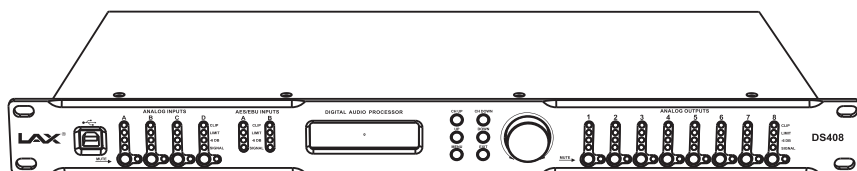# USER'S MANUAL

DS 408

PROFESSIONAL AUDIO PROCESSOR

USER'S MANUAL

PROFESSIONAL AUDIO PROCESSOR

Due to our continuous product improvement,
all specification and feature are subject to change without prior notice.

CONTENTS

Warranty Card

锐丰企业 RUIFENG CORPORATION

Customer Hot Line:
400-707-6800

LAX   C.V. products which produced by RUIFENG Corporation(RUIFENG in brief) are ensured with highquality by strict quality inspection and contr -ol before delivery.our will provide warranty according terms mentioned in this warranty card. The warranty is viod if warranty card or purchasing invoice is missed.The service is suggested to start by calling our service No.:400-707-680 in advance.

一、This warranty is valid within one year form date of sale (except sone occasions specially mentioned as below),and covers for manufacturing defects and all matters concerning the workmanship of LAX&C.V. products。

二、Warranty period for particular item。
     1、offering one month free change service and one year warranty for amplifiers,mixers or processors.
     2、Offering three months warranty for which categorized as spare parts:
     1.)The voice coil and diaphragm of microphone.
     2.)The voice coil and diaphragm of speaker.

C、RUIFENG willnot be responsible for any warranty if/for：
    1、Non-this company proxy product；2、Fake products；
    3、Products without intact number or serial number is changed,deleted or transferred or has not been recorded in the company s record。

D、Thewarranty becomes void if product problems result from；
     1、improper operation and misuse(overloading or bumed with feedback)；
     2、Careless moving.
     3、Terrible conditions(over-higt input voltage and humidity)
     4、Irresisitible situation such as STORMor EARTH QUAKE…etc；
     5、repaired or restructured by unofficial technicians outside RUIFENG ；
     6、Failure assemble/installation from the instructions of manufacturer。

E、The warranty applies only on the cost of repair or replacement and does not include the relevant freight charges。

F、RUIFENG is not responsible for the loss caused by equipment or component damaging。

G、RUIFENG charges reasonably if warranty period expired。

H、The warranty card is subject to change without prior notice。

Purchaser Name:	Manufacturer:

Dealer Name:	Dealer Address:		
	City	State	ZIP

Purchase Date:	Model:	Serial No.:	Invoice No.:
(d) (m) (y)			

Add:114 Yushan Rd,panyu,guangzhou,guangdong Province,china   Tel:(86 20)84817168

The information furnished in this manual does not include all of the details of design and engineering of this particular product; not does it cover every possible application or situation concerning its usage, which may occur during the installation, operation or maintenance of said LAX product.

IMPORTANT
THE PRODUCT REQUIRES
CLASS 2 OUTPUT WIRING.

⚠ CAUTION ⚡

TO PREVENT ELECTRIC SHOCK, DO NOT REMOVE TOP OR BOTTOM COVERS. NO USER SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL. DISCONNECT POWER CORD BEFORE REMOVING REAR PANEL COVER TO ACCESS GAIN SWITCH.	Shock Hazard - Do Not Enter Choc Hasard - N*entrent Schocke Hazard - Test Nicht Betrete Urto Hazard - Do Non Entrano

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WARNING
TO REDUCE THE RISK OF ELECTRIC SHOCK, DO NOT EXPOSE THIS EQUIPMENT TO RAIN OR MOISTURE!

Magnetic Field
CAUTION: Do not locate sensitive high-gain equipment such as preamplifiers or tape decks directly above or below this unit. Because this amplifier has a high power density, it has a strong magnetic field which can induce hum into unshielded devices that are located nearby. This field is strongest just above and below the unit. If an equipment rack is used, we recommend locating the amplifier(s) at the bottom of the rack and the preamplifier or other sensitive equipment at the top.

The lightning bolt triangle is used to alert the user to the risk of electronic shock	The exclamation point triangle is used to alert the user to important operating and/or maintenance instructions.	Printed on recycled paper.

Foreword:

Thank you for purchasing the DS408 processor. LAX has always strived to become a leading brand in the field of high-end digital audio processing.After you are familiar with the features and operation of the DS408 processor, we recommend that you find the most efficient operation for your system by using the powerful processing capabilities of the Ds408.

This manual describes how to set up the DS408 processor on the front panel, or it can be operated by connecting a PC, MAC, iPad or iPhone. For details, please refer to the Software User Manual in the remote control application.

Controlled via the front panel or remote control app, DSP settings and coefficients are calculated in real-time as parameters are changed. The frequency setting can reach 1Hz precision, the delay can reach 15 microsecond precision, and the gain can reach 0.01dB precision. When editing values via the front panel, the precision interval for some values is limited; eg, in this case, the gain is set in 0.25dB steps.

