LMS608

Loudspeaker Management System

User's manual

February 2020

Safety Instructions

1. All safety instructions must be read before using this device.

2. Keep and follow these instructions

3. Heed all warnings
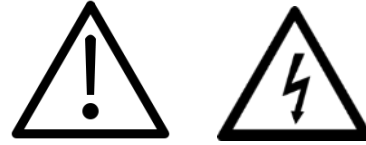
4. The exclamation mark in the triangle indicates internal components which if replaced can affect safety.

5. The lightning symbol within the triangle indicates the presence of dangerous uninsulated voltages.

6. Only clean the device with a dry cloth.

7. Do not block any ventilation openings. Install in accordance with the manufacturer's instructions.

8. Do not install the device near heat sources such as radiators, heaters or other heat-emitting elements.

9. Protect the power cord from being walked on or pinched, particularly at plugs, convenience receptacles, and the point where they exit from the apparatus

10. The equipment must be repaired by qualified technical service personnel when:
A. The mains supply cable is damaged, or
B. Any object or liquid has damaged the device; or
C. The equipment does not function normally or correctly; or
D. The equipment has been exposed to the rain; or
E. The chassis is damaged

11. Disconnect the device in the case of electric storms or during long periods of disuse.

12. WARNING – To reduce the risk of fire or electric shock, do not expose this device to rain or moisture

13. The equipment shall not be exposed to dripping or splashing and no objects filled with liquids, such as vases, shall be placed on the device.

14. For hanging and installation, use manufacturer recommended accessories only.

1 INTRODUCTION

The **LMS**608 is a complete digital loudspeaker management system designed for the touring or fixed sound installation markets. The absolute latest in available technology is utilized with 64-bit floating point processors and high performance 24-bit Analogue Converters. The high-bit DSP prevents noise and distortion induced by truncation errors of the commonly used 24-bit fixed-point devices. A complete set of parameters include I/O levels, 2 second-delay per channel, polarity, 10 bands of parametric EQ per channel, multiple crossover selections, full function compressor and peak limiter. Precise frequency control is achieved with its 1 Hz resolution. Inputs and outputs can be routed in multiple configurations to meet any requirements.

The **LMS**608 can be controlled or configured in real time on the front panel or with the intuitive PC/Mac GUI accessed via the USB interface or Ethernet. Software upgrade for CPU and DSP via PC keeps the device current with newly developed algorithms and functions once available. Multiple setup storage and system security complete this professional package.

Shipped contents:

- **LMS**608/608D unit
- AC power cord
- USB cable for PC connection

2 FEATURES

LMS608
- Electronically balanced inputs
- Matched-impedance balanced outputs
- 64-bit floating point DSP
- High Performance 24-bit A/D Converters
- 1 Hz Frequency Resolution
- 10 Parametric Equalizers for each Input and Output (EQs can be set as Bell, Notch, High Shelf, Low Shelf, Notch, Allpass, Band Pass, High Pass, Low Pass)
- Multiple Crossover types: Butterworth, Bessel, Linkwitz-Riley, up to 8th order (48dB / oct).
- Up to 2 seconds delay per each input/output
- RMS compressor and ultra-fast attack Peak Limiter.
- Precise Level, Polarity and Delay
- 2-Line x 16 Character Blue Backlit LCD Display
- Signal LED's on every Input and Output
- Security Lock
- USB Interface for PC/Mac Control and Configuration (on front panel)
- CPU and DSP firmware upgrade via PC/Mac interface
- 4 Analog Inputs, 2 Digital Inputs (AES/EBU) and 8 Outputs with flexible routing.
- 125dB dynamic range (inputs) / 120dB dynamic range (outputs)
- 96kHz sampling rate
- Very low latency (0.35ms)

- Ethernet connection (RJ45, on back panel)
- Storage of up to 100 Program Setups

3 MAIN CONTROLS AND CONNECTIONS

3.1 The front panel


A) **Mute keys** – Press for Mute/Unmute input and output channels. When a channel is muted, this key will light up in red for indication.

B) **Input signal LEDs** - Show the current level of the Signal: Signal (-36dBu), -6dBu, Limit (orange), Clip (red). The Limit led lights up if a gain reduction is taking place (due to a programmed compressor). The Clip LED references to the device's maximum headroom (+24dBu).

C) **USB Connector** – A standard type B USB connector for interface with a PC or Mac. Software and driver must be installed prior to usage.

D) **Menu Control keys** - There are 6 menu keys: *<Channel* (Previous Channel), *Channel>* (Next Channel), *<Select* (Previous Option), *Select>* (Next Option), *Menu* and *Exit*. The functions of each key is explained below:

<Channel:	Choose previous channel for editing.
Channel>:	Choose next channel for editing
<Select:	Select previous parameter for editing
Select>:	Select next parameter for editing
Menu:	This key has different functions depending on when it is used: - In the Main Screen: Access the System Menu. - In the Main Menu allows entering the System Menu.
Exit:	Exit to the Main Menu

E) **LCD** - Shows all the necessary information to control the unit.

F) **Rotary Thumb Wheel** – Turn the wheel to change parameter data values. Click on it to confirm the value entered. The center click of the wheel is also use to browse different parameters of the same feature.

G) **Output signal LEDs** - Show the current level of the output Signal: Signal (-36dBu), -6dBu, Limit (orange), Clip (red). The Limit led lights up if a gain reduction is taking place (due to a programmed compressor). The Clip LED indicates a gain this reduction is higher than 12dB.

3.2 The rear panel
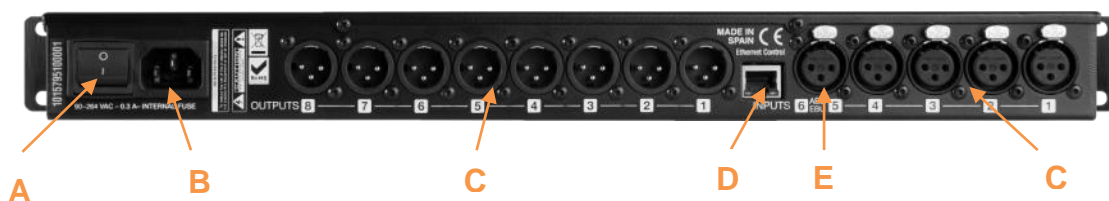
A) **Power switch** - Controls power On/Off.

B) **Main Power** - Connects via a standard IEC socket. A compatible power cord is supplied with the unit. The input voltage range is 85 to 240VAC, 50-60Hz.

C) **XLR input and outputs** - Separate 3-pin XLR connectors are provided for each audio input and output. The device's output stage employs the balanced impedance topology. All I/O connectors have pin 1 as ground (shield), pin 2 as + and pin 3 as -.

D) **Ethernet port** – RJ45 connector to connect with the PC via Ethernet. Use CAT-5 or CAT-6 cable.

E) **AES/EBU Input** – This XLR connector accepts a 2-channel digital input in AES/EBU or S/PDIF format, 16 or 24-bit. Supported sample rates are 44.1, 48, 88.2 and 96 kHz.

4 QUICK INSTALLATION

4.1 Before you start

Before powering up the unit, make sure that the input and output XLR cables are in good state and following the following pinning diagram: 1 for shield, 2 for live(+), 3 for return (-) as defined by the AES14 standard.

Do not connect the unit to your computer before installing the software and the USB Driver. Please refer to installation instructions in this manual.

