer connected to that network. By using this software, user has complete flexibility to configure and monitor the Q24 processors. The supported

functionality includes the ability to configure parameters specific to a processors, preset selection, detailed parameter tuning, custom processing profile creation, and much more. Monitoring capability include global view of the processor health and VU meters tied to the Input/Output of each processor.

2.4.1 Installing Perceptual Soundmax Manager

- **Minimum system requirements**

Operating System: SoundMax Manager is currently available for Windows XP, Windows Vista and Windows 7 (32-bit) operating systems.

LAN Connection: The Manager communicates with the Station through a network connection.

Hardware: The SoundMax Manager can run on any light weight PC. We recommend at least a single core Pentium with a clock speed of 2.2 GHz (or above).

- **Instruction for installation**

Setting up the MANAGER

- ✓ Run the SoundMax Manager setup on a PC and follow the instructions
- ✓ After the installation SoundMax Manager user interface screen (UI) will come as shown below:

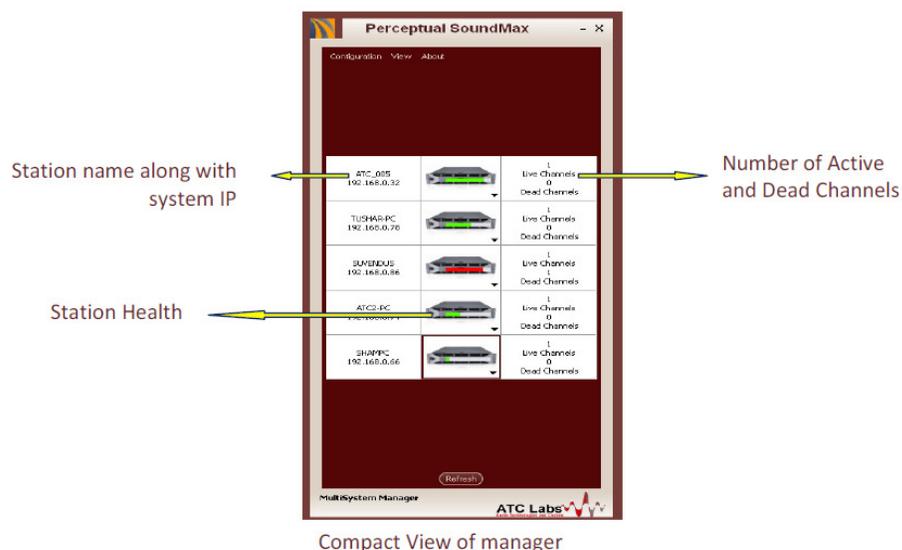


Figure 2.2: Compact view of Manager

- ✓ SoundMax Manager will auto detect the “Model Q24 rack Mount processor” as Live Station. By clicking on the station from this list, you can get a detailed view and options to configure that particular station. This view provides the ability to configure parameters specific to a channel such as profile selection, save and remove profiles, tune profile parameters. A snapshot of this detailed view is shown in the figure below.

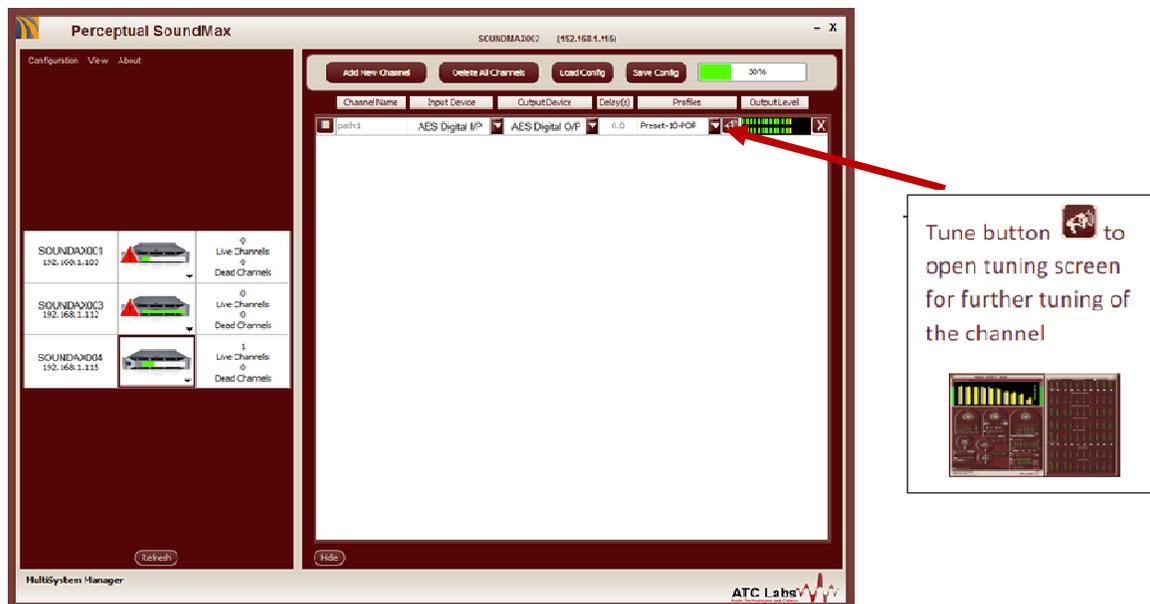


Figure 2.3: Detailed view of Manager

- ✓ 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 more detailed explanation on TUNING through manager ref section 4.
- ✓ If SoundMax Manager is controlling the Processor then front-panel TFT interface (see below) will be disabled.

2.5 Front Panel Controls

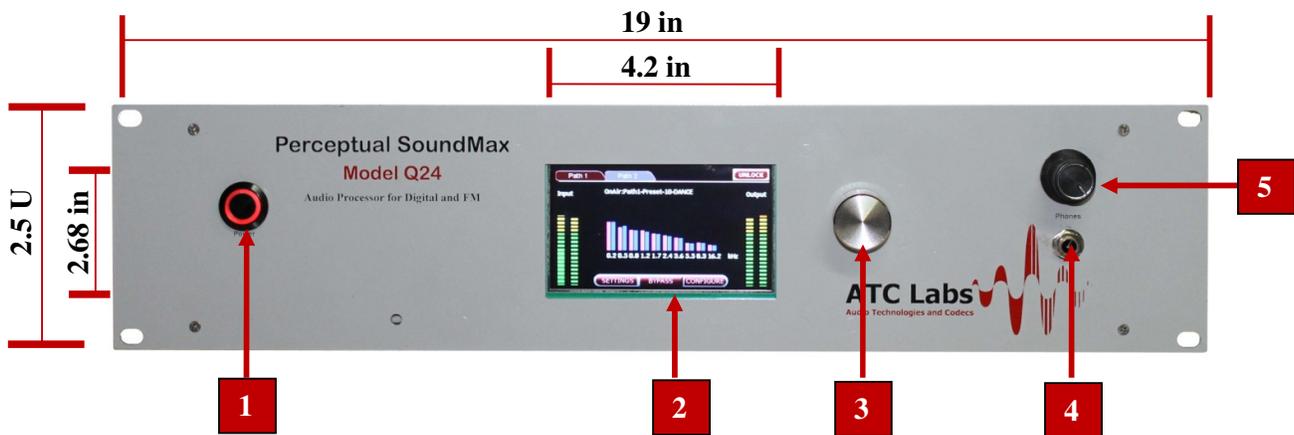


Figure 2.4: Model Q24 FRONT PANEL

- 1. Power Switch:** Once the rear-panel master switch is in ON position (i.e. equipment is in power standby mode) use this switch to turn the unit ON or OFF.
- 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 in lieu of the touch screen control for controlling the unit. Generally user can also select the particular option displayed over 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 Audio Output – 1/4" jack:** Used for quick monitoring of processed output of audio processor. User can listen to the processed output through this audio jack.
- 5. Headphone Volume Control**

2.6 Quick Start Guide

(WARNING: Do not skip to this section without familiarizing yourself with the material in the above sections)

- ✓ Make sure Installation procedure is complete as per instruction given in *Section 2.1-2.3*
- ✓ Model Q24 ships with digital input and digital output as the default I/O connections. So during the quick start the audio coming in through the digital input will be processed and the processed output will be available on the digital output connection. If analog I/O is needed instead please refer to Section 3.2 to change I/O configuration (System will be start with default settings or last user settings, if applicable).
- ✓ Once the front-panel power switch is pressed, the unit will power ON and LED around the ON/OFF switch will light up. It may take several seconds for the unit to power up and during this time various start-up messages will be displayed on the front-panel TFT screen. After the unit is fully powered up the front-panel displays a HOME screen which SHOULD BE similar to the figure below. If not then refer to section 3.2.5.