Features:

- 2 DANTE inputs
- 118dB dynamic range (inputs) / 114dB dynamic range (outputs)
- Electronically balanced inputs
- Matched-impedance outputs
- Frequency range 20Hz....26kHz
- 10 parametric filters per input and output
- EQs can be set as Bell, High Shelf, Low Shelf, Notch, Allpass, Band Pass, High Pass, Low Pass
- 2000ms delay per input, 2000ms delay per output
- Butterworth, Bessel, Linkwitz-Riley filters up to 24dB / oct.
- True RMS Compressor
- Zero Attack Peak Limiter
- Latency 990 microseconds
- Full configuration and real-time monitoring via PC, MAC, iPad or iPhone
- 100 Presets
- 64 bits multi-mode digital processing
- Loudspeaker Library support

Safety Instructions:

WARNING, FOR YOUR PROTECTION READ THE FOLLOWING: The apparatus shall not be exposed to dripping or splashing liquid and no object filled with liquid, such as vases, shall be placed on the apparatus. CLEAN ONLY WITH A DRY CLOTH. DO NOT BLOCK ANY VENTILATION OPENINGS. INSTALL IN ACCORDANCE WITH THE MANUFACTURERS INSTRUCTIONS. DO NOT INSTALL NEAR ANY HEAT SOURCES SUCH AS RADIATORS, HEAT REGISTERS, STOVES, OR OTHER APPARATUS (INCLUDING AMPLIFIERS) THAT PRODUCE HEAT. ONLY USE ATTACHMENTS/ACCESSORIES SPECIFIED BY THE MANUFACTURER. UNPLUG THIS APPARATUS DURING LIGHTNING STORMS OR WHEN UNUSED FOR LONG PERIODS OF TIME. Do not defeat the safety purpose of the polarized or grounding-type plug. A polarized plug has two blades with one wider than the other. A grounding type plug has two blades and a third grounding prong. The wide blade or third prong is provided for your safety. If the provided plug does not fit your outlet, consult an electrician for replacement of the obsolete outlet. Protect the power cord from being walked on or pinched particularly at plugs, convenience receptacles, and the point where they exit from the apparatus. Use only with the cart stand, tripod bracket, or table specified by the manufacture, or sold with the apparatus. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over. Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as power-supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped. POWER ON/OFF SWITCH: If the equipment has a Power switch, the Power switch used in this piece of equipment DOES NOT break the connection from the mains. MAINS DISCONNECT: The plug shall remain readily operable. For rack mount or installation where plug is not accessible, an all-pole mains switch with a contact separation of at least 3 mm in each pole shall be incorporated into the electrical installation of the rack or building. FOR PRODUCTS EQUIPPED WITH EXTERNALLY ACCESSIBLE FUSE RECEPTACLE: Replace fuse with same type and rating only. MULTIPLE-INPUT VOLTAGE: This equipment may require the use of a different line cord, attachment plug, or both, depending on the available power source at installation. Connect this equipment only to the power source indicated on the equipment rear panel. To reduce the risk of fire or electric shock, refer servicing to qualified service personnel or equivalent. If connected to 240V supply, a suitable CSA/UL certified power cord shall be used for this supply. This equipment is intended for rack mount use only.

WARNING:

THIS APPLIANCE SHALL BE CONNECTED TO A MAINS SOCKET OUTLET WITH A PROTECTIVE EARTHING CONNECTION. The cores in the mains lead are colored in accordance with the following code: GREEN and YELLOW - Earth; BLUE - Neutral; BROWN - Live. As colors of the cores in the mains lead of this appliance may not correspond with the colored markings identifying the terminals in your plug, proceed as follows: The core which is colored green and yellow must be connected to the terminal in the plug marked with the letter E, or with the earth symbol, or colored green, or green and yellow. The core which is colored blue must be connected to the terminal marked N or colored black. The core which is colored brown must be connected to the terminal marked L or colored red. This equipment may require the use of a different line cord, attachment plug, or both, depending on the available power source at installation. If the attachment plug needs to be changed, refer servicing to qualified service personnel who should refer to the table below. The green/yellow wire shall be connected directly to the product's chassis.

1. Overview of Connectors and Controls

In addition to the available features described in this manual, you also have the option of controlling the processor remotely over your network. Please refer to the Software Manual for details.

1.1. Front Panel
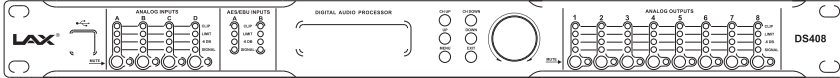
1.1.1. USB Connector

The USB connector is used to connect the ASP-4M8 to a PC or MAC. It can then be configured and controlled in real time with the freely available PC and MAC configuration program. Please check our website regularly for updates.

1.1.2. Input VU Meters

There are 6 independent input meters available. The LEDs indicate the signal level at the inputs, in dBu, after the input gain faders. The Peak LED lights when the input level exceeds +20dBu.

1.1.3. Function Buttons

The function buttons allow direct access to all editing and navigating functions. See section three for a detailed description on the functions of each of these buttons.

1.1.4. Rotary Encoder

The rotary encoder is used to scroll through the menu items by pushing it, and to adjust values by rotating it.When values are changed, the DSP is updated in real-time.

1.1.5. Output VU Meters

There are eight independent output meters available. The indication refers to the output level w.r.t. the threshold of the output peak limiter. The Limit LED lights when any gain reduction is taking place in that channel. The Peak LED lights when more than 12dB of gain reduction is taking place.

1.1.6. Output Mute Buttons

The eight output mute buttons are used to independently mute each output.

1.2. Rear Panel

1.2.1. Power Connector

The ASP-4M8 has an internal power supply that will accept voltages ranging from 100V - 240V at frequencies from 50Hz-60Hz. A power cord is included.

1.2.2. Power Switch

The Power Switch turns the ASP-4M8 on and off. As with all professional audio installations, power amplifiershould be turned on last when powering up and must be turned off first when powering down.