When connecting the **LMS**608 to the amplifiers, mute the **LMS**608 outputs (or turn down the amplifier's volume knobs) until you configure your processing. Loudspeakers may be damaged due to a wrong setup. It is advisable to unmute first the high frequency channels: in case they are connected to low frequency drivers by mistake, they cannot be damaged. Otherwise, high frequency speakers may be damaged because of trying to reproduce low frequencies.

4.2 Power Up

After powering up the unit, all LEDs will be lit for about six seconds, while the following message is displayed on the LCD:

```
Initialising...
```

After that, the **LMS**608 unit displays its main screen:

```
AMATE AUDIO
DEFAULT PRESET
```
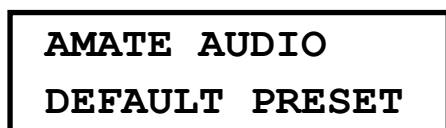
Now the **LMS**608 unit is ready to operate. The screen shows the name of the unit (above line) and the program name currently active (below line). The program assigned is always the last program the user recalled or stored before powering down the unit.

An admiration mark (!) beside the program name means that the program has been modified but not stored. This symbol does not appear when the modified program is the default preset.

```
AMATE AUDIO
! My_Preset
```
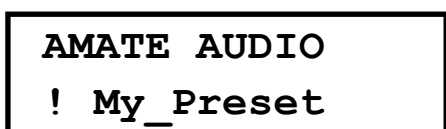
4.3 1 Control Software

The **LMS**608 units are shipped with a Computer Software which provides a Graphic User Interface (GUI) application - DSPLink. DSPLink allows the user to control the **LMS**608 unit from a computer via the USB communication link or through Ethernet, in models. For the USB connection, a driver must be installed (included with the DSPLink installation package).
The GUI application makes it much easier to control and monitor the device, allowing the user to get the whole picture on one screen. Programs can be recalled and stored from/to Computer's hard drive, thus expanding the storage to become virtually limitless. See Section 6 of this manual for instructions to operate the software.

DSPLink is available for PC and Mac. A USB pen drive or CD-ROM with the software is included in the original package of the unit. Check Amate Audio's website for a latest version download (www.amateaudio.com/DSPLink).

Installation for PC-Windows:
Double click the installation file, depending whether you have a 32 or a 64-bit system:
Amate_Audio_DSPLink_32_bit_v_10_X_X_BuildNr.msi
Amate_Audio_DSPLink_64_bit_v_10_X_X_BuildNr.msi
Follow the on-screen instructions. You may be prompted to install the Microsoft Visual Runtime libraries before finishing the installation. Click the checkbox to proceed with this action.

Installation for Mac:
First unzip the provided installation file:
Amate_Audio_DSPLink_v_10_X_X_BuildNr.mpkg.zip
Then select the .mkpg file with CONTROL+Click and select "Open". Then follow the on-screen instructions.

4.4 Connecting the device to a computer

4.4.1   USB

WARNING: Always install the software package DSPLink before connecting your unit to the computer. See previous section for details.

After installing DSPLink, please use the provided USB cable to connect the unit to your computer. The first time you connect a device, the system will ask you to look for the driver. Choose the option "do not look for updates and to automatically select the best driver".

Run DSPLink and the unit will be automatically detected.
Click on the arrow next to the detected unit to open the window for parameter edition.

4.4.2   Ethernet

Use a standard CAT5 or CAT6 cable with RJ45 connectors (not supplied) to connect the unit to your Local Area Network. The most convenient way is to connect it to a router that provides automatic IP addresses (DHCP enabled), and that is sharing the same network with the computer in which you are running DSPLink. In this case the unit will be detected and configured automatically.

For direct connection with a computer, or through a switch not supporting DHCP, then you can either let your computer get an automatic IP of the ZeroConfig range, or assign a fixed IP to it:

To obtain an automatic IP in the ZeroConfig range:
1. Connect the unit to the switch or computer. The computer network card's settings must specify "Obtain an automatic IP" (by default)
2. DSPLink will detect the unit through its MAC address.
3. After some time (it may take a couple of minutes), the system will assign the address and notify the message "This connection has limited or no connectivity".

To set a fixed IP in the ZeroConfig range:
1. Open your Network Connection settings and edit TCP/IP address
2. Set the IP in the ZeroConfig Range (169.254.0.0 – 169.254.255.255)
3. Set a Subnet Mask as 255.255.0.0.
4. Other parameters such as Gateway or DNS Servers may be left blank.
5. Connect the unit to the network where your computer is connected to.
6. DSPLink will detect the unit

LMS608   Feb 2020

4.5 System Optimization

In order to have a proper start of your system and an optimized configuration, follow these steps by the first configuration:

1. Play a signal at the nominal level from your mixing desk, and set the input gain of your processor to 0dB (Default Preset setting)
2. Set the crossovers that you want to use, while keeping the output gains also at 0dB.
3. With DISCONNECTED loudspeakers, turn up the volume of the power amplifiers entirely clockwise (full volume).
4. Reduce the output gain and / or the output limiter setting to get the desired gain, so that the amplifier is just clipping and the built-in limiters of your processor are just limiting (Orange led blinking at the output channels).
5. Turn down the volume of the power amplifiers, connect your speakers, and slowly increase the volume while checking the sound.
6. Check if the loudspeakers are reproducing distortion-free sound, and the limiter LEDs are flashing or off, but not continuously on. If they are continuously on, reduce the output gain of your processor.
7. If you cannot reach enough signal level, increase the processor's input gain or turn up the level from your mixing desk.