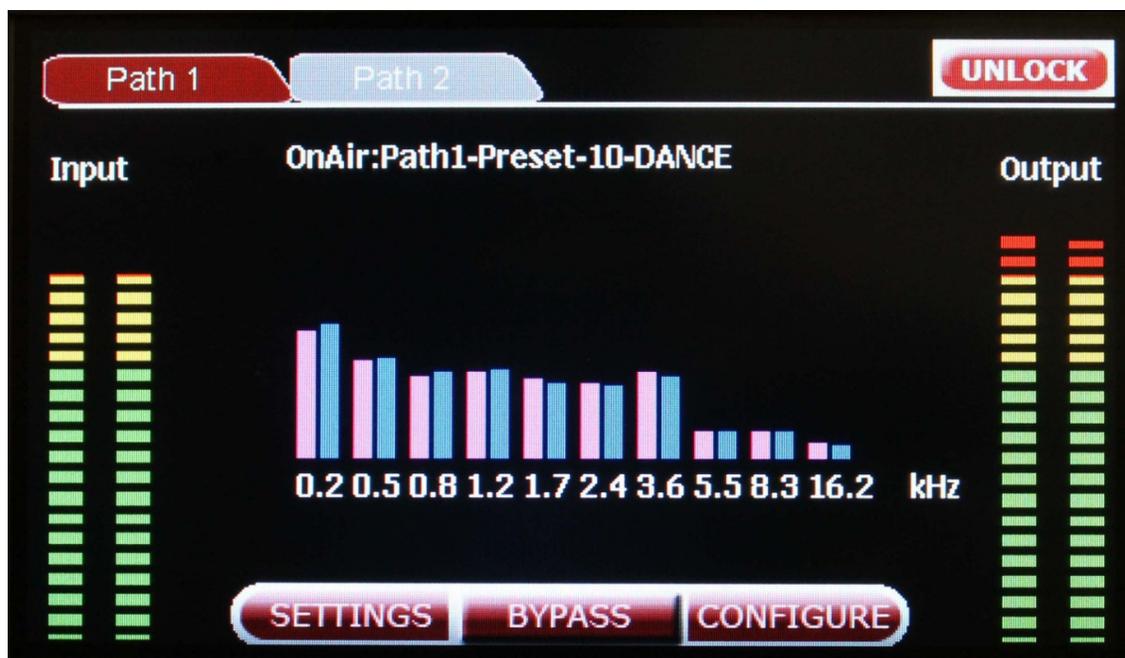


Figure 2.5: HOME Screen

- ✓ Processor settings can be modified only after unlocking the controls by providing valid authentication key. After clicking on UNLOCK tab follow the instructions given below:
 - Enter a valid password and SAVE it (the unit ships with a default password of “1234”).
 - For changing the password refer section 3.2.5 (SET PASSWORD).

- ✓ Various Controls on the Home Screen (and other screens) can be exercised either by pressing the buttons using the stylus pen or using the Rotary Encoder control; when the Rotary Encoder is rotated highlights move around the screen in a pre-defined pattern and a highlighted control can be exercised by pushing in the Rotary Encoder.

- ✓ Click on SETTING key to configure Analog I/O setting, Digital I/O setting, composite and SCA/RBDS setting (These are Advanced Settings described in Section 3.2 and onwards)
 - For the first time the unit starts with default I/O settings, i.e. digital input and output)
 - If user has changed these settings and shut down the unit then Unit will start with last configuration.

- ✓ Click on CONFIGURE key to get the list of all factory presets and custom profiles.
 - There are presets for different genre being displayed with explicit information of number of processing bands, i.e.
 - “Preset-10 BASS-TREBLE”, is the 10 band processing preset.
 - “Preset-24 BASS-TREBLE”, is the 24 band processing preset.



Figure 2.6: Preset Profile Selection Screen

- With the soft keys of NEXT  and PREVIOUS  user can traverse between next page and previous page respectively.
 - User can go to the recent last preset with **LAST** key.
 - With **TUNE** key user can see the different processing module available and their ON/OFF status.
- ✓ Click on **BYPASS** key to switch between processed and unprocessed output.
- If output is in **BYPASS** mode then display text will be “OnAIR: Path1- Processing ByPassed”.
 - If output is in **PROCSSED** mode then preset/custom profile being applied will be displayed as text like “OnAIR: Path1- Preset-10 BASS-TREBLE”.

Section 3

Detailed System Setup Instructions

3.1 Model Q24 Main Screen Controls: Home Screen

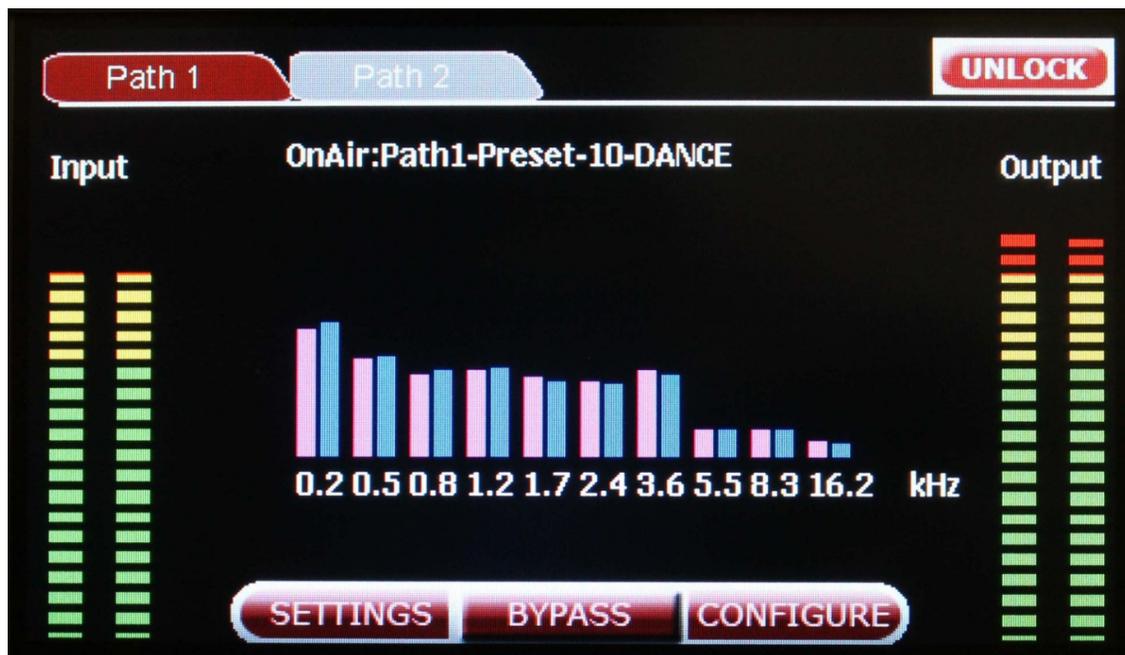


Figure 3.1: Home Screen

Main Screen Display labels the following items:

- Soft keys: UNLOCK, SETTING, BYPASS, CONFIGURE
- Input and Output VU meter
- Output Spectrum in 10 Bandwidth central frequency display.
- On Air preset which displays current profile

User can **recall** Main Screen from any of the other screens by pressing the HOME soft key.

HOME

- ✓ **Click on SETTING key** to get access of setting screen corresponding to Analog I/O setting, Digital I/O setting, composite and SCA/RBDS setting and Info screen
 - Use rotary to go back and forth between various control or click on any of the key.
 - Click on HOME key to go back to MAIN SCREEN.
 - For functional Description of each of the soft key refer *Sec 2.1*.

- ✓ **Click on BYPASS key** to switch between processed and unprocessed output.
 - If output is in BYPASS mode then display text will be “OnAIR: Path1- Processing ByPassed”.
 - If output is in PROCSSSED mode then preset/custom profile being applied will be displayed as text like “OnAIR: Path1- Preset-10 BASS-TREBLE”.

- ✓ **Click on CONFIGURE key** to get the list of all factory presets and custom profiles.
 - There are presets for different genre being displayed with explicit information of number of processing bands, i.e.
 - “Preset-10 BASS-TREBLE”, is the 10 band processing preset.
 - “Preset-24 BASS-TREBLE”, is the 24 band processing preset.



Figure 3.2: Preset Profile Selection Screen

3.2 Input/Output Configuration

With SETTING screen user can setup input and output levels according to studio appliances. Model Q24 is a complex audio processor so before proceedings with any change user must understand the architecture of the processor:

3.2.1 Processor Signal Flow Architecture

The figure below shows the signal processing paths in the system. While Model Q24 offers one processing path, its sister product Model Q24 Duo offers two processing paths with independent output and independent control of the processing parameters. The second processing path is attractive for example to simulcast the audio over a secondary channel such as Internet or digital broadcast. Path 1 directs the output to AES/EBU Digital Output1 as well as Balanced XLR Output which can be used for analog systems; Path 2 (Model Q24 Duo Only) provides output meant for digital systems at AES/EBU Digital Output 2. Path 1 is also directed to an FM composite baseband signal generation units, the output of which is available on two coaxial BNC connectors.

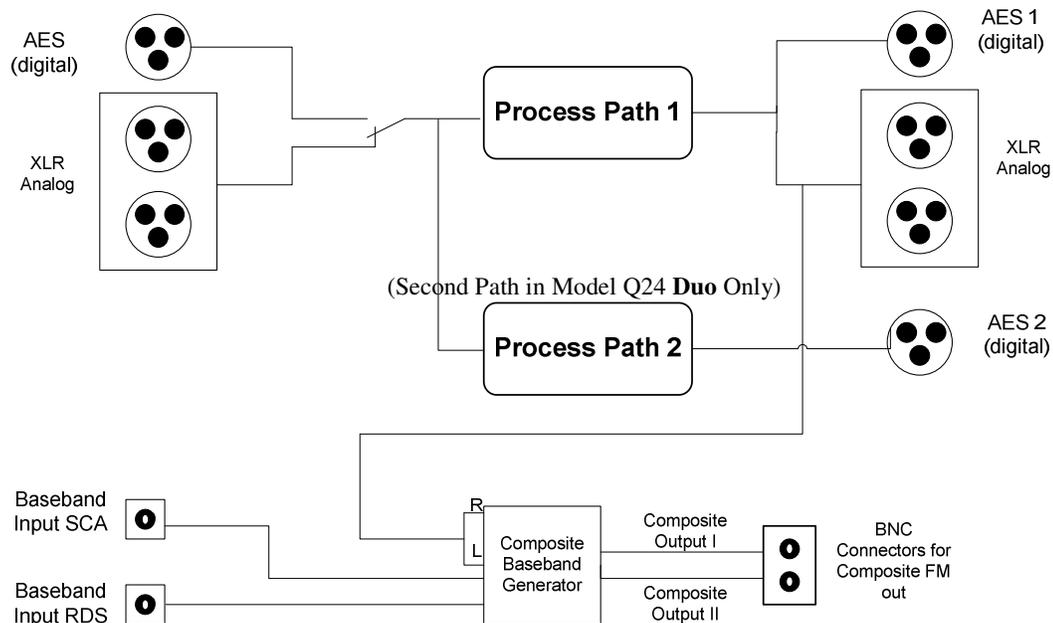


Figure 3.3: Signal Processing Paths in Perceptual SoundMax™ Model Q24

3.2.2 Input Configuration

Model Q24 supports Stereo Input either AES/EBU standard or balanced XLR type. Input selection is software configurable and De-emphasis is also software selectable.