1.2.3. Ethernet Connector

This RJ-45 connector is used to connect your product to a network. It can then be configured iand controlled in real time with our PC and MAC configuration program, or with our iPad / iPhone control software (available on the Apple App Store). From the PC or MAC you can also update your processor to the latest firmware.Please check our website regularly for updates.

1.2.4. Analog inputs

The analog input section of the ASP-4M8 offers four electronically balanced XLR connectors. The input level is +23dBu max, electronically balanced.

1.2.5. AES3 Inputs

The digital input of the ASP-4M8 offers two input channels on one electronically balanced XLR connector.The input will accept AES3 input signals at 44.1, 48, 88.2, or 96kHz. An internal asynchronous sample rate converter (ASRC) converts the sample rate to the ASP-4M8's native sample rate, thus preserving the highest possible processing quality even when running off a 44.1kHz digital signal.

1.2.6. DANTE Inputs

The ASP-4M8 offers two Dante input channels. No separate connector is needed; the ASP-4M8 uses one Ethernet connection for Dante signals and Command & Control signals. The processor will accept sample rates of 44.1, 48, 88.2, and 96kHz. The Dante inputs can be configured though Audinate's Dante Controller Software. An internal asynchronous sample rate converter (ASRC) converts the sample rate to the ASP-4M8's native sample rate, thus preserving the highestpossible processing quality even when running off a 44.1kHz digital signal.

1.2.7. Analog Outputs

The output section of the ASP-4M8 offers eight load balanced XLR connectors. The output level is +12dBu max, impedance matched.

2. Getting Started

2.1. Quick Start

For those of you that wish to jump right in, the following information has been provided to act as a quick start guide for optimizing performance of your processor.

2.2. Connections

When setting up your processor, make connections as follows:

2.2.1. Signal Connections
- Always make connections prior to applying power to the processor.
- Connect the output(s) from the sending device (mixer) to the Ch. 1 XLR input connector (mono) or both, Ch. 1 and Ch. 2 XLR input connectors (stereo).
- Make output connections from the output XLR connector(s) to the input connector(s) of the selected power amplifier(s).

Always make sure that your power amps are the last item turned on and the first turned off. Once all of the connections have been made, and the processor is powered up, you can navigate through the entire signal path of your processor from the front panel. The display provides you with a clear and concise overview of each aspect of the signal path from the input (left side of display) to the output section (right side of display).

2.2.2. Remote Monitoring and Control
For the quickest and easiest results we recommend to install and use our freely available iPad, IPhone, PC or MAC configuration software. It provides a complete and comprehensive tool When connecting the ASP-4M8 via an Ethernet network, you have several options.The PC/MAC application has an auto-discovery tool that will automatically connect to any unit found on the network. Please refer to the Software Manual available on our website for details on the control software.
Note: Please make sure there is only one device (Windows PC, MAC PC or iOS device) running the LAX DSP Control software at any time. The software can control as many devices simultaneously as the network can handle. In a typical (default) DHCP configuration,this is 200 units.

2.2.2.1. Standard DHCP Network
This is the recommended connection method. For plug-and-play connection, use a standard Ethernet Router,plug in the unit and your computer, and the connection should be made automatically. Standard routers have a DHCP server built in and enabled. A DHCP server assigns a network address to your computer and to the ASP-4M8, allowing them to connect automatically.

2.2.2.2. Connection without DHCP server

Alternatively, you can use a direct (cross) cable or a simple Ethernet Switch to connect the unit to your PC or MAC. Please note that in this case, especially Windows computers running Windows XP may take up to several minutes to assign themselves a network address (indicated by the text: "This connection has limited or no connectivity"). The computer and the ASP-4M8 both will assign themselves a network address in the ZeroConfig range(169.254.0.0 - 169.254.255.255), and the ASP-4M8 will be discovered automatically by the PC/MAC application. To facilitate connections without DHCP router, it may be convenient to set your computer to a fixed IP address. If you do this, please choose any address in the ZeroConfig range, and setthe subnet mask to 255.255.0.0.
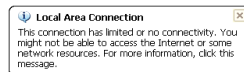
2.2.2.3. Fixed IP address

It is also possible to set a fixed IP address and subnet mask for the ASP-4M8. WARNING: If you set a wrong address (in a different range than your computer), you may not be able to connect anymore. It is strongly recommended to consult a system administrator before making such settings. It is not possible to set the IP address via the front panel menu. If you have set a fixed IP address and subnet mask, you must set your computer to an address within the same subnet. To verify this, you can read out the IP address via the local LCD display: Press "Menu" 4 times, then press the rotary encoder until the IP address and subnet mask appear.

2.2.2.4. Firewall

After starting the software application for the first time, your computer may ask you to allow or block the application access to the network. Please make sure to allow this; if there is a firewall between the application and the ASP-4M8, the application will not find the ASP-4M8.

2.3. System Setup and Gain Structure

This product offers a wide range of tools for sound system design and setup. These tools can make your system more efficient and better sounding, but to get the best possible sound it is important to use these tools properly.The following section explains how to maximize system gain and how to use the limiters to protect your amplifiers from clipping. In traditional system design, the output of your console would be routed to a system EQ, a compressor, and a crossover with output level control. From the crossover, there may be additional filters that are employed to improve the response of your speakers. There may also be limiters set up to keep your amplifiers from going into clipping and protect your speakers from the hazards of a clipped signal. Your amplifiers play a vital role in system setup, because they are last item in the chain before your speakers and offer the greatest amount of gain (that is their job after all). If your limiters and amplifiers are incorrectly setup you will not be using your system to its fullest potential and could be harming your speakers.