10   Amate Audio

4.6 Typical application example

The following image depicts a possible application diagram, with 3 independent audience zones.
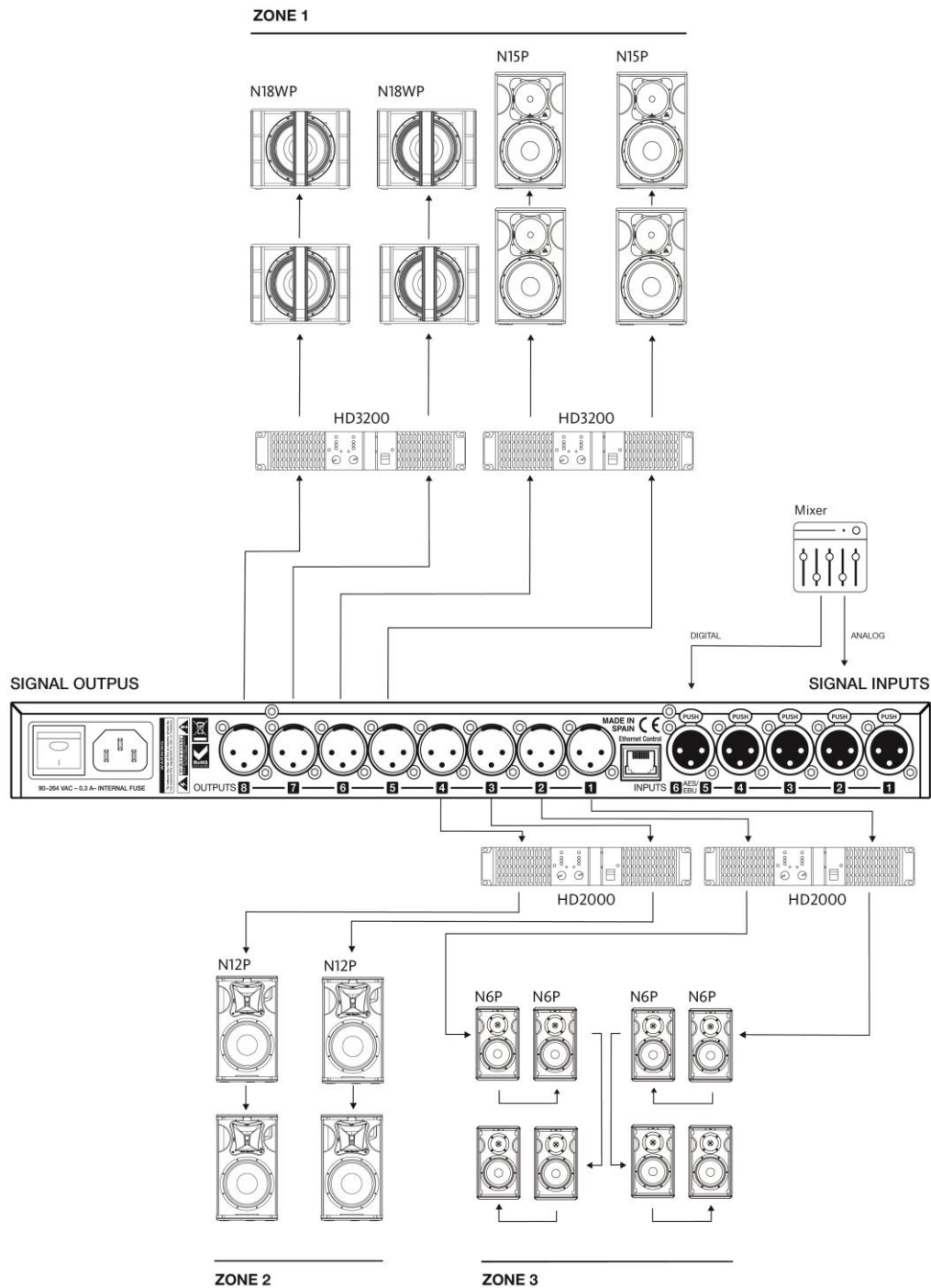
5 OPERATING THE DEVICE

5.1 Input Menus

To access the Input Menus press the *<Channel* or *Channel>* key. Press *Exit* to finish editing or again *<Channel* or *Channel>* to edit other channels.

The following menus are available for each input channel. Please note that by default the inputs and outputs channels are linked in groups of two. The following options are shown for the input group composed of In1 and In2

5.1.1   Input Gain

Adjust the gain of each input, using the rotary knob.

```
In 12    Gain
     -1.25dB
```

5.1.2   Input Selection

Inputs 1&2 are always assigned to the analog inputs 1 and 2, while inputs 3&4 can be assigned to the analog or digital source (AES/EBU).Turning the rotary knob, the source can be selected.

```
In 34    Source        In 34     Source
   Analog                 AES Stereo
```

5.1.3 Delay

Adjust the delay by turning the rotary knob. Push this knob to switch the unit between ms, ft or m.

```
In 12  Delay
      0.090ms
```

5.1.4 Crossover

• **Low Pass** – First adjust the Low Pass Filter cut-off frequency. To switch it off, turn the rotary knob clockwise until frequency reaches 20 kHz.
Second, press the rotary knob and select from the available types and slopes:
Type: Butterworth, Slopes: 6dB, 12dB, 18dB or 24dB
Type: Bessel. Slopes: 6dB, 12dB, 18dB or 24dB
Type: Linkwitz-Riley. Slope: 12dB or 24dB

```
In 12    LowPass        In 12     LowPass
Freq:    15000Hz        Type: BUT 24dB
```

• **High Pass** – First adjust the High Pass Filter cut-off frequency. To switch it off, turn the rotary knob counter-clockwise until frequency reaches 20 Hz.

Second, press the rotary knob and select from the available types and slopes:
Type: Butterworth, Slopes: 6dB, 12dB, 18dB or 24dB
Type: Bessel. Slopes: 6dB, 12dB, 18dB or 24dB
Type: Linkwitz-Riley. Slope: 12dB or 24dB

```
In 12    HighPass          In 12    HighPass
Freq:    40Hz              Type:    BUT 24dB
```

5.1.5   Parametric EQ

Select between one of the 10 available Equalizers on each input channel by using the keys < *Select* and *Select* >. Browse the parameters by turning the rotary knob and push the knob to select and confirm the values. The following parameters can be adjusted for each EQ:

• **Freq** - EQ center frequency.
• **Gain** - EQ level gain.

```
In 12    PEQ 1             In 12    PEQ 1
Freq:    1000Hz            Gain:    0.00dB
```

• **Q** - EQ Bandwidth. For shelving filters the Q sets the transition in dB/Oct. See Annex[1] for further information about the Q factor implementation in the **LMS**608.
• **Type** - Shape of EQ. The available types are:
Bell: Modifies the gain of a certain frequency range, with bell shape
Notch: Eliminates a range around a center frequency
Low-Shelf: Modifies the gain of all the range below a selected frequency.
High-Shelf: Modifies the gain of all the range above a selected frequency.
All Pass: Modifies the phase response, without influencing the frequency response.
Band Pass: Filters out all the range except the defined band.
High Pass: Filters out all the range below a certain frequency.
Low Pass: Filters out all the range above a certain frequency.

```
In 12    PEQ 1             In 12    PEQ 1
Q:  2.36                   Type:    Bell
```

• **Enabled** – When set On, the currently selected EQ is on line.

```
In 12    PEQ 1
Enabled: On
```

5.1.6   Dynamics Compressor

A true RMS compressor can be set to avoid the input signal go above a certain RMS value. The following parameters can be adjusted:

• **Thr.** - Compressor Threshold. Sets the level at which the compressor will activate.
• **Att.** - Attack time. Time it takes the compressor to start actuating after reaching the threshold.

```
In 12    Compr.:
Thr.: 24.00dBu
```
```
In 12    Compr.:
Att.:  20ms
```

• **Hold** - Hold time. Sets up a delay before the compressor enters the release cycle. Useful for compressing low frequency long notes.
• **Rel.** - Release time. Time the compressor uses to return to unity gain after the signal is below the threshold.

```
In 12    Compr.:
Hold:   10ms
```
```
In 12    Compr.:
Rel.: 200ms
```

• **Ratio** - The compressor ratio determines the slope in which the signal is compressed. The higher this value is, the higher the compression.
• **Gain** – The compressor make up Gain. Use it in case the level of the compressed signal must be corrected (0dB by default)

```
In 12    Compr.:
Ratio: 5.00:1
```
```
In 12    Compr.:
Gain: 1.00dB
```

5.1.7 Limiter

A peak-limiter can be set at each input. It is a zero attack time limiter, so it will immediately act on the signal. The parameters that can be changed are:

• **Thr.** – Threshold: input level at which the signal will be limited.
• **Rel.** – The release value, expressed in dB/seconds

```
In 12    Limiter
Thr.: 24.00dBu
```
```
In 12    Limiter
Rel.: 50
```

5.1.8 Phase – Phase inversion

Change the polarity of the channel by a 180º phase inversion. Select between Normal or Inverted.

```
In12   Phase:
Normal
```

5.1.9 Channel Link

Input channels 1 & 2, and 3 & 4 can be linked in order to set the same parameters on both. The factory setting is that channel 1 and 2 are linked and 3 and 4 too. Turn the rotary knob set the link off.

```
In 12    Link:
On
```

5.2 Output Menus

To access the Input Menus press the *<Channel* or *Channel>* key. Press Exit to finish editing or again *<Channel* or *Channel>* to edit other channels.