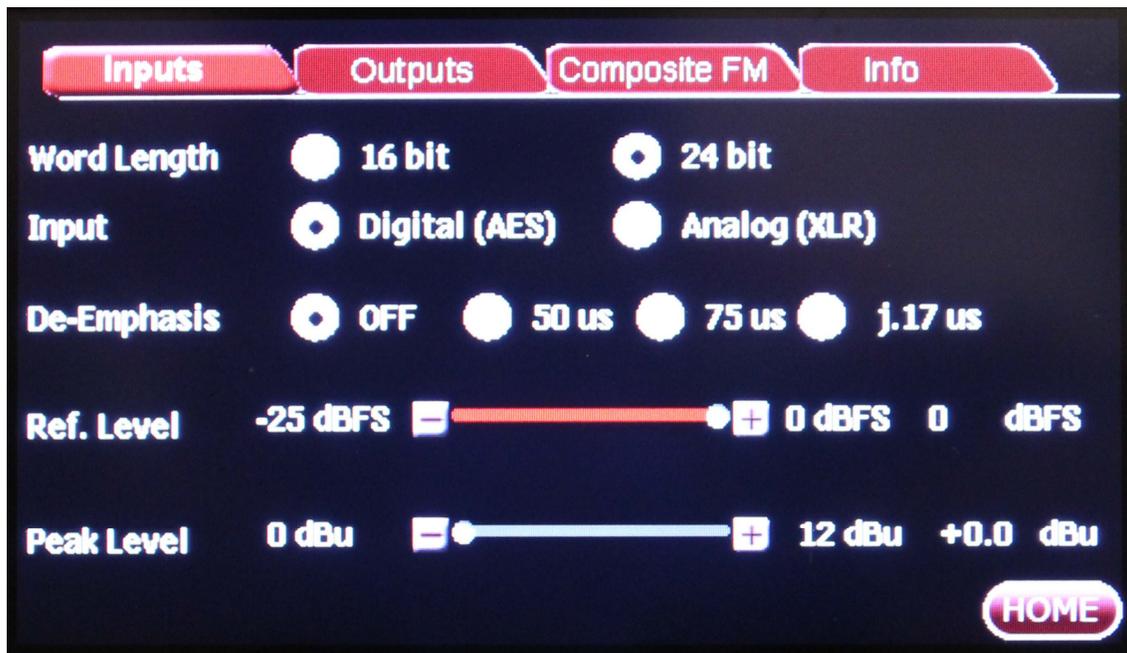


Figure 3.4: Input Configuration

- **Word Length:** It is relevant in case of Analog Input only. This can be selected as *24 or 16 bit resolution*. In case of Digital Input this option will be auto detected from AES bit stream. So WORD LENGTH option will be disabled for Digital Input.
- **Input Type:** To be processed input type either ANALOG or DIGITAL can be selected from here.
- **De-Emphasis:** If your input is Pre-emphasized with type *J.17, 50μs or 75 μs* then corresponding De-emphasis should be applied before processing.
- **Reference Level:**
 - **Analog reference level:** This step matches the level at which the Q24's A-D (Analog-to-Digital) converter clips to the absolute maximum peak level that

your installation supplies to the Q24's analog input. It has range of -10dBu to +12dBu.

- **Digital reference level:** It has range from -25dBFS to 0dBFS
- **Peak Level:**
 - **Analog peak level:** This tunable option provides necessary HEADROOM to take care of peaks above the reference level. It has range of 0dBu to 12dBu.
 - **Digital peak level:** Digital Headroom is provided with the Highly Effective Automatic Gain Control Module, Which will detect the Peak level and adjust the overall gain.

3.2.3 Output Configuration

Model Q24 has Stereo XLR balanced analog out and AES out. Output could be either flat or pre-emphasized (at 50 μ s or 75 μ s), software-selectable.

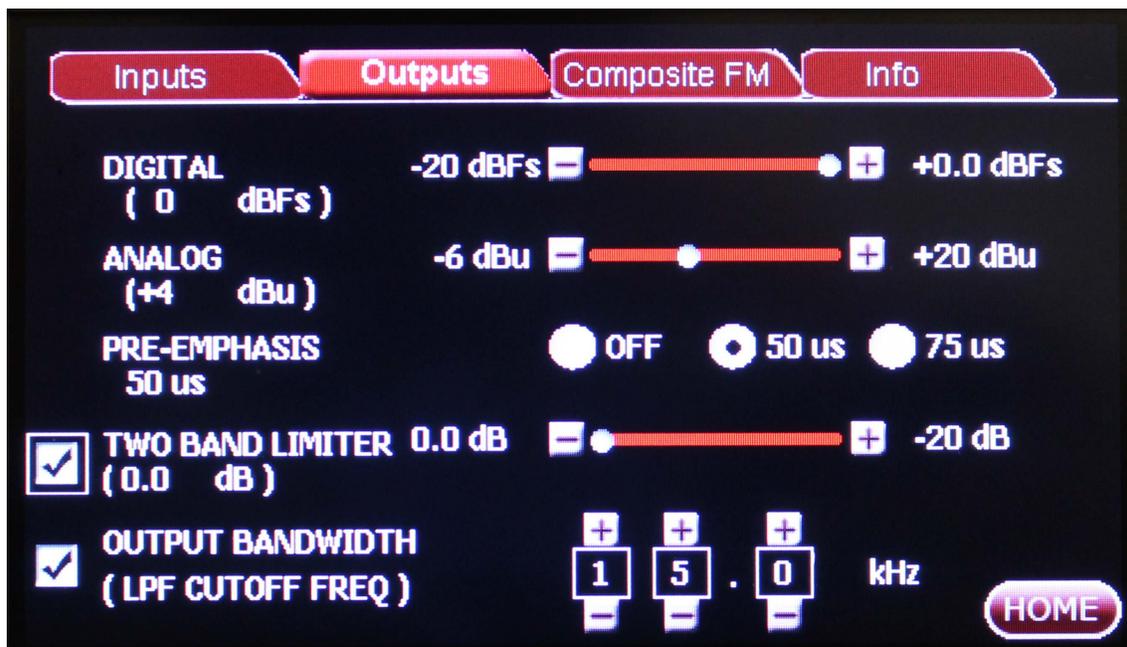


Figure 3.5: Output Configurations

- **Output Level:** User should calibrate and set output levels by playing a full scale TONE as input.
 - Analog output levels: It has a range from *-6dBu to +20dBu*.
 - Digital output levels: It has a range from *-20dBFS to 0dBFS*

- **Pre-emphasis:** This option provides Pre-emphasized ANALOG and DIGITAL outputs. If user wants pre-emphasis only on COMPOSITE PATH then refer to Section 3.2.4.
- **Two Band Limiter:** To avoid distortion caused by over amplification done through pre-emphasis, Q24 is equipped with TWO BAND LIMITER. Two band limiter will control the gain of higher frequency by applying filter having attenuation from *0dB to -20dB*.
- **Output Sample Rate:** Output sample rate is software selectable with following options:- 32kHz, 44.1kHz and 48kHz.
- **Output Bandwidth:** Selecting a 32kHz output sample rate will automatically set 15kHz as the highest available audio bandwidth. User can further reduce this output bandwidth.

3.2.4 Composite FM

Model Q24 has two Composite Baseband Outputs, each with an independent user-controlled output level and connectors.