To ensure an optimal gain structure:
Play a signal at the nominal level from your mixing desk, and set the input gain of your processor to 0.Set the crossovers that you want to use, while keeping the output gains also at 0.
With DISCONNECTED loudspeakers, turn up the volume of the power amplifiers entirely clockwise (fullvolume).

Now 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. If the amplifier does not have a built-in Peak limiter, set your processor's limiter so that the amplifier does not clip. Now turn down the volume of the power amplifiers, connect your speakers, and slowly increase the volume while checking the sound.If all is well, there should be 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.If you can not reach enough signal level, increase the processor's input gain or turn up the level from your mixing desk.

3. Setup And Operation

Before plugging the processor in, always make sure that the power supply matches the product specification components of the system are set up and connected properly. Make sure your power amplifiers are switched on last in order to avoid transients, which could damage your speakers or annoy your audience.

3.1. System Check

After connecting all cables, you should mute all outputs first or turn the gain/level setting on your amplifiers to minimum. Activate the HF-outputs first. In case of wrong cabling, High Frequency (HF) audio signals will come out of bass-speakers that cannot be harmed this way. Vice versa, the Low Frequency (LF) audio signals would destroy your HF-speakers. It is advisable to install a large capacitor in series with HF drivers (47 - 100 uF). That way, drivers are somewhat protected against accidental instrumentation errors.

3.2. Input Setup

Select the input channel you wish to edit by pressing the channel button below the VU meter for the corresponding channel once. Make the first settings with the output turned low or muted.To toggle through the different items, press the Item Up or Item Down buttons.

3.2.1. Gain

Adjust the gain by dialing the rotary encoder. The gain is adjusted in steps of 0.25dB. Smaller steps (0.01dB) can be set via the PC or MAC interface.

```
In AB Gain
     -2.25dB
```

3.2.2. Input Selection

Select the source (Analog, AES3 or Dante) by rotating the encoder.

```
In AB Input
Analog
```
```
In AB Input
AES/EBU
```

3.2.3. Delay

Set the delay time by turning the encoder. Select the displayed unit (ms or s, mm or m, feet, inches, or mils) by pushing the encoder.

```
In AB Delay
     1.020ms
```
```
In AB Delay
     100.29m
```
```
In AB Delay
     244.5Feet
```

3.2.4. Low Pass Filter

Adjust the Low Pass Filter frequency by turning the rotary encoder. You can switch the low pass filter off by turning the rotary encoder up (clockwise) until the frequency passes 20kHz.Press the rotary encoder to change the filter type. Select the type by turning the rotary encoder. You can choose from: Butterworth 6dB, Bessel 6dB, Butterworth 12dB, Bessel 12dB, Linkwitz Riley12dB, Butterworth 18dB, Bessel 18dB, Butterworth 24dB and Bessel 24dB. Higher order filters may be set by adding filter sections in the PEQ blocks (see below).

```
In AB Low Pass
Freq: 14500Hz
```
```
In AB Low Pass
Type: But24
```

3.2.5. High Pass Filter

Adjust the high pass frequency by dialing the rotary encoder. You can switch the high pass filter off by dialing the rotary encoder down (counterclockwise) until the frequency passes 20Hz. Press the rotary encoder to change the filter type. Select the type by turning the rotary encoder. You can choose from: Butterworth 6dB, Bessel 6dB, Butterworth 12dB, Bessel 12dB, Linkwitz Riley12dB, Butterworth 18dB, Bessel 18dB, Butterworth 24dB and Bessel 24dB. Higher order filters may be set by adding filter sections in the PEQ blocks (see below).

```
In AB High Pass
Freq: 34Hz
```
```
In AB High Pass
Type: Bes12
```

3.2.6. Parametric Equalizer (PEQ)

There are 10 bands of parametric equalization. Each band can be adjusted freely over the frequency range of 20Hz to 20kHz. Adjust the frequency by dialing the rotary encoder.Press the rotary encoder to select the parameters. The available parameters are: Frequency (20Hz to 20kHz), Gain (-12dB to +12dB), Q (0.2 to 25), Enabled (On or Off), Type (Bell, High Shelf, Low Shelf, Notch, All Pass, Band Pass, High Pass, Low Pass). For the Shelving filters, the Q value sets the steepness of the filter in dB/Oct.

```
In AB PEQ 5
Freq: 14500Hz
```
```
In AB PEQ 5
Gain: -7.75dB
```
```
In AB PEQ 5
Q:     1.5
```

```
In AB PEQ 5
Type: Bell
```
```
In AB PEQ 5
Enabled: On
```

3.2.7. Compressor

The compressor is a true RMS compressor. Turn the rotary encoder to set the threshold. Press the rotary encoder to select the parameters. The available parameters are: Threshold, Attack, Hold, Release, Ratio,and Makeup Gain.

```
In AB Compressor
Thr.: 23.00dBu
```
```
In AB Compressor
Att.: 50ms
```
```
In AB Compressor
Hold: 10ms
```

```
In AB Compressor
Rel.: 500ms
```
```
In AB Compressor
Ratio: 1:4.5
```
```
In AB Compressor
Gain: 4.75dB
```

3.2.8. Limiter

The limiter is a zero-attack peak limiter. Only the threshold and release can be set. Press the rotary encoder to select the parameters. The release value is displayed in dB per second.

```
In AB Limiter
Thr.: 23.00dBu
```
```
In AB Limiter
Rel.: 50dB/s
```

3.2.9. Channel Link

By linking 2 channels, the settings are guaranteed to be identical for both channels, except for mixer and mute. Turn the encoder to set the Channel Link on or off.

```
In AB Link        In A Link
On                Off
```

3.3. Output Setup

Select the output channel you wish to edit by pressing the channel button elow the VU meter for the corresponding channel once. Make the first settings with the output turned low or muted.To toggle through the different items, press the Item Up or Item Down buttons.