The following menus are available for each output channel. Please note that by default the inputs and outputs channels are linked in groups of two. The following options are shown for the output group composed of Out1 and Out2

5.2.1 Input Gain

Adjust the gain of each output, using the rotary knob.

```
Out12  Gain
     0.50dB
```

5.2.2 Mixer

Select the level to be routed from each input by turning the rotary knob. Select 0dB for maximum level of an input. To disable one input, turn the rotary knob counter-clockwise until "Off" is displayed. Push the rotary knob to select the next input. This menu is only shown when channels are unlinked. The settings are kept after linking the channels again.

```
Out1    Mixer
Input 1  0.00dB
```
```
Out1     Mixer
Input 2 Off
```

5.2.3 Delay

Adjust the delay by turning the rotary knob. Push this knob to switch the unit between ms, ft or m.

```
Out12  Delay
     0.000ms
```

5.2.4 Crossover

• **Low Pass** – First adjust the Low Pass Filter cut-off frequency. To switch it off, turn the rotary knob clockwise until frequency reaches 20 kHz.

Second, press the rotary knob and select from the available types and slopes:

Type: Butterworth, Slopes: 6dB, 12dB, 18dB or 24dB
Type: Bessel. Slopes: 6dB, 12dB, 18dB or 24dB
Type: Linkwitz-Riley. Slope: 12dB or 24dB

```
Out12   LowPass
Freq:   1500Hz
```

```
Out12   LowPass
Type:   BUT 24dB
```

• **High Pass** – First adjust the High Pass Filter cut-off frequency. To switch it off, turn the rotary knob counter-clockwise until frequency reaches 20 Hz.

Second, press the rotary knob and select from the available types and slopes:
Type: Butterworth, Slopes: 6dB, 12dB, 18dB or 24dB
Type: Bessel. Slopes: 6dB, 12dB, 18dB or 24dB
Type: Linkwitz-Riley. Slope: 12dB or 24dB

```
Out12   HighPass
Freq:   40Hz
```

```
Out12   HighPass
Type:   BUT 24dB
```

5.2.5   Parametric EQ

Select between one of the 10 available Equalizers on each output channel by using the keys < *Select* and *Select* >. Browse the parameters by turning the rotary knob and push the knob to select and confirm the values. The following parameters can be adjusted for each EQ:

• **Freq** - EQ center frequency.
• **Gain** - EQ level gain.

```
Out12   PEQ 1
Freq:   1000Hz
```

```
Out12   PEQ 1
Gain:   0.00dB
```

• **Q** - EQ Bandwidth. For shelving filters the Q sets the transition in dB/Oct. See Annex[1] for further information about the Q factor implementation in the **LMS**608.
• **Type** - Shape of EQ. The available types are:
Bell: Modifies the gain of a certain frequency range, with bell shape
Notch: Eliminates a range around a center frequency
Low-Shelf: Modifies the gain of all the range below a selected frequency.
High-Shelf: Modifies the gain of all the range above a selected frequency.
All Pass: Modifies the phase response, without influencing the frequency response.
Band Pass: Filters out all the range except the defined band.
High Pass: Filters out all the range below a certain frequency.
Low Pass: Filters out all the range above a certain frequency.

```
Out12 PEQ 1
Q:     2.36
```

```
Out12 PEQ 1
Type: Bell
```

• **Enabled** – When set On, the currently selected EQ is on line.

```
Out12 PEQ 1
Enabled: On
```

5.2.6 Dynamics Compressor

A true RMS compressor can be set to avoid the output signal go above a certain RMS value. The following parameters can be adjusted:

• **Thr.** - Compressor Threshold. Sets the level at which the compressor will activate.
• **Att.** - Attack time. Time it takes the compressor to start actuating after reaching the threshold.

```
Out12   Compr.
Thr.:   24.00dBu
```
```
Out12   Compr.
Att.:   20ms
```

• **Hold** - Hold time. Sets up a delay before the compressor enters the release cycle. Useful for compressing low frequency long notes.
• **Rel.** - Release time. Time the compressor uses to return to unity gain after the signal is below the threshold.

```
Out12   Compr.
Hold:   10ms
```
```
Out12   Compr.
Rel.:   200ms
```

• **Ratio** - The compressor ratio determines the slope in which the signal is compressed. The higher this value is, the higher the compression.
• **Gain** – The compressor make up Gain. Use it in case the level of the compressed signal must be corrected (0dB by default)

```
Out12   Compr.
Ratio: 5.0:1
```
```
Out12   Compr.
Gain:   3.0dB
```

5.2.7 Limiter

A peak-limiter can be set at each input. It is a zero attack time limiter, so it will immediately act on the signal. The parameters that can be changed are:

• **Thr.** – Threshold: input level at which the signal will be limited.
• **Rel.** – The release value, expressed in dB/seconds

```
Out12   Limiter
Thr:    12.00dBu
```
```
Out12   Limiter
Rel:    50dB
```

For Amate Audio passive cabinets, there is a tool that easily computes the correct value of the limiter threshold for each output. These values are the result of a deep study made in out R+D facilities, thus its use guarantee protection without compromising the performance of the system. This tool is available at http://www.amateaudio.com/en/LimCal.

5.2.8   Phase – Phase inversion

Change the polarity of the channel by a 180º phase inversion. Select between Normal or Inverted.

```
Out12   Phase:
Normal
```

5.2.9   Channel Link

Output channels can be linked in order to set the same parameters on both. The factory setting is that output channels 1&2, 3&4, 5&6 and 7&8 are linked. Turn the rotary knob set the link off.

```
Out12   Link:
On
```

5.3 System Menu

The System Menu allows the user to control and change parameters that are related to the system behaviour and general operation. It can be accessed by pressing the *Menu* key on the front panel. The available options are:

5.3.1   Load - Program Recall

The DSP unit has a built-in non-volatile memory that can store different program setups. A program can be recalled using this menu. Use the rotary knob to browse the desired program to load and click it to accept

```
Load:          1
My_Preset
```

Confirm your selection by turning the rotary knob until "YES" is in capital letters and press enter again.

```
Load Preset ?          Load Preset ?
    NO/yes                 no/YES
```

5.3.2 Save - Program Store

A program can be stored using this menu. The old program with the same program number will be replaced. Once the program is stored in the flash memory, it can be recalled at a later time, even after power down.

Select the number of slot where the current setting will be stored by rotating the thumbwheel and pressing it to confirm. After that, it is required to enter a name for the new preset, using the rotary wheel to select the characters and the center click to confirm:

```
Save:           4

Empty
```

```
Set Presetname:

My_Pre_
```

Once you finish typing the name of the preset, press again the *Menu* key to confirm. Press the rotary knob to select "YES". A confirmation message is displayed.

```
Save Preset?

       no/YES
```

```
Preset Saved OK

My_Preset
```

5.3.3 Access Level – Lock the front panel through a password

```
Access Level:

     Unlocked
```

To lock the system, turn the rotary knob until the word "Locked" is displayed. The system will prompt for a password.

```
Access Level:

**** Locked ****
```

```
Set Password:

     MyPass
```

```
Confirm Password

     MyPass
```

The password may be up to 8 characters long. When entering shorter passwords, use the *Menu* key to finish and confirm. Use the *Exit* key to go back and make the password shorter.