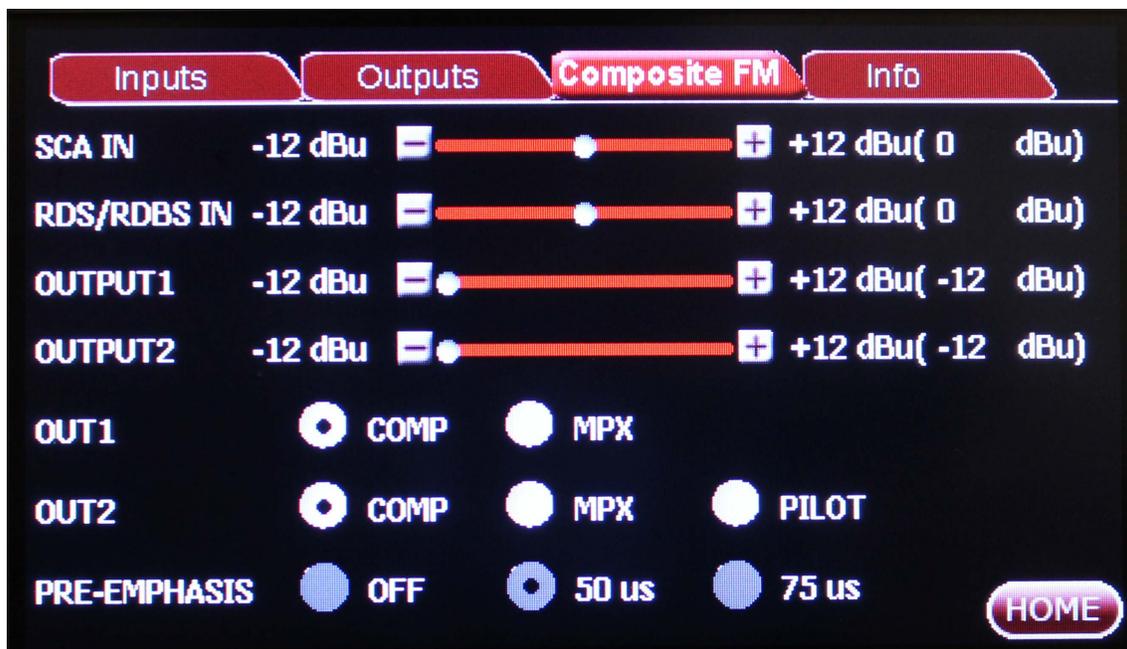


Figure 3.6: Composite FM

- **SCA IN:** Subcarrier inputs sum into composite baseband outputs and provides MPX outputs. SCA and RBDS levels will make sure proper A/D will match to the absolute maximum peak level of analog signal to the full scale digital level with word length 24 bit. It has range from **-12dBu to +12dBu**, configurable as shown above.
- **RDS/RDBS IN:** It has range from **-12dBu to +12dBu**, configurable as shown above.
- **OUTPUT1:** It's the coaxial output and software selectable as COMP or MPX. It has range from **-12dBu to +12dBu**, configurable on the settings screen above.
- **OUTPUT2:** It's the coaxial output and software selectable as COMP, MPX or PILOT tone of 19 kHz. It has range from **-12dBu to +12dBu**, configurable on the settings screen above.
- **OUT1:** With this selection tab user can configure above mentioned OUTPUT1.
- **OUT2:** With this selection tab user can configure above mentioned OUTPUT2.
- **Pre-emphasis:** With this option user can configure type of applied pre-emphasis on COMPOSITE PATH only.

3.2.5 Consolidated Information Screen:

Info screen summarizes the status of all important settings at one place. Info screen displays features like up-gradation of software, setting the user password and diagnostic feature. These new features are accessible through soft keys: UPGRADE, SET PASSWORD, RUN DIAGNOSTICS.

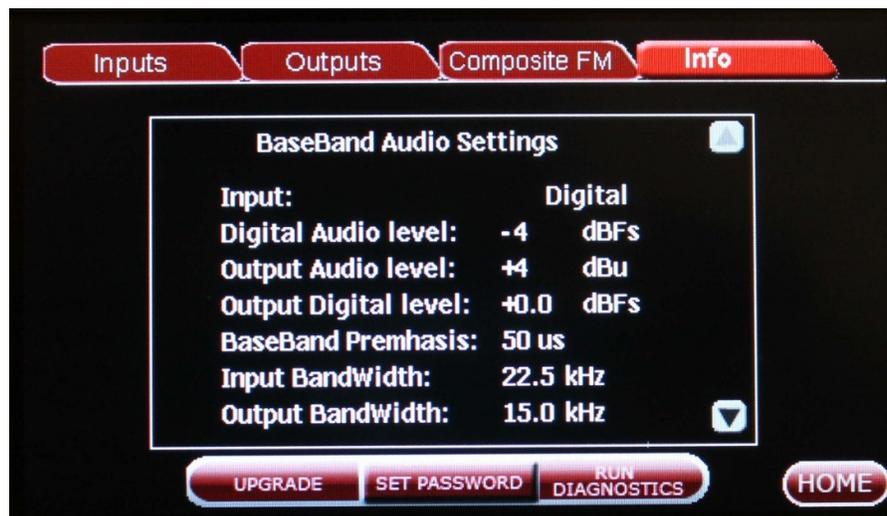


Figure 3.7: Baseband Audio Settings

- Summary of different levels: As a summary Info screen will give following information:

Note: we are providing default factory settings of Q24 for illustration purpose in table below. These setting will be different if user has changed the settings before.

SETTING	Q24 STATUS
Input	Digital
Digital Audio Level	-4 dBFS
Output Audio Level	+4 dBu
Output Digital Level	0.0 dBFS
BaseBand Pre-emphasis	OFF
Input Bandwidth	22.05kHz
Output Bandwidth	22.05kHz
SCAN Input Level	+2.4 dBu
RDS Input Level	+2.4 dBu
Output Level	4.0 dBu
Output1	COMP
Output2	COMP
Baseband Pre-emphasis	50 μ s
Profile	Preset -10-Dance
Station Version	3.0.0.0
PreProcessor Engine Number	2.0.0.0
Mac address
SoundMax IP

Table 3.1: Different Q24 Settings and their Status

- **UPGRADE:** Updating the software is three step procedure:
 - ✓ Download the updates from ATC-labs website.
 - ✓ Extract the folder and keep extracted folder in USB storage
 - ✓ Plug the USB device into Q24's USB port and press UPGRADE key

NOTE: User can also upgrade the software from PC to Q24 interface (refer section 5).

- **SET PASSWORD:** Security key setting screen shown below will come after pressing SET PASSWORD soft key.



Figure 3.8: Set Password Screen

- A valid Password should consist of more than 5 characters.
 - After entering the password press SAVE soft key.
-
- **RUN DIAGNOSTICS:** Run diagnostics feature will help you to determine whether your unit is malfunctioning. When the RUN DIAGNOSTICS BUTTON is pressed a diagnosis screen as shown below is displayed and diagnostics can be run for each of the 3 paths (analog/digital/composite) as described below.

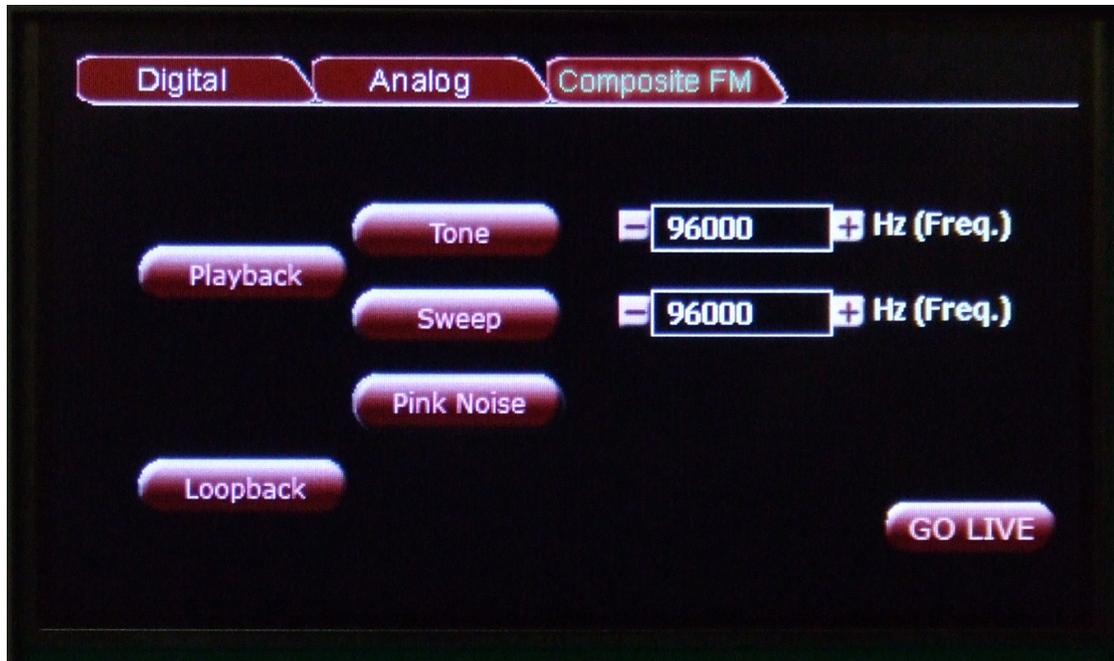


Figure 3.9: Diagnostic Screen

To diagnose each path with built in test tone follow these steps:

- ✓ Select any one of the path from Digital/Analog/Composite FM option.
- ✓ Press Playback key
- ✓ Now select option from Tone/Sweep/Pink Noise. In case of Tone, fundamental frequency can vary from 1 kHz to 24 kHz. User can also configure Start and stop frequency in case of Sweep tone.
- ✓ Loopback
- ✓ Press GO LIVE key to get back into 'Processing' mode.

Section 4

Audio Processing Operation

4.1 Overview of Available Processing Tools

In this section we take a brief look at our processing philosophy and the different signal processing blocks available in Perceptual SoundMax™ Model Q24 and provide a description of their functionality. CONFIGURE screen has the option of TUNE, which give access to all processing module at one place.

4.1.1 Preset Tuning Screen



Figure 4.1: Illustration of Processing Blocks & Controls available in Perceptual SoundMax™

In this section we take a brief look at our processing philosophy and the different signal processing blocks available in Perceptual SoundMax™ Model Q24 and provide a description of their functionality. In this description it will be helpful to keep the UI illustrated in the **Figure** above in mind; as explained in the next section this UI is also accessible through a remote networked PC interface and also same control may be applied using the front panel TFT touch panel control of the product.

Reducing the peak-to-average ratio increases the audio loudness. If one can reduce the peak levels, the average level can be increased within the modulation limits. The effectiveness with which this can be accomplished without introducing any distortion and any other objectionable sound artifacts is the main objective of the audio processing. This is one best measure to evaluate the effectiveness of the audio processor.

Compression reduces the difference in audio level between soft and loud sounds to make efficient use of permitted peak level limits. This results in a subjective increase in the loudness of soft sounds, but cannot make loud sounds seem louder. In processing, there is direct trade off between loudness and distortion. We can improve one only at the expense of other. It's wiser to give up ultimate loudness to achieve low distortion. Although such a signal might fall slightly short of ultimate loudness, it compensates this with a more open and punchy audio. To clean up audio with all these distortion's we have novel tools like Vocal Enhancement, DLFR and ICS (see below) which remove the excess stridency, harshness and distortion. These processing tools play a main role in reducing the listener fatigue reduction and thus allowing user to have a pleasant long term listening experience.

The key processing blocks in the product are described below.