3.3.1. Gain

Adjust the gain by dialing the rotary encoder. The gain is adjusted in steps of 0.25dB. Smaller steps (0.01dB)can be set via the PC or MAC interface.

```
Out12 Gain
       -2.25dB
```

3.3.2. Mixer

Turn the rotary encoder to mix the signal from the selected input to the selected output. Push the rotary encoder to select the input. Attention: The mixer is only available while the outputs are not linked by Channel Link. If the channels are linked, the mixer will be skipped.

```
Out1  Mixer       Out1  Mixer
In A: -6dB         In B: Off
```

3.3.3. Delay

Set the delay time by turning the encoder. Select the displayed unit (ms or s, mm or m, feet, inches, or mils)by pushing the encoder.

```
Out12 Delay       Out12 Delay       Out12 Delay
     1.020ms           100.23m           244.5Feet
```

3.3.4. Low Pass Filter

Adjust the Low Pass Filter frequency by turning the rotary encoder. You can switch the low pass filter off by turning the rotary encoder up (clockwise) until the frequency passes 20kHz.Press the rotary encoder to change the filter type. Select the type by turning the rotary encoder. You can choose from: Butterworth 6dB, Bessel 6dB, Butterworth 12dB, Bessel 12dB, Linkwitz Riley12dB, Butterworth 18dB, Bessel 18dB, Butterworth 24dB and Bessel 24dB. Higher order filters may be set by adding filter sections in the PEQ blocks (see below).

```
Out12 Low Pass    Out12 Low Pass
Freq: 14500Hz     Type: But24
```

3.3.5. High Pass Filter

Adjust the high pass frequency by dialing the rotary encoder. You can switch the high pass filter off by dialing the rotary encoder down (counterclockwise) until the frequency passes 20Hz.Press the rotary encoder to change the filter type. Select the type by turning the rotary encoder. You can choose from: Butterworth 6dB, Bessel 6dB, Butterworth 12dB, Bessel 12dB, Linkwitz Riley12dB, Butterworth 18dB, Bessel 18dB, Butterworth 24dB and Bessel 24dB. Higher order filters may be set by adding filter sections in the PEQ blocks (see below).

```
Out12 High Pass          Out12 High Pass
Freq: 34Hz               Type: Bes12
```

3.3.6. Parametric Equalizer (PEQ)

There are 10 bands of parametric equalization. Each band can be adjusted freely over the frequency range of 20Hz to 20kHz. Adjust the frequency by dialing the rotary encoder. Press the rotary encoder to select the parameters. The available parameters are: Frequency (20Hz to20kHz), Gain (-12dB to +12dB), Q (0.2 to 25), Enabled (On or Off), Type (Bell, High Shelf, Low Shelf, Notch,All Pass, Band Pass, High Pass, Low Pass). For the Shelving filters, the Q value sets the steepness of the filter in dB/Oct.

```
Out12 PEQ 5              Out12 PEQ 5              Out12 PEQ 5
Freq: 14500Hz           Gain: -7.75dB            Q:      1.5

Out12 PEQ 5             Out12 PEQ 5
Type: Bell             Enabled: On
```

3.3.7. Compressor

The compressor is a true RMS compressor. Turn the rotary encoder to set the threshold. Press the rotary encoder to select the parameters. The available parameters are: Threshold, Attack, Hold, Release, Ratio,and Makeup Gain.

```
Out12 Compressor        Out12 Compressor        Out12 Compressor
Thr.: 23.00dBu          Att.: 50ms              Hold: 10ms

Out12 Compressor        Out12 Compressor        Out12 Compressor
Rel.: 500ms             Ratio: 1:4.5            Gain: 4.75dB
```

3.3.8. Limiter

The limiter is a zero-attack peak limiter. Only the threshold and release can be set. Press the rotary encoder to select the parameters. The release value is displayed in dB per second.

```
Out12 Limiter           Out12 Limiter
Thr.: 23.00dBu          Rel.: 50dB/s
```

3.3.9. Phase Inversion

Turn the rotary encoder to switch phase inversion on or off.

```
Out12 Invert            Out12 Invert
Off                     On
```

3.3.10. Channel Link

By linking 2 channels, the settings are guaranteed to be identical for both channels, except for mixer and mute. Turn the encoder to set the Channel Link on or off.

```
Out12 Link        Out1 Link
On                Off
```

3.4. System Menu

Push the Menu button to enter the system menu. Push the Menu button again to toggle through the menu items. The available menu items are:

3.4.1. Load Preset

Turn the rotary encoder to select the preset you wish to load. Push the rotary encoder to select it, then turn it again to select "YES" and confirm again by pushing the rotary encoder knob. The preset is loaded and all settings are applied immediately. All settings that were in the unit prior to loading the preset will be erased.This action can not be undone.The presets contain all filter, dynamics, gain settings etc; in other words, everything that makes out a loudspeaker configuration. Presets do NOT contain the name of the unit, network configuration, automatic standby delay, user access rights and passwords.