WARNING: If blank spaces are included in the password, they will be stored and must be entered in the same position in order to unlock the device.
NOTE: The factory default password is "Password"

After that the system will be LOCKED and only the Mute buttons and the Menu key will be active.

To UNLOCK the device, proceed with the following steps:
Press the menu key. The Lock screen will appear. Turn the rotary knob counterclokwise to select "Unlocked"

```
┌─────────────────────┐        ┌─────────────────────┐
│ Access Level:       │        │ Access Level:       │
│                     │        │     Unlocked        │
└─────────────────────┘        └─────────────────────┘
```

Then enter the password. Click enter to finish.

```
┌─────────────────────┐
│ Enter Password      │
│     Pas_            │
└─────────────────────┘
```

5.3.4 Version Info

Shows the device software and hardware information. Turn the rotary knob to display the information available:

```
┌─────────────────────┐        ┌─────────────────────┐
│ Version Info        │        │ Version Info        │
│ SN: 0000010262      │        │ SW:10.0.3.105796    │
└─────────────────────┘        └─────────────────────┘
```

```
┌─────────────────────┐
│ Version Info        │
│ HW: 4.9.3           │
└─────────────────────┘
┌─────────────────────┐        ┌─────────────────────┐
│ IP Address          │        │ MAC Address         │
│ 169.254.50.254      │        │ B896:74:01:01:C1    │
└─────────────────────┘        └─────────────────────┘
```

5.3.5 Routing

The routing function assigns the inputs to the outputs using pre-established templates. The templates are shown in the table below:

Routing		Out 1	Out 2	Out 3	Out 4	Out 5	Out 6	Out 7	Out 8
Stereo	1-Way	In1	In2	Off	Off	Off	Off	Off	Off
Stereo	2-Way	In1	In2	In1	In2	Off	Off	Off	Off
Stereo	3-Way	In1	In2	In1	In2	In1	In2	Off	Off
Stereo	4-Way	In1	In2	In1	In2	In1	In2	In1	In1
Mono	8-Way	In1	In1	In1	In1	In1	In1	In1	In1

The unit assigns the Input source for the corresponding outputs when each routing mode is selected. The Stereo configuration will also activate the links between paired channels (Out 1&2, Out 3&4, Out 5&6, Out 7&8)

```
Routing:
Stereo 3-Way
```

NOTE: The configuration mode configures the input sources when selected. The user can change the source afterwards if desired. It does not keep the configuration in memory.

5.4 Copy/Paste Keys

The Copy and Paste keys can be used to copy the parameters of one configuration item (i.e. a parametric EQ, a compressor, a delay, etc.) from one channel to another.

The process is shown with the following example: The PEQ2 of Out3 will be copied to PEQ1 of Channel 5.

First, browse the menu until any of the parameters of the item PEQ2 of Out3 is displayed:

```
Out3    PEQ 2
Freq:   350Hz
```

At this point press the "Copy" key on the front panel. Note that all parameters in PEQ2 (Frequency, Type, Gain and Q) will be copied. The message "Copied" will be displayed.

Now browse until the LCD displays the destination parameter and press the "paste" key. A confirmation message is shown:

```
Paste Item?           Out5    PEQ 1
Push to Confirm       Freq:   350Hz
```

Push the rotary knob and all the parameters are copied to the new item.

NOTE: If the copied item is pasted to linked channels, the parameter will be copied to both channels.

6 PRESETS

The **LMS**608 is shipped with several presets pre-loaded in the unit that help in the set-up of the sound system. These presets have been designed and selected for deliver an active loudspeaker response (with internal amplification) when using them with our passive cabinets, as well for positive interfering in low frequency range and for protecting the sound system in over-voltage case.

In the following chart, the list of the presets is displayed:

Nr	Type	Name	OUT1 - OUT4	OUT 5 & 6
1	Read-Only	N6P	N6P / FLAT	N6P / FLAT
2	Read-Only	N6P & N12WP	N6P / XOVER	N12WP / LPF90+3
3	Read-Only	N26P	N26P / FLAT	N26P / FLAT
4	Read-Only	N26P & N12WP	N26P / XOVER	N12WP / LPF90+3
5	Read-Only	N12P	N12P/ FLAT	N12P/ FLAT
6	Read-Only	N12P & N18WP	N12P / XOVER	N18WP / LPF90+3
7	Read-Only	N12PR	N12PR / FLAT	N12PR / FLAT
8	Read-Only	N12PR & N18WPR	N12PR / XOVER	N18WPR/LPF90+3
9	Read-Only	N15P	N15P / FLAT	N15P / FLAT
10	Read-Only	N15P & N18WP	N15P / XOVER	N18WP / LPF90+3
11	Read-Only	N15P & N218WP	N15P / XOVER	N218WP/LPF90+3
12	Read-Only	N15PR	N15PR / FLAT	N15PR / FLAT
13	Read-Only	N15PR & N18WPR	N15PR / XOVER	N18WPR/LPF90+3
14	Read-Only	KEY10	KEY10 / FLAT	KEY10 / FLAT
15	Read-Only	KEY10 & N12WP	KEY10 / XOVER	N12WP / LPF90+3
16	Read-Only	KEY10 & N18WP	KEY10 / XOVER	N18WP / LPF90+3
17	Read-Only	KEY12	KEY12 / FLAT	KEY12 / FLAT
18	Read-Only	KEY12 & N18WP	KEY12 / XOVER	N18WP/ LPF90+3

NOTE: Outs 7&8 have in these presets a default (flat response) configuration.

In the case of set-ups without subwoofer units, the presets configure the passive loudspeakers as full-range loudspeakers so they will reproduce all the frequency range.

For the case of set-ups with subwoofers, the output channels from 1 to 4 (Corresponding to the "tops") are configured with a high pass filter, and the channels 5 to 6 (Corresponding to the "subs") are configured with a low pass filter and a boost of 3 dB in the tuning frequency. In this way we obtain a set-up with a balanced frequency response.

These presets are read-only, in consequence they can't be overwritten, edited nor deleted. If you want to edit a preset, before is necessary to copy it to another free memory available.

The presets don't include the proper limiter threshold, since it depends in the number of cabinets and amplifier type that are part of the sound system. For computing the proper limiter threshold, it is highly recommended to use the tool we have available in the following link: amateaudio.com/LimCal

7 OPERATING THE SOFTWARE

7.1 Device List

Once the software DSPLink is started, a window with the list of connected devices is shown. The main controls are as following:
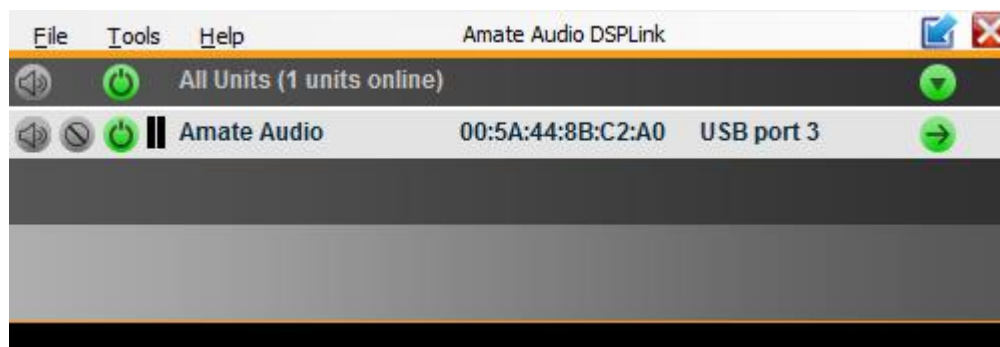
MUTE: The device will be completely muted when this button is clicked. Click it again to unmute.