4.1.1 Intelligent Loudness Control (LOUDNESS):

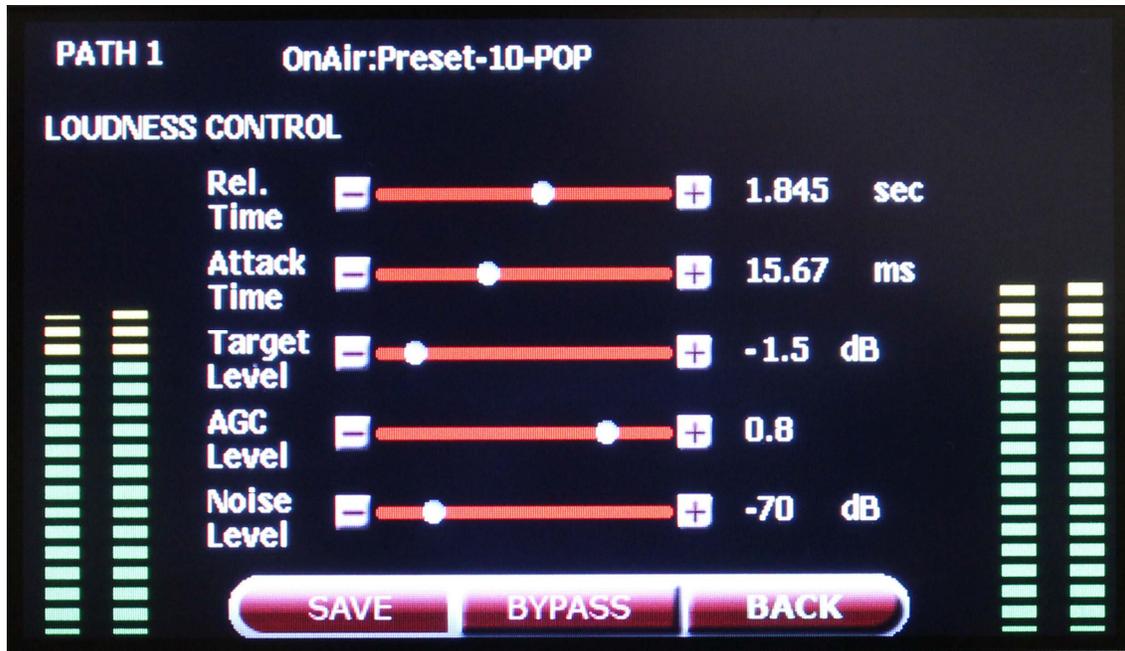


Figure 4.2: Loudness Control

This module detects the current input *stereo* loudness and controls the output *stereo* loudness depending on the two sliders value.

- The first slider is for the release time, to control how fast the audio loudness should get adapted to desired level. For fast adaptation slider should be at the left most position and vice versa.
- The second slider is for the attack time. Applied gain will last over a definite period of time defined as attack time. If attack time is set to left most side , then gain will change very instantaneously (which is not very desired as it may cause some distortion or poor quality audio).
- The third slider is for the output loudness level indication, to set the loudness 0 *dB* below the maximum level i.e., 0 *dBFS*, slider should be at the left most position, and for -15 *dB* below the maximum i.e. least allowed loudness (-15 *dBFS*); slider should be right most position.

- The fourth slider decides the maximum permissible gain of the AGC. If it is at left most side then gain is close to 1 and if its at right most side then gain will be highest. Moderate setting advised is 0.55 to 0.8 for good quality audio.

4.1.2 Accurate Multiband Dynamic Range Compression (DRC):

Multiband Dynamic Compression improves the consistency of the FM stations sound. It also remarkably increases the loudness of the program material, without producing unpleasant side effects.

To change the dynamic range in different bands individually this module is provided with greater functionality than any other product available on the market. We use a specially designed filterbank for generating a time-frequency representation of the signal that provides a very high level of control both in time and frequency domain. *Moreover, the processing filterbank is completely linear-phase, eliminating phase distortions and the need to compensate for phase distortions.*

Internal processing is in 24 processing bands. The number of bands that are available for user control is flexible; user can configure dynamic range control parameters in 2, 4, 6, 10, or 24 bands.



Figure 4.3: DRC Configuration Screen

The menu located just below the **DRC BANDS** tag is for bands selection. Appropriate numbers of sliders are selected depending on the band selection e.g., For TEN, SIX, FOUR, and TWO bands. If any slider is at bottom position then dynamic range in that band is unchanged and if the slider is at top position then dynamic range

is adjusted in corresponding band. The 10 and 6 band options is used to produce a consistent processed sound, free from undesirable artifacts. The 4 and 2 band processing is to produce partially controlled sound, which preserves the frequency balance of the original program material for lighter processing. The adjacent drop down box named as DRC BASE PROFILE is for genre selection.

- **TUNE:** With TUNE soft key user can enter inside the DRC tune option. User has flexibility to tune DRC's processing level (individual band or all frequency band together), Operating point tune with OP TUNE soft key, Compressor's threshold and attack and release time with BAND TUNE soft key.

NOTE: 24 band tuning can't be done through TFT interface. Tuning of 24 band can be done through interactive remote PC interface (Section 5). 10/6/4/2 band DRC can be fine tuned with the front-panel TFT interface as well.

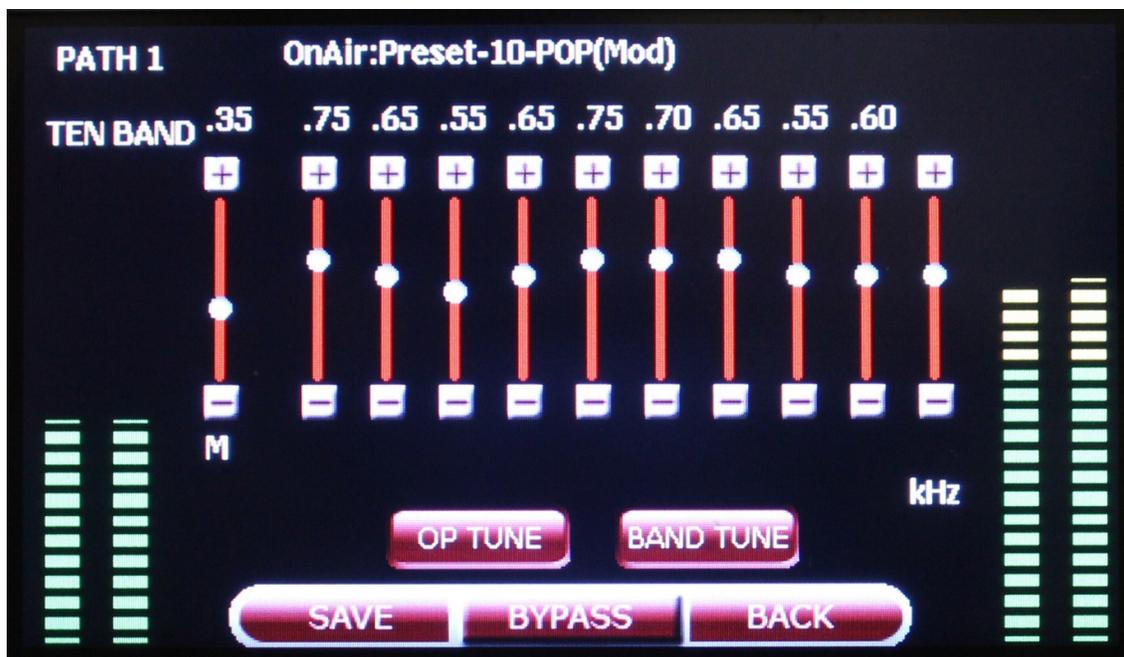


Figure 4.4: DRC Tuning Screen

The M slider above can be used for changing the all sliders simultaneously. Individual slider can be modified after selecting it with Rotary press/ touch. All slider corresponds to one frequency band and the center frequency on the band is mentioned at bottom of every slider. The level of the slider is indicated at the top of the slider. OP TUNE and BAND TUNE option is explained below.

- **OP TUNE**

Operating Points: These thresholds are the levels defined in *dB* for each band. It's the master threshold and rests of the thresholds are derived from it. Operating points are set for each genre in such a way that it will apply desired gain on particular frequency. So ideally operating points must be different for different genres. As in broadcast scenario audio content is not always mapped with right genre, we are providing floating option which will automatically the adjust operating points according to input stream. *The Operating Points can be used to boost or lower the prominence of a particular frequency region and control audio consistency in the corresponding band.*

Master Floating and Floating: If MASTER FLOTING option is checked then every band is in floating mode i.e. their operating points are calculated according to input stream. With Individual floating check, above mentioned floating operating point calculation will happen on particular band.