3.4.2. Save Preset

Turn the rotary encoder to select the location of the preset. If that location is not empty, the preset in that location will be overwritten. This action can not be undone.The system will ask you to give a name to the preset name. Select a character in the cursor position by turning the rotary encoder; pushing the rotary encoder confirms the character selected and moves the cursor to the next character to edit. Pushing the exit/ESC button erases the last confirmed character. Once the name is set up, push the menu button again. If you want to continue storing the preset, select Yes by turning the rotary encoder and confirm by pushing it. To cancel, push the exit button. Now your preset is saved in the selected location.

The presets contain all filter, dynamics, gain settings etc; in other words, everything that makes out a loudspeaker configuration. Presets do NOT contain the name of the unit, network configur-ation, automatic standby delay, user access rights and passwords.

3.4.3. Access Level

The ASP-4M8 has the option of locking away the front panel controls to avoid tampering of settings by unauthorized persons. To lock the unit, select "Locked" by turning the rotary encoder, and push it to confirm.The system will ask you to enter a password. Select a character in the cursor position by turning the rotary encoder; pushing the rotary encoder confirms the character selected and moves the cursor to the next character to edit. Pushing the exit/ESC button erases the last confirmed character. Once the password is set up, push the menu button again and the unit will be locked.

ATTENTION: Make sure to remember the password! When the unit is locked and you forgot the password, it is not possible to unlock it without contacting your local service representative.To unlock the unit select "Unlocked" and enter the password. The password is automatically checked after each entered character, and the unit will exit the system menu when the password is confirmed.
The default password is "Password".

3.4.4. Version Information

By pushing the rotary encoder, the display toggles through the version information about the unit (serialnumber, firmware version), as well as some parameters like IP address, subnet mask, MAC address etc.

Using FIR and RTA

Introduction

The ALLCONTROL Software provides FIR calculation functions for hardware that supports FIR filters, and an RTA (Real Time Analyzer) option that can be used in combination with the FIR filter, but is independent from the hardware. This document describes the use of both options. It is assumed that you know what a FIR filter is, what the difference between IIR and FIR filters is and that you understand the concepts of truncating, convolution, group delay and related topics, as well as transfer function, power compression, directivity etc.

Using the RTA function

The RTA enables you to visualise the spectrum of the audio signal. By using this to measure the output of the loudspeaker producing white noise, you can find the transfer curve of the loudspeaker. Although other applications exist, we will focus on this application.

To measure a transfer curve, connect a signal generator playing white noise, or select "White Noise" from the input selection drop-down in the ALLCONTROL software (selected hardware only). Adjust the volume to a reasonable level; there is no need for high SPL measurements unless you want to test power compression.

Now go into the channel you want to adjust. For a 2-way system this may be the HF or LF channel, or it may be the input channel to adjust the overall transfer function. If you plan to use the RTA measurement to control a FIR filter, choose a channel that supports FIR.