IDENTIFY: When pressing this button, the device will blink 5 times its leds on the front panel. Useful for installations with several DSP connected to the computer, in order to identify each unit.

STANDBY: When this button is clicked, the DSP will go into standby mode (low power consumption). In this mode, there is no output signal. Click the button again to recover the normal operation mode.

EDIT: Open control window for full control of the device.

The following options are available in this window:
- **File / Quit**: Close the program.
- **Tools**
  - **Change software password**: allows the user to create a password to access the software. By default, no passwrod is needed. If a password is set and then forgotten, the software must be reinstalled.
  - **New Group**: create a group that associates 2 or more DSP.
  - **Enter Demo Mode**: enables to use the software without a connected device.
  - **Enable Update**: Enables the firmware upgrade of the units in the list (an Administrator password is required, see Section 8).
- **Help**
  - **Request Support**: sends a report about a problem in the software,
  - **About**: Shows basic info about the application

7.2 Device Options

7.2.1 Main window

In this window the Preset options, the device name, and the input and output levels and VUMeters, as well as the link and mute settings are displayed.

Here, also the source for In 3 & 4 can be selected. The available options are Analog or AES.
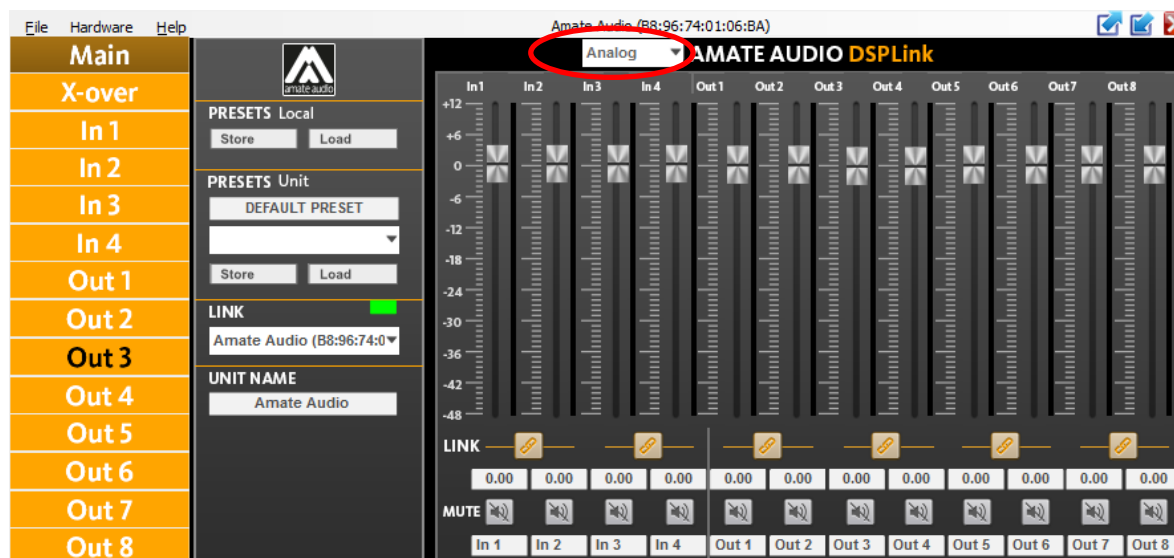

7.2.2 X-over window

In this window all the crossover settings for each output are displayed and can be set. The same information can be found on each channel's window.

7.2.3 Input / Output window

When selecting one of the inputs or outputs on the left side, a window with all available controls will be displayed: Mixer, PEQ, Xover, Gain, Mute, Polarity, Limiter and Compressor.

To access all available parameters of the Compressor, click on the word "COMPRESSOR" besides the Limiter control. A new window will display showing the controls for Threshold, Attack, Hold, Release, Ratio and Makeup Gain.
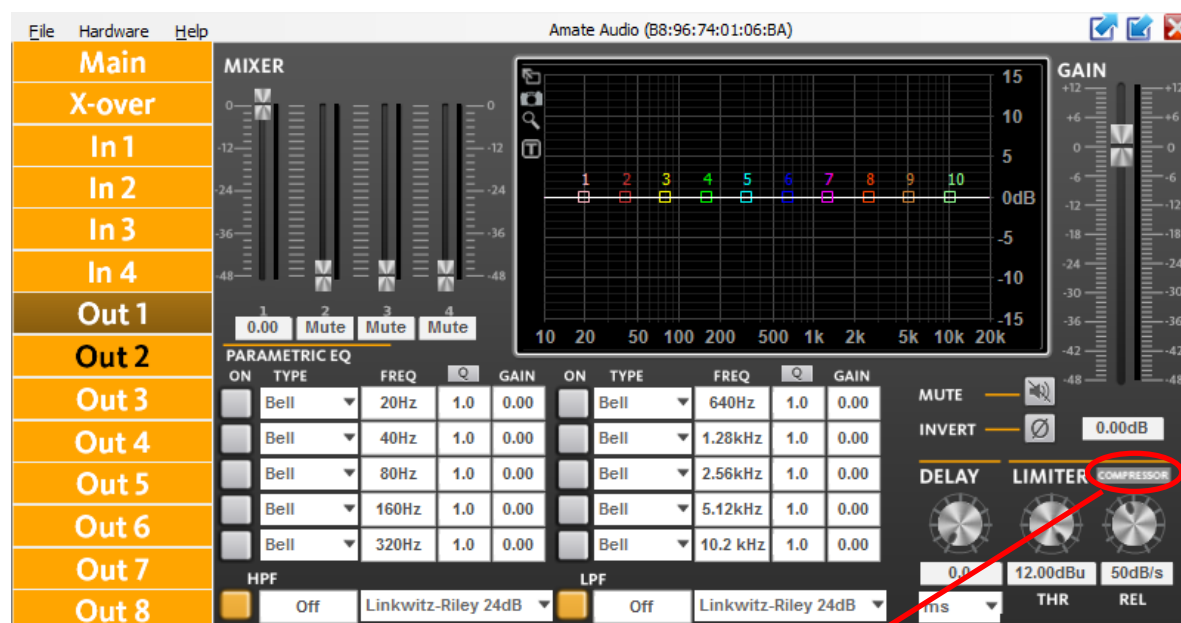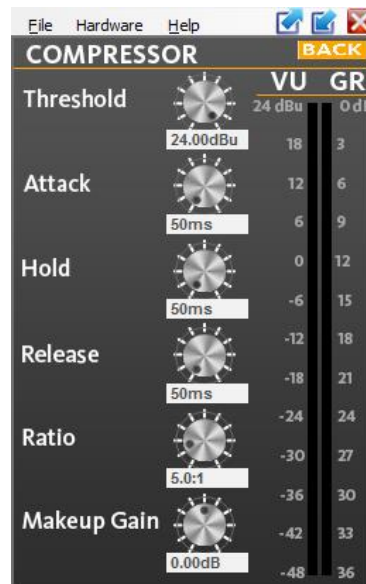
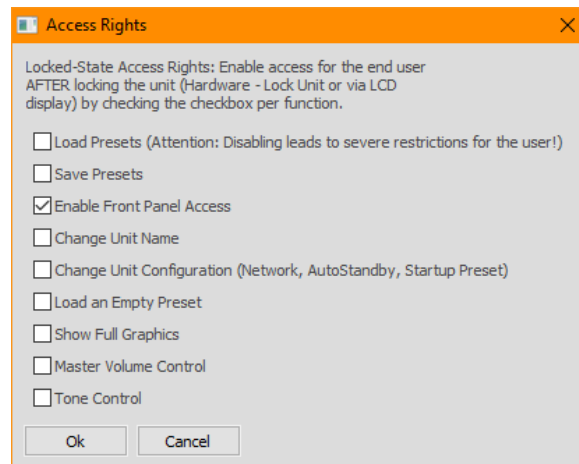### 7.2.4 File Menu