Figure 4.5: DRC OPTUNE Screen

- **BAND TUNE:** In BAND TUNE screen user can tune following options for one selected band:

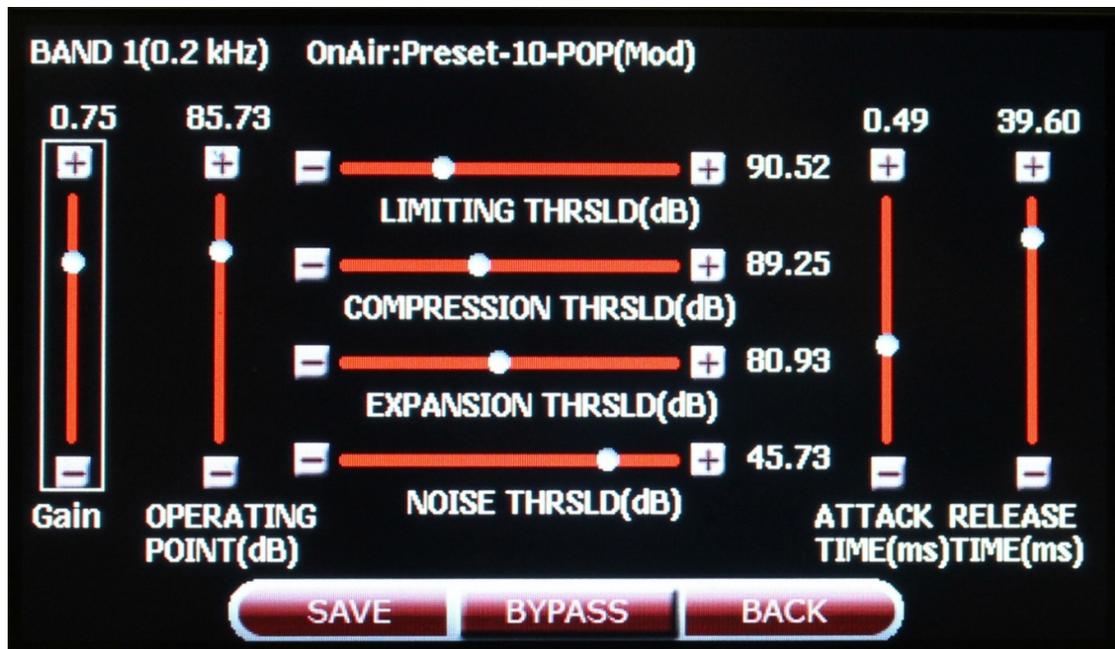


Figure 4.6: DRC BAND Tuning Screen

Gain: With this gain slider user can modify DRC intensity slider (which is explained in fig 4.4) for individual bands.

Operating Point: User can also tune individual band's operating point with this slider that is same as OP TUNE screen's operating points.

Attack and Release time: Attack and release times are associated with compression as well as expansion process. How fast a compressor/ expander react to the input signal when a threshold is crossed, is determined by attack time.

Multiband Dynamic Compression in combination with attack and release time makes different flavors like "dense", "medium" and "open". These refer to the density produced by the processing. "Open" uses a slow multi-band release time, "Medium" uses a medium slow release, and "Dense" uses medium-fast.

Noise Thresholds: This threshold is used for noise gating purpose. That is the extreme form of Expansion process. Any signal below this threshold is considered as noise and aggressively attenuated.

Expansion Threshold: Expansion is the complementary process to compression: it actually increases the level of the signal. When a signal falls below an expansion threshold level, the signal undergoes amplification.

Compression Thresholds: This threshold determines the signal level above which compression will occur. When the input signal exceeds the threshold level, the compressor is engaged and the signal undergoes attenuation. To make the effect of compression/expansion smooth and natural-sounding, compressors often allow attack and release time parameters to be set by the user, but occasionally these are fully automated.

Limiting Threshold: This threshold determines the level at which limiting will occur. When the input signal exceeds the threshold level, the signal above the threshold is limited. A higher level of compression is applied on the signal's which cross the limiting threshold.

4.1.3 Voice/Vocal Enhancement (VE):

This module enhances the speech/vocal content of the audio if the corresponding box is checked. This module uses the accurate harmonic analysis statistics of the input audio, which help in effectively enhancing the vocal content. The input audio after this processing sounds more clear and clean with improved presence.

This module is also useful in sports/news broadcast to make the commentators voice more intelligible, where sufficient background audience noise exist. For music type audio, the singer's voice will sounds more clear and punchy.

The slider is used to adjust the level of enhancement for the vocal portion of the audio. A new user can set safely a value of 0.4 to 0.7 for speech broadcast and 0.2 to 0.4 for music broadcast.

4.1.3 Multi-Band Equalizer

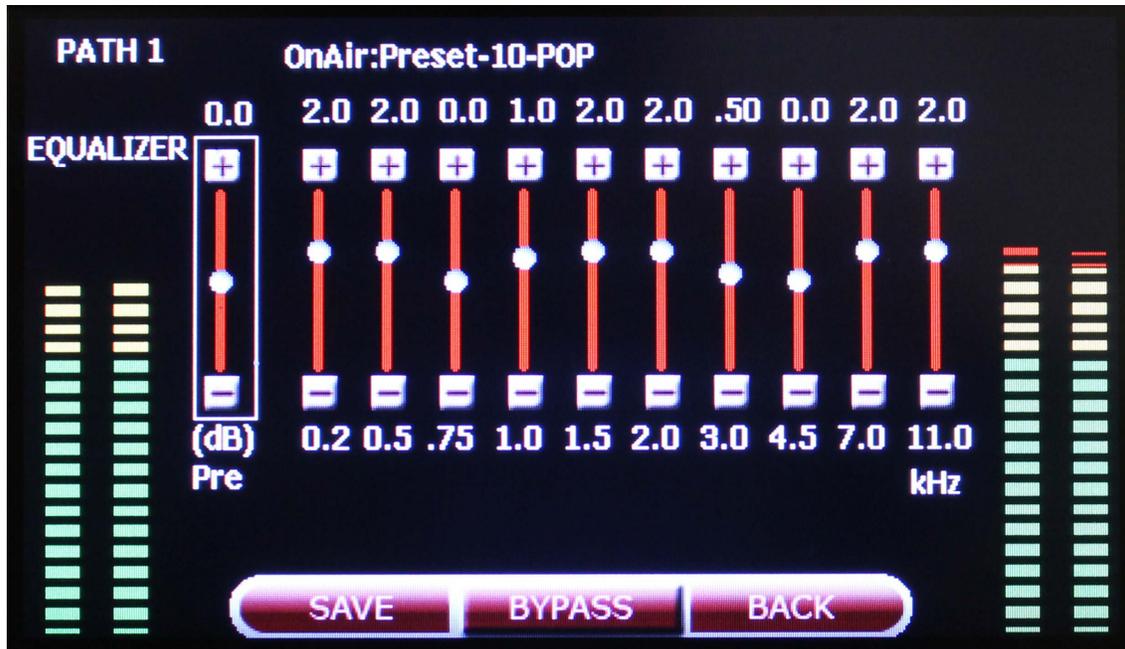


Figure 4.7: Multi-Band Equalizer

This module equalizes the audio in 10 bands according to the 10 slider setting. For all the slider at the middle of the range equalizer is effectively null. Each slider can go $5dB$ up and $5dB$ down. The reset button is used to set equalizer to transparent functionality. The 3 radio buttons are used for coupling the sliders, namely Tight Coupling, Loose Coupling and No Coupling. In Tight coupling each slider controls other sliders and has high impact on the level of controls of neighboring sliders. In Loose Coupling each slider affects the neighborhood sliders moderately. In case of No-Coupling mode each of the slider can be tuned individually.

4.1.5 Bass Enhancement

Q24 is equipped with dynamic and powerful Bass enhancement feature. The algorithm in bass enhancement is such that, it will apply a boost given by **Gain** slider on the frequency (between 30 Hz to 330 Hz) given by **Freq** slider. There is third slider **PLevel**, which will define intensity of applied Boost.



Figure 4.8: Bass Enhancement

4.1.6 Stereo Enhancement Processing (STEREO):

This module creates a new balance in the stereo. It widens or shrinks the stereo image depending on the slider setting. The scale is divided between 0.0 and 1.0. The higher the value of this slider, the more balanced audio we have with increased stability in 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 dial at 0.3.



Figure 4.9: Stereo Processing

Monodownmix: With this option output can be streamed as mono. User can select any of the Mono type from check box named as

- SUM
- LEFT
- RIGHT

4.1.7 Wideband Automatic Noise Removal (WANR):

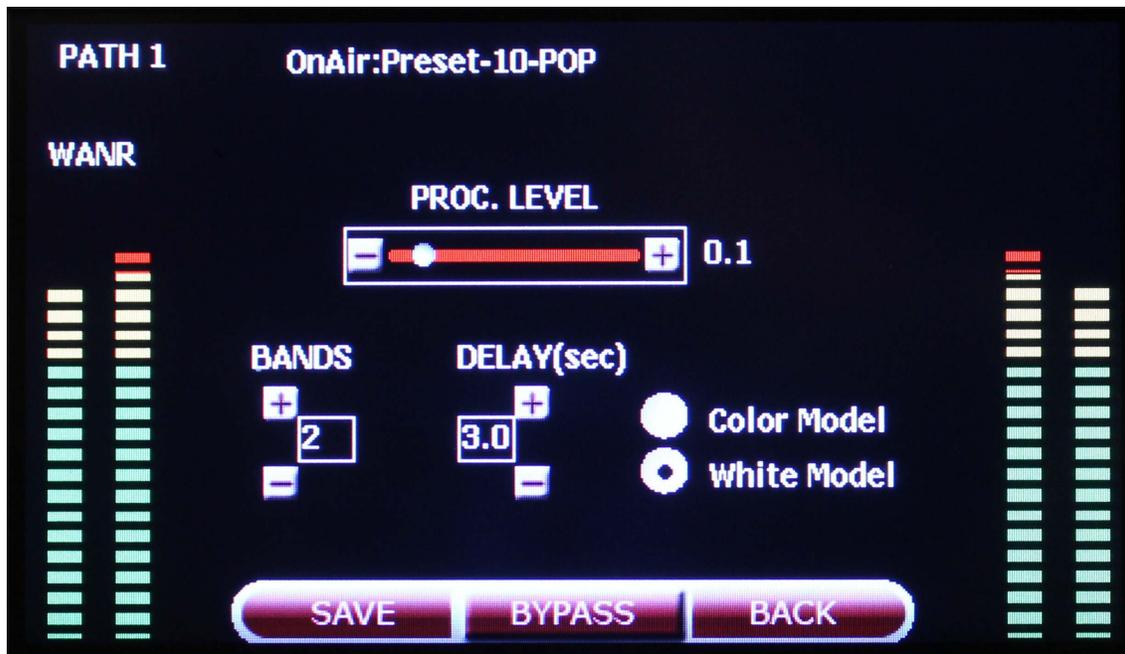


Figure 4.10: Wideband Automatic Noise Removal Settings

Model Q24 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, the White Noise model is recommended and the dial runs values from 0.0 to 1.0. A user can choose the level of noise filtering by varying the value of the dial. An increasing dial value increases the level of noise filtering in the audio. A user who is uncertain of the level of noise for an input can conveniently leave the dial in the 0.2-0.4 range.