STARTING A MEASUREMENT

First you need to enable the RTA function. To do so, click the RTA icon in the graphical window.
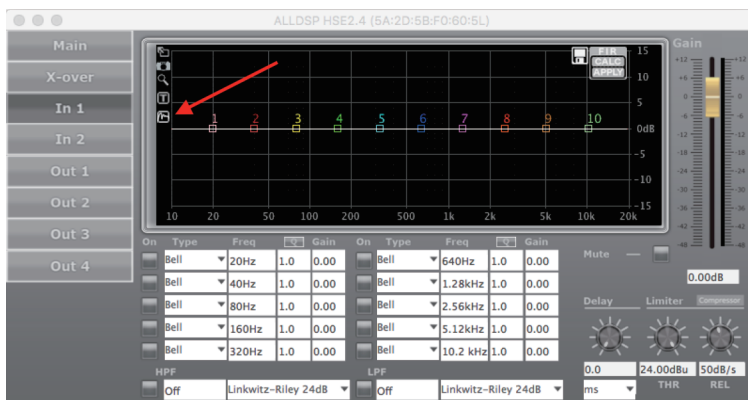
The RTA function will be enabled. To start the measurement, click Play:
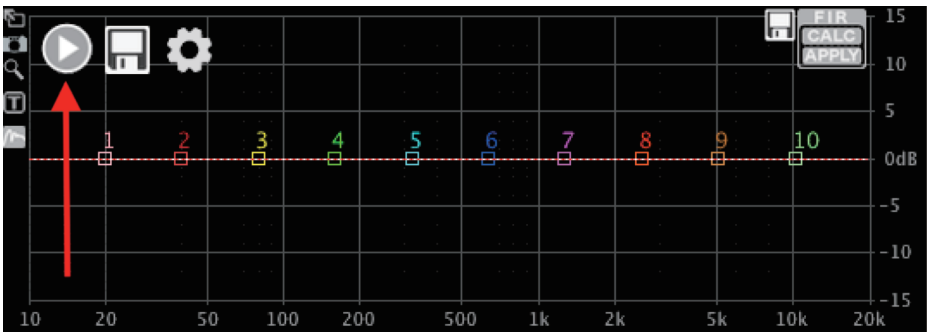
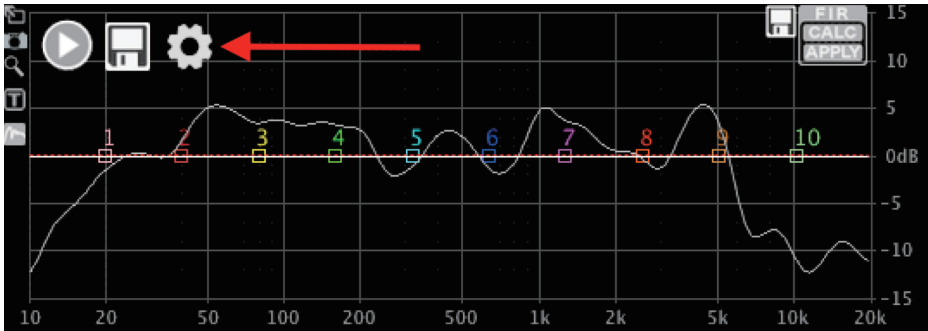This will start the RTA with the default settings. A measured response will be shown:
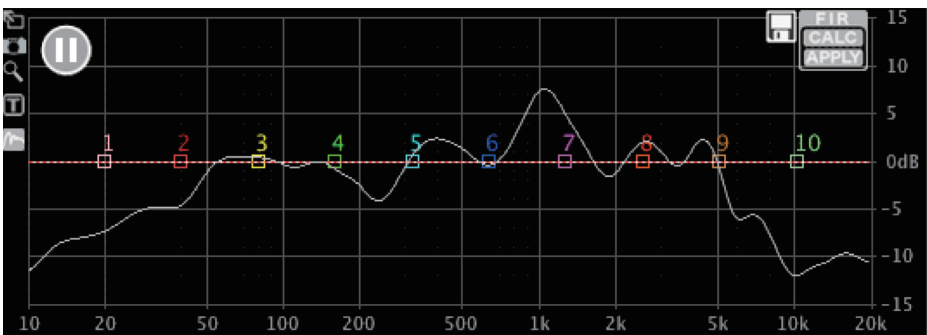
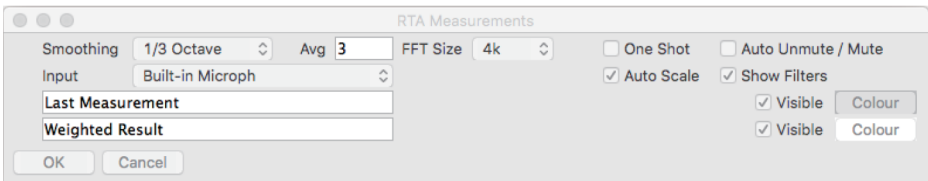Click Pause to stop measuring:
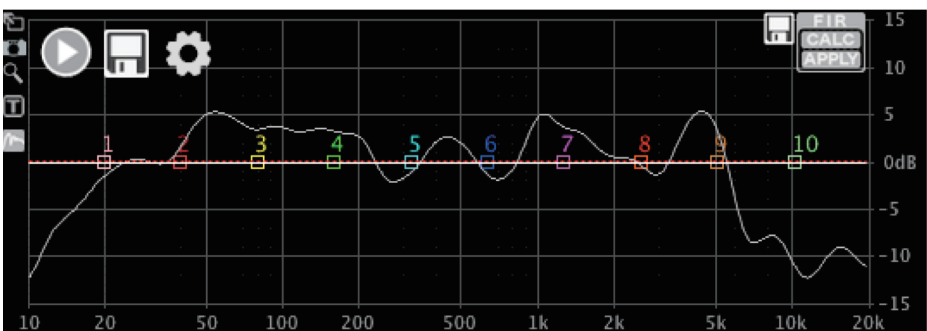
MEASUREMENT OPTIONS

Now that you have made a first measurement, let's have a look at the measurement options.

Click the Settings button:

A window will open with the measurement options:

- ➤ Smoothing: Choose the frequency smoothing. For manual measurements and settings, 1/6th or 1/3 Octave is recommended. For automatic FIR adjustment, this may lead to unexpected results due to room interference; in this case, a setting of 1/2 to 1 Oct is recommended.

- ➤ Avg: Choose the number of measurements that are averaged. The averaging function is a moving average over Avg samples. For fast indications, 1 to 5 is OK; for accurate measurements, you may want to increase this value to 10 or more.

- ➤ FFT size: The size (length) of the sampled data. With a size of 4k, lower frequencyes are not accurately measured, but the measurement is fast. Larger size leads to more accuracy in the low region, but longer measurement times. To see the accuracy, try setting the smoothing to "Off"; then you will see the actual data points.

- ➤ One Shot: When selected, one reading consisting of "Avg" measurements will be taken. When deselected, the measurement is continuous.

- ➤ Auto Mute / Unmute: When selected, the software will automatically mute the hardware at the end of each measurement, and unmute it before the next measurement. Use this in combination with the One Shot setting to go easy on your neighbours.

- ➤ Input: Select the source fo the measurement. A proper measurement microphone is recommended; the measurement is only as good as the microphone.

- ➤ Auto Scale: Manual scale is not supported yet. Please leave this check box selected.

- Show Filters: When deselected, the filter (PEQ, HPF, LPF, FIR) response curves will not be drawn.Below the options, a list of saved measurements appears. When no measurements have been saved yet, only the default curves are listed:

- Last Measurement: This is the as-yet-not-saved measurement. Select if it is visible or not by clicking " Visible"; select the colour by clicking "Colour".

- Weighted Result: When saving multiple measurements, these can be combined into one Weighted Result. Select visibility and colour here.Feel free to play around; it will not explode.

SAVING AND COMBINING MULTIPLE MEASUREMENTS

After making a measurement, look at the curve and see if it is plausible. Measurements can be distorted by nearby objects (walls,floor, people, bookshelves). Try measurements in different places to see what happens. When you are satisfied with a measurement, click the Save button.
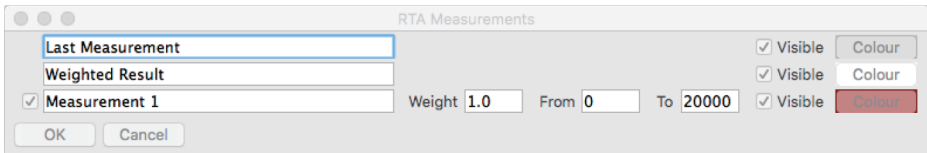
One more line is added to the measurements. You can set some options:

- Enabled: When deselected, this measurement will not be visible and will not be taken into account for the weighted result.