- **Open**: Load a preset from a previously stored file.
- **Save**: Save current preset to a file in the hard disk.
- **Restore Presets**: Load a set of presets from a previously stored file.
- **Backup Presets**: Backup all the presets of the device to the hard disk.
- **Quit**: Exit the device's main window.

7.2.5 Hardware Menu

- **Enter Password**: Enter a password here to unlock the unit (only needed if the unit has been previously locked using the option "Hardware>Lock Unit").

- **Configure**:
  - **Change password**: change the factory default password. The new password may have up to 8 characters. You will be first asked for the old password.
    NOTE: The factory default password is "Password" (case sensitive)
  - **Network Settings**: allows to manually configure the IP of the device (static IP).
  - **Power On Preset**: Set here the preset that will be selected at startup. The default setting is "last setting".
  - **Global Access Rights**: select which controls will be unlocked even the "Lock Unit" control is selected. Click on the checkbox of the functions that should be available to the user without entering a password:

Access Rights

Locked-State Access Rights: Enable access for the end user
AFTER locking the unit (Hardware - Lock Unit or via LCD
display) by checking the checkbox per function.

☐ Load Presets (Attention: Disabling leads to severe restrictions for the user!)
☐ Save Presets
☑ Enable Front Panel Access
☐ Change Unit Name
☐ Change Unit Configuration (Network, AutoStandby, Startup Preset)
☐ Load an Empty Preset
☐ Show Full Graphics
☐ Master Volume Control
☐ Tone Control

Ok   Cancel

- o **Output mode**: pre-select the input assignment to outputs, in groups of two (Dual Bridge or Mono).
- o **Routing**: Changes the input assignments and links of outputs as described in Section 5.3.5.
- o **AES -> Analog Failover**: when this feature is On, and the unit is operating with AES input, the processor will automatically switch to the analog input if the AES signal is lost. A valid analog input must be connected to the desired inputs.

- • **Lock unit**: In order to lock the unit by password, select "Hardware > Lock Unit". To unlock the unit again, select "Hardware > Enter Password".

- • **Go to Standby:** the DSP will go into standby mode (low power consumption). In this mode, there is no output signal. Is is equivalent to press on the Standby button at the Device List.

- • **Set Pin**: Set a 4-digit pin to allow the access to the unit via software. If the PIN needs to be removed, select this function and leave the PIN field blank (the message "Invalid PIN" will be shown). If the PIN is forgotten, a firmware upgrade is needed (see Chapter 8).
- • **Status Details**: A quick information window about the device is displayed.

8 QUICK REFERENCE

Menu Function	Channel	Parameter	Value	Units
Gain	I / O	Gain	-48 to +12; Step: 0.25	dB
Source	Inputs	Source	Analog / AES Stereo	
Mixer	Outputs	Input 1/2/3/4	-48 to 0; Step: 0.25	dB
Delay	I / O	Delay	I: 0 to 2000; Step depends on range O: 0 to 50	ms
LowPass / High Pass	I / O	Freq.	20 to 20000; Step depends on range	Hz
		Type	BUT 6dB / BES 6dB / BUT12 dB / BES 12 dB / LR 12 dB / BUT 18 dB / BES 18dB / BUT 24dB / BES24 dB / LR 24dB / LR 48dB	
		Enabled	On / Off	
PEQ 1 to PEQ 10	I / O	Freq.	20 to 20000; Step depends on range	Hz
		Gain	-12 to 12; Step: 0.25	dB
		Q	0.2 to 25; Step: 0.1	
		Type	Bell / Notch / High Shelf / Low Shelf / Allpass/ Band Pass/ High Pass/ Low Pass	
		Enabled	On / Off	
Compressor	I / O	Thr.	-48 to +24; Step: 0.25	dBu
		Att.	1 to 10000; Step: 1	ms
		Hold	1 to 10000; Step: 1	ms
		Release	1 to 10000; Step: 1	ms
		Ratio	1.2:1 to 25:1	
		Makeup Gain	-12 to +12; Step: 0.25	dB
Limiter	Input	Thr.	-48 to +24; Step: 0.25	dBu
		Rel.	10 to 100; Step: 1	dB/s
Limiter	Output	Thr.	-48 to +12; Step: 0.25	dBu
		Rel.	10 to 100; Step: 1	dB/s
Phase	Output	Phase	Normal / Inverted	
Link	I / O	Link	Off / On	

9 TROUBLESHOOTING

9.1 How to perform a firmware upgrade

To perform a firmware upgrade, proceed as following:

WARNING: In the Firmware Upgrade Process all your presets will be erased. Please make a backup of them before upgrading the unit. During the upgrade process you will be asked to perform this backup.

1. Connect the unit to the computer
2. Using DSPLink, enter the Device Main Window
3. Choose "Hardware > Enter Password"
4. Enter the administrator password, which is "Ad_min"
5. Go to "Hardware > Firmware Upgrade"
6. Be careful to answer "Yes" if you want your settings and presets to be backed up
7. Wait until firmware is upgraded. Do not plug off or disconnect the unit from the computer during this process.
8. Your presets will be restored automatically

9.2 Password recovery

The unit is shipped unlocked, so no password will be necessary unless the unit is locked by the user. The default user password is "Password".

In case this Password has been changed by the user (Using the Menu "Hardware > Configure > Change Password") and is no longer remembered, a Firmware Upgrade is needed to re-establish the password to the default value (see previous section).

10 SPECIFICATIONS

	LMS608
Analog Inputs	
Number	4
Input Impedance	21 kOhms
Maximum Level	+24 dBu
Type	Electronically balanced
Digital inputs	
No. of channels / connectors	2 / 1 XLR3
Type	AES/EBU
Accepted formats	16 or 24 bit; 44.1, 48, 88.2 or 96kHz
Analog Outputs	
Number	8
Maximum Level	+12 dBu
Type	Impedance Matched
Audio Performance	
Frequency Response	20Hz to 20kHz (+/- 0.5dB)
Dynamic Range	116 dB (A-weighted)
Crosstalk	< -85 dB
THD	< 0.005%
Digital Audio Performance	
Processing	64-bit
Analog Converters	High Performance 24-bit
Sampling Rate	96 kHz
Propagation Delay	350 us
Front Panel Controls	
Display	2 x 16 Character. White Backlit LCD
Level Meters	Per I/O: Signal Present, -6dB; Limiter, Clip
Buttons	Illuminated Mute Controls, Menu Controls
Dial Encoder	Rotary Thumb Wheel
Connectors	
Analog Audio	3-pin XLR Pin 1: shield 2: live (+) 3: return (-)
Digital Audio (AES/EBU)	3-pin XLR
USB	Type B (on front panel)
Ethernet	RJ45 (on rear panel)
Power	Standard IEC Socket
General	
Power	85 to 240 VAC (50 / 60 Hz)
Dimensions (H x W x D)	1U 19" Rack 44 x 483 x 165 mm
Weight	2.5kg

System Parameters	
No. of Programs	100
Program Names	16 character length
Security Lock	Password Lock/Unlock

NOTE: Specifications subject to change without notice

11 ANNEX

[1] For bell parametric filters (PEQ), the bandwidth definition differs among the different DSP manufacturers. For this reason, it is difficult to successfully copy filter settings between different processors.