Color Model: For heavy noise reduction of a particular environment such as machine and instruments noise type, the more complex Color Model is recommended and the dial once again runs values from 0.0 to 1.0. A user can choose the level of noise filtering by varying the value of the dial. An increasing dial value increases the level of noise filtering in the audio. A user who is uncertain of the level of noise for an input can conveniently leave the dial in the 0.2-0.3 range.

4.1.8 Dynamic Listening Fatigue Reduction (DLFR):

This is an audio *sweetening* tool that conditions the temporal envelope of the audio signal in multiple frequency bands. To smooth the audio the dial value should be at high end and to let in more envelope dynamics dial value should be at lower end.

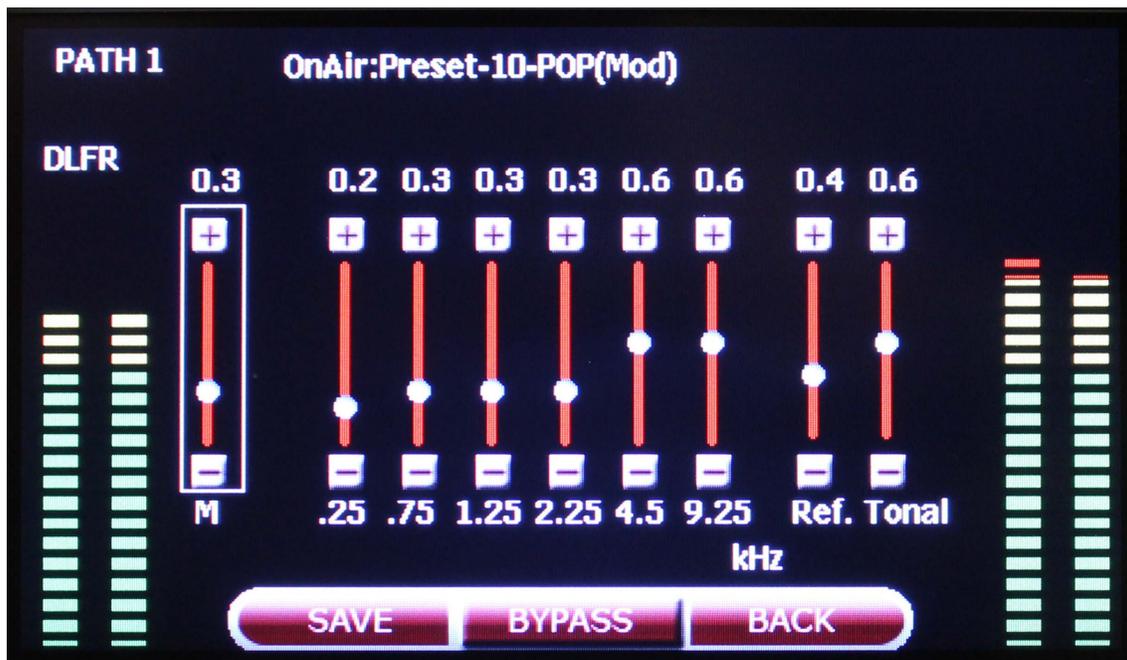


Figure 4.11: Dynamic Listening Fatigue Reduction Tool

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 smoother is the audio in that band. There are two associated slider controls which control respectively the tonal characteristics of overall audio and the refinement level of envelope conditioning. For a new user, good initial setting for all the DLFR controls may be found by selecting the “Music-General-purpose-Light”

preset profile from the profile menu. The other controls like WANR and ICS etc may then be modified while maintaining these values for DLFR control.

4.1.9 Incoherent Component Suppression (ICS):

This module detects the incoherent sections of the input audio and smoothes 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.



Figure 4.12: ICS Settings

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.

4.1.10 Sibilants and Fricative Smoothing (SFS):

Sibilants tend to add to the some of the unwanted harshness in the broadcast audio since the perceived loudness of sibilants is quite high due to their high

frequency content. This module dynamically detects the sibilant, fricative sections of the input audio and smoothes the same form that audio section only.



Figure 4.13: SFS Settings

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.

4.1.11 Final Look-Ahead Limiter

This module determines absolute maximum of the output. It ranges from 0dBFS to -20dBFS. Model Q24 uses a sophisticated look-ahead limiting algorithm that ensures output will not exceed the full scale target set by this control and at the same time hard-limiting will be avoided. It also works as software level controller for Digital Output path. The control interface for the limiter on the TFT is as below.



Figure 4.14: Final Look Ahead Limiter Settings

Section 5

Remote Control Using Networked PC

Perceptual SoundMax™ Model Q24 allows for elaborate control of its operation through either its front panel TFT touch screen control or through a software GUI interface running on a networked PC connected to the same Ethernet as the processor. Below we describe both these options in some detail.

5.1 Remote Control – Networked PC Interface

Model Q24 supports an Ethernet port to connect to a local network. The primary utility of this is to allow for remote control using a *Controller Software* called **SoundMax Manager** running on any PC present on the same Ethernet. The presence of one or more Perceptual SoundMax™ processors on the network is automatically sensed once the Controller is run on a PC. In other words, the Controller can come up anywhere on the local network of the existing system and be used to monitor and control several hardware preprocessors running on the network. There Controller can optionally be password protected.

There are three different views to the SoundMax Manager GUI which have different functionality, scope and granularity. The master view shows the available Q24 processors on the network, along with the health of each processor. Any problem within a processor is indicated with a warning sign. In such a situation, the processor tries to recover from the issue automatically. A detailed listing of processing paths available on each processor will appear in the second view of the Controller. The processing paths are shown with the input/output and profile selection; the user can choose a different profile from the available options from this screen. Figure 5 illustrates Screens 1 & 2 of the SoundMax Manager.

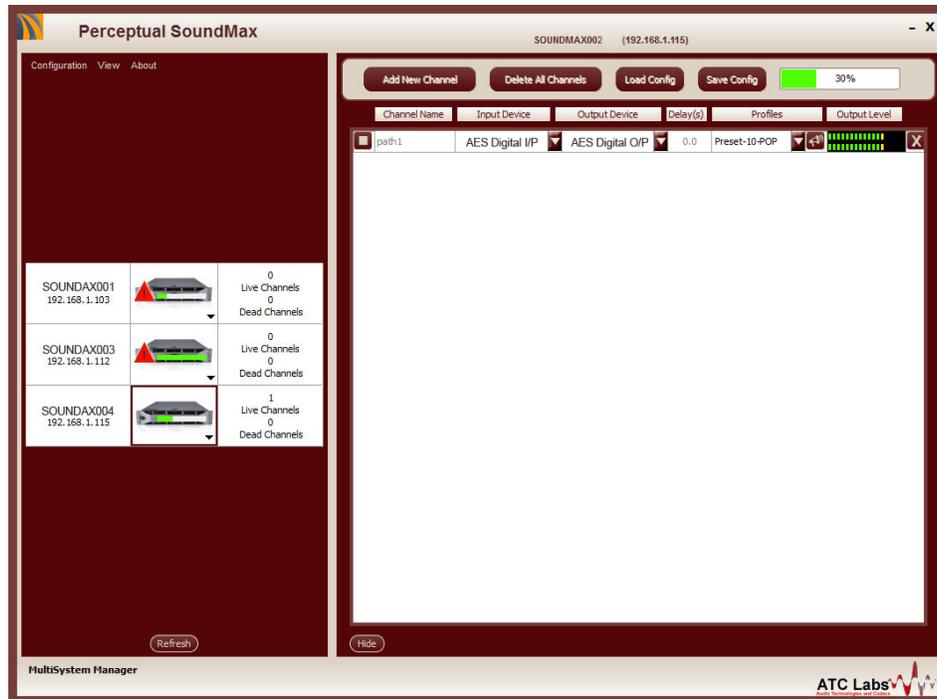


Figure 5.1: Screen 1 and 2 of the SoundMax Manager Software running on a Networked PC

5.2 Guide to Creating Custom Processing Profiles using SoundMax Manager

As described in Section 2.4.1 a tune screen opens up when the “tune” button on the second screen is pressed. With the tuning screen user can use all the modules described in previous section. We are describing all the information assuming that user is operating the Model Q24 station through remote PC only. As described above Model Q24 incorporates different processing tool sets:

- Dynamic Range Compression (DRC),
- Intelligent Loudness Control,
- Wideband Automatic Noise Removal (WANR),
- Envelope/Stereo Processing,
- Voice/Vocal Enhancement (VE),
- Base Enhancement(BE),
- Sibilant/Fricative Smoothing (SFS),

- Incoherent Component Suppression (ICS),
- Dynamic Listening Fatigue Reduction (DLFR),
- Multi-Band Graphic Equalizer (Equalizer),
- Pre-Emphasis, and
- Low Pass Filtering.

These different tools when managed judiciously can provide crystal clear, pleasant, noise free high quality audio and offer the user flexibility for creating a customize *signature* sound. However, an expert’s work is not required to handle the product satisfactorily. The product is designed with a simple and easy to use interface and it should not take more than few minutes for one to master the essential controls after the software installation.

5.3 Tuning Guide

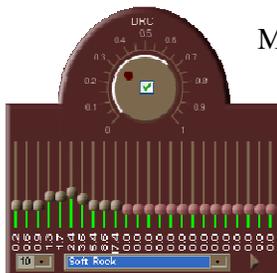
This chapter briefs the use of different tools by explaining their functionality.

- **Bypass Processing:**



By checking the bypass processing button the processing done in the Q24Soft is bypassed. This helps the user judge the quality of audio before and after processing instantaneously by switching between modes.