The frequency span of the filter is defined by either the bandwidth or the Q factor. These parameters are inversely proportional, meaning the bigger the bandwidth, the smaller the Q. The formulas relating both parameters are:

$$Q = \frac{f_0}{f_H - f_L} = \frac{f_0}{B}$$

$$B = f_H - f_L$$

The border frequencies of the bandwidth are usually set at the point where the energy decreases three decibels, as shown in *Figure 1*.
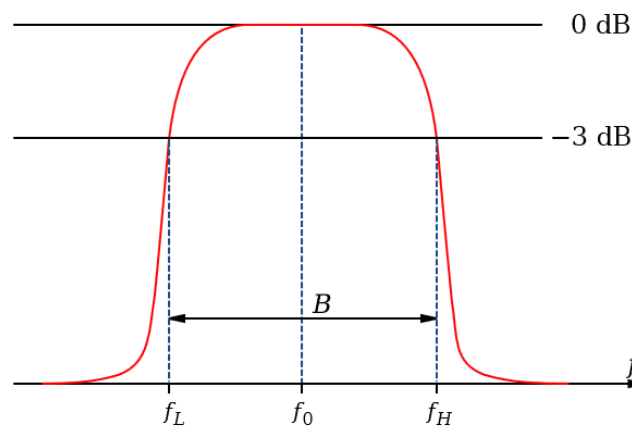

Fig. 1: Bandwidth graphical representation

The majority of audio processor manufacturers implement the Q in four different ways: Bandpass-Q, Constant-Q, dB/2 and 3dB hybrid method. The first two are based in analogue filters, while the last two take advantage of the capabilities of digital processing.

The Bandpass-Q method builds the bell filter adding the response of a band-pass filter plus a gain block to the input signal, as shown in Figure 2.
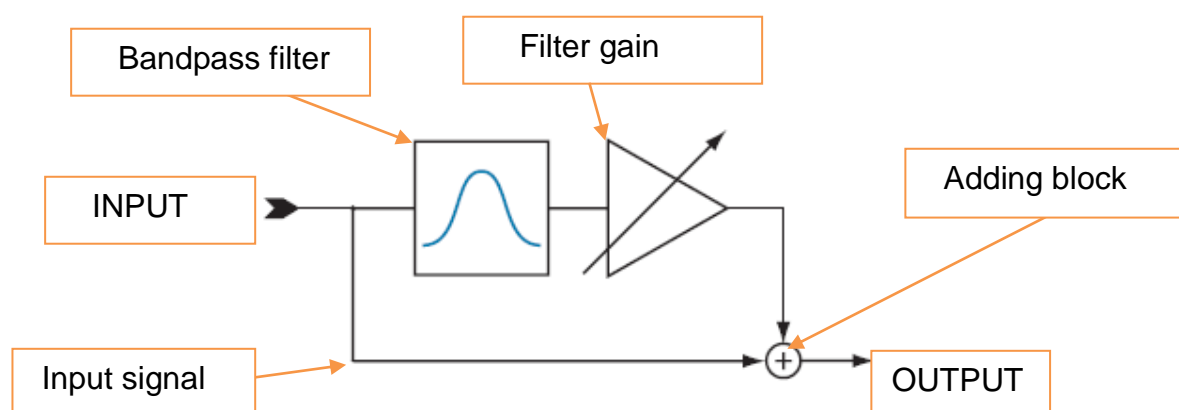
Fig. 2: Block diagram of a bell filter

As shown in Fig. 2, when adding both processed and original signals at the output, the bandwidth of the resulting signal is higher than the bandwidth of the bandpass filter used. The Bandpass-Q method defines the Q of the bandpass filter block, and not of the resulting filter. The difference is shown in Figure 3.
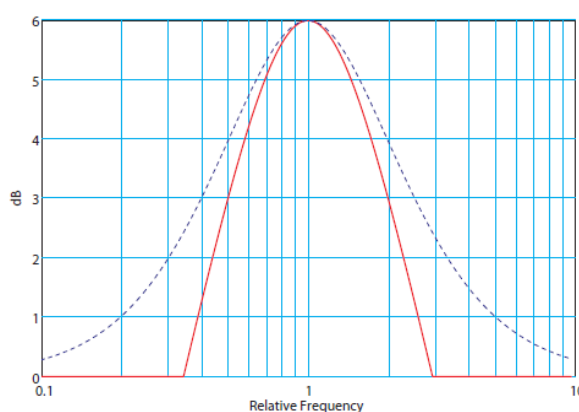

Fig. 3. Red line: Bandpass filter with 6dB gain. Dotted blue line: Parametric bell filter.

To avoid this difference, Constant-Q implementations lower the Q that the user selects in the bandpass filter, just enough to compensate this difference. In consequence, the resulting Q at the output is the same as the user expects. However, this solution cannot be applied for filters with a peak gain lower than 3dB.

The dB/2 method always sets the bandwidth as one-half of the peak gain, so the definition works no matter what the peak gain is. Amate Audio **LMS**608 uses the dB/2 method, because it keeps the bell shape independent from the peak gain.

The 3dB Hybrid Method works in two ways: If the peak gain is greater than 6dB, the bandwidth is set at -3dB of the peak (Constant-Q), otherwise it uses the dB/2 definition, setting the bandwidth to one-half of the peak gain.

CERTIFICATION

DECLARATION OF CONFORMITY

In accordance with EN 45014:1998

Manufacturer's Name: "AMATE AUDIO S.L."

Manufacturer's Address: C/ Perpinyà 25, Polígon Industrial Nord
08226 Terrassa, (Barcelona), SPAIN

Brand: "AMATE AUDIO"

We declare under our own responsibility that:

Product: Audio signal processor. Audio apparatus for professional use
Name: LMS608

Conforms to the following product specifications:

Safety: IEC 60065-01 + A1

EMC: EN 55022:2006
EN 55103-1:2009
EN 55103-2 2009
FCC Part 15

WARNING:

In accordance to EN55022, this is a class A product. In a domestic environment this product may cause radio interference in which case the user may be required to take adequate measures.

Supplementary Information:
The product herewith complies with the requirements of the:

Low Voltage Directive 2006/95/EC
EMC Directive 2004/108/EC
RoHS Directive 2002/95/EC
WEEE Directive 2002/96/EC

With regard to Directive 2005/32/EC and EC Regulation 1275/2008 of 17 December 2008, this product is designed, produced, and classified as Professional Audio Equipment and thus is exempt from this Directive.

Date of issue: January 28th, 2020

Signature:

AMATE AUDIO S.L.
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General Manager

$C \in$

Conformity
Marking

NOTES:

LMS loudspeaker management systems have been designed, engineered and manufactured in Barcelona – SPAIN by

Amate Audio S.L.

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