- **DRC:**



Multiband Dynamic Compression is an integrated 24-band Dynamics Processor. Driven by the principles of Psychoacoustics, it employs completely linear phase signal processing. The integrated processing engine performs compression, limiting, expansion and noise-gating in a single processing block. 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.

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

Model Q24 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.

- **Intelligent Loudness Control:**



This module detects the current input stereo loudness and controls the output stereo loudness depending on the four sliders value. The first slider is for the release time, to control how fast the audio loudness should get adapted to desired level. For fast adaptation slider should be at the left most position and vice versa. The second slider is for the output loudness level control, to set the loudness 0 dB below the maximum level i.e., 0 dBFS, slider should be at the left most position, and for -15 dB below the maximum i.e. least allowed loudness (-15 dBFS); slider should be right most position.

- **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 the some of the unwanted harshness in the broadcast audio since the perceived loudness of sibilants is quite high due to their high frequency content.

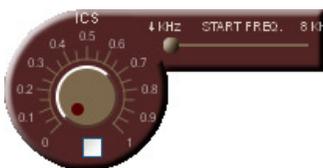
- **Dynamic Listening Fatigue Reduction (DLFR):**



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 smoothes the corresponding audio component resulting in a sound that is easier to process by the human auditory perception system, thus

increasing the listen 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 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.

- **Multi-Band Graphic Equalizer:**



This module equalizes the audio in 10 bands according to the 10 slider setting. For all the slider at the middle of the range equalizer is effectively flat. Each slider can be moved between -5dB and 5dB. There are three options in the form of buttons, namely Tight Coupling, Loose Coupling and No Coupling, supported for coupling the sliders.

- **Available Low Pass Filter:** Set any limit between 4-20 kHz



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



- **Bass Enhancement:**



- **Final Limiter:**



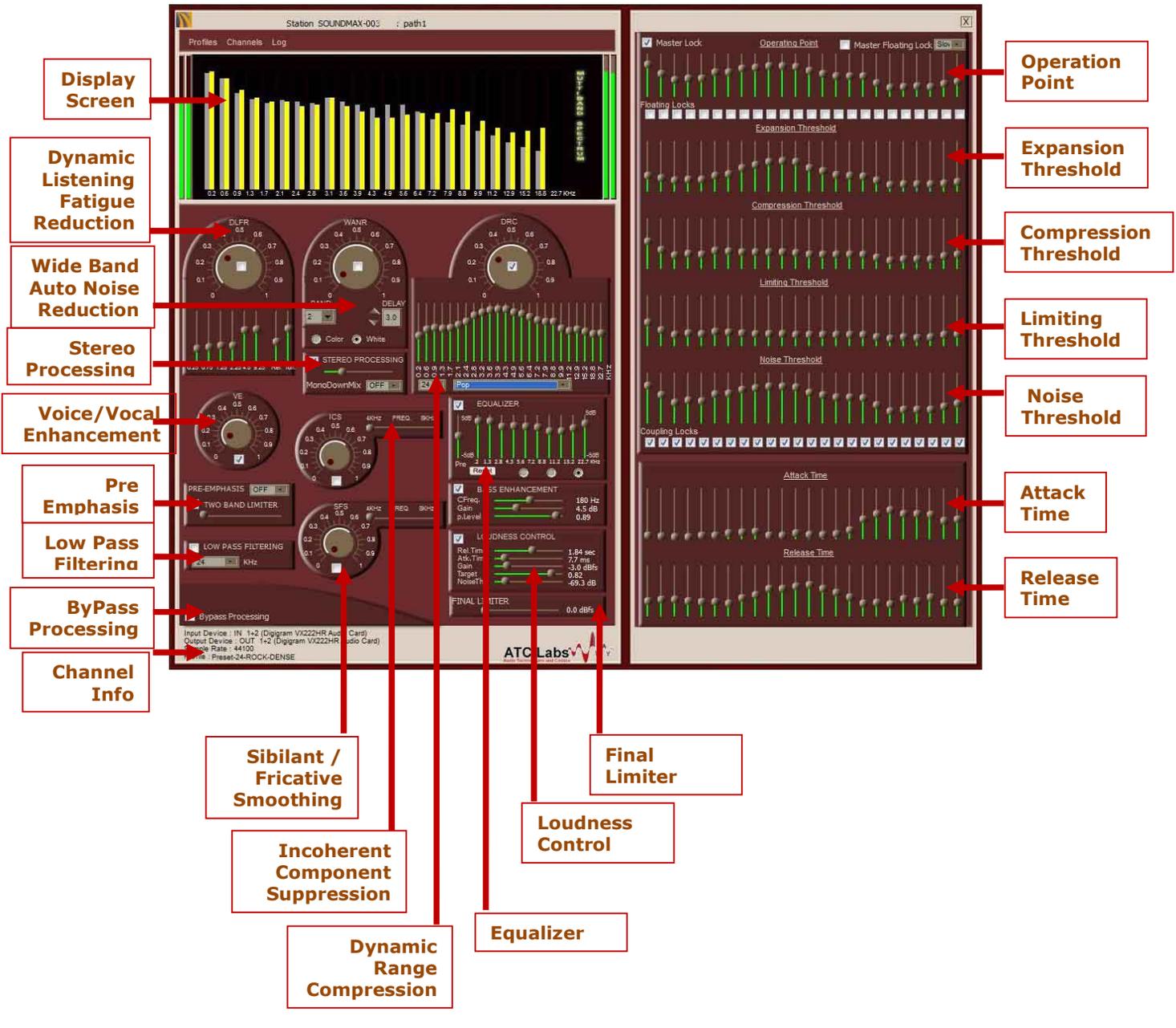


Figure 5.2: Perceptual SoundMax Manager Tuning Screen

Section 6

Preset Processing Profiles and Custom Profiles

6.1 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.
- **ROCK:** The ROCK presets are designed for bright high and punchy low end. A midrange boost provides enough presence energy to ensure that vocals stand out.
- **JAZZ:** JAZZ is 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-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.
- **SPORTS:** Similar to NEWS-TALK except the Threshold is lower.
- **COUNTRY:** It has a gentle bass lift and a mellow, easy-to-listen-to high end, along with enough presence energy to help vocals to stand out.

- **DANCE:** This preset is designed to preserve the punch and slam in dance music percussion (such as the beater click in kick drums). It is loud and has a bright high frequency texture.
- **LIGHT INSTRUMENTAL:** An alias for the JAZZ preset.
- **HIP HOP:** Similar to ROCK but having less compression.
- **COMEDY:** Similar to NEWS-TALK but having less compression.

6.2 Custom Profile Creation

As mentioned above a user may start from a factory preset and then tune parameters associated with one or more processing blocks to create a *signature on-air sound*. Alternatively user may want to invoke a processing block which was previously unused. These tasks can be accomplished using either the TFT touch 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 TFT or Controller and available as an option in future.

Section 7

Hardware Specifications

Installation

Analog Audio Input:

- Configuration:** Left and Right/Stereo
- Maximum Input Level/Impedance:** +24dBu/ >10k Ω
- Nominal Input Level:** Adjustable -10.0 to +12.0 dBu
- A/D Convertor Resolution:** 24-bit
- EMI Suppressed:** Yes
- Connectors:** Balanced XLR Female

Analog Audio Output:

- Configuration:** Stereo
- Output Level:** Adjustable -6.0 to +20 dBu
- SNR:** > 100 dB (20 Hz to 20 kHz)
- Crosstalk:** 20Hz to 20 kHz < -100 dB
- Distortion:** THD < 0.01%
- D/A Convertor Resolution:** 24-bit
- EMI Suppressed:** Yes
- Connectors:** Balanced XLR Male

Digital Audio Input (AES/ EBU):

- Configuration:** Stereo
- Sampling Rate:** 32, 44.1 or 48kHz (Auto Sense)
- Word Length:** 16 bit or 24 bit
- Input Reference Level:** -25.0 to 0 dBFS
- EMI Suppressed:** Yes
- Connector:** AES– XLR Female

Digital Audio Output (AES/EBU):

- Configuration:** Stereo
- Output Level:** -20.0 to 0.0 dBFS
- Sampling Rate:** 32, 44.1, 48kHz (User Selectable)

EMI Suppressed: Yes

Connector: AES – XLR Male

Composite Baseband Output:

Configuration: Two Outputs with Individual Trim Control

Maximum Output Level (unbalanced): +12dBu

D/A Conversion: 24 Bits

Signal to Noise Ratio: 108dB @ 48kHz (A-Wtd), > 100 dB, 20Hz-20kHz

Dynamic Range: 109dB @ 48kHz (A-weighted)

THD: < 0.01% (20 Hz to 20 kHz)

Stereo Separation: >100 dB

Cross Talk: -105 dB, channel to channel

38kHz Suppression: > 90 dB

Pilot Protection: 65 dB relative to 9% injection, +/- 500 Hz

57 kHz (RDS) Protection: >60 dB relative to 4% injection, +/- 2 kHz

Connectors: Coaxial BNC, EMI-Suppressed

Power

Power Requirements: 105-260 V AC Single Phase, 45-60 Hz

Connector: IEC, EMI Suppressed, Detachable 3-wire chord with earthing.

Safety Standards: UL, CE

Ethernet & Remote PC Control:

Configuration: Auto Presence Broadcast on Ethernet

Connectors: RJ-45, EMI-Suppressed

Control: Controller Software (SoundMax Manager) on any networked PC
for auto sense monitoring and control.

USB Connection:

Connector: 1 USB 2.0 (EMI-Suppressed)

Weight: 5.5 Kg

Dimensions:

Width: 19"

Height: 2.5 U (4.37")

Depth: 20"

Operating Temperature range: 0° to 50° C

Relative Humidity: 95%



Figure 6.1: Model Q24 in Rack

Contact Us:

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