- Name: You can set the name for each measurement.

- Weight : ht of eThe weigach measurement determines the relative importance of each measurement when calculating the weighted result. For instance: When you have one measurement with weight 2 and another with weight 1, the first measurement will be twice as influential in the result. Choose any number other than zero.

- From: The bottom end of the frequency range where this measurement is used. More on this on the next page.

- To: The top end of the frequency range where this measurement is used. More on this on the next page.

- Visible: When deselected, the measurement will not be visible, but will still be taken into a ccount for the weighted result.

- Colour: Select the colour of the displayed curve plot.

Normally, one would make several measurements, e.g. for different listening positions to take the room into account, or from different angles w.r.t.the loudspeaker to take the directivity of the loudspeaker into account. I will not go into the details of why and how to measure for optimal results; that is not the scope of this document, and there is plenty of information on that in the Internet.

After saving several measurements, the Settings window looks like this:
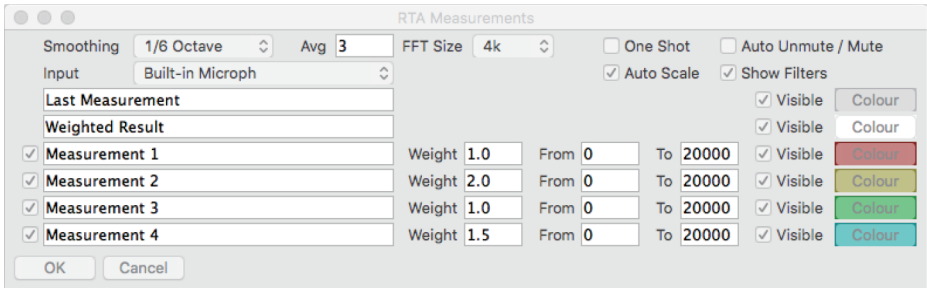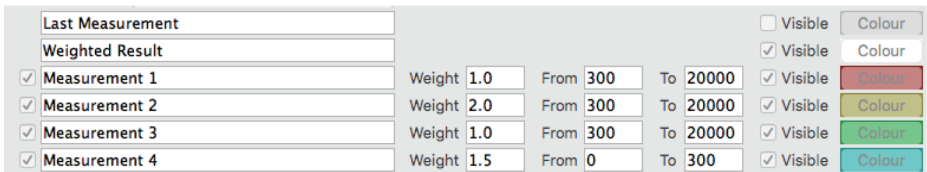

And the graphical window like this:
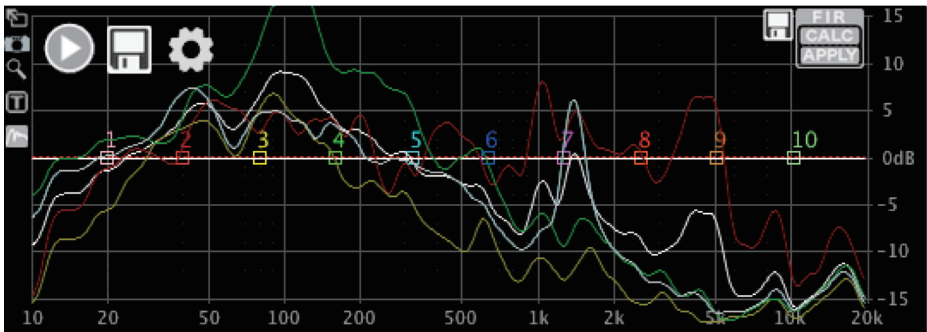
Now you can combine these measurements into one averaged result. In the image above, measurements 2 and 4 have a greater weight than the other two; these would be e.g. on-axis results, or my favourite listening position.

Now let's assume that measurement 4 （the blue one, also the last one, so shown light grey here) was done close to the woofer, and I think this data is very useful for the low end up to 300Hz, but irrelevant for the rest of the frequency spectrum; and the other measurements are relevant for 300Hz and higher. I can se the "From" and "To" fields like this:

I also set the Last Measurement to invisible so as not to be in the way, since I am done measuring for now.

The graphical window will look like this:
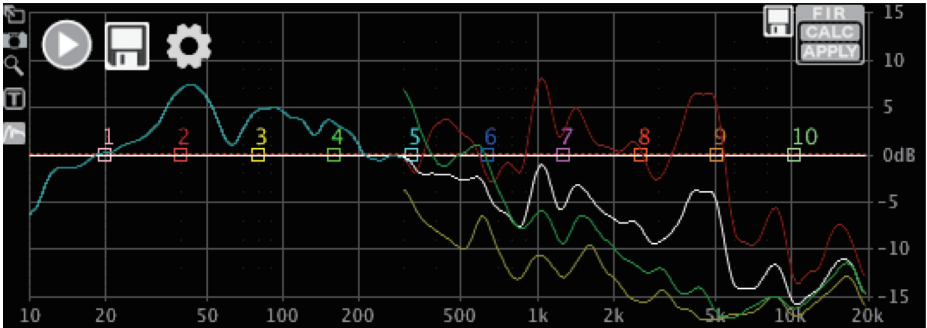
For more detail, I will go to Full Screen Mode. Click the Full Screen button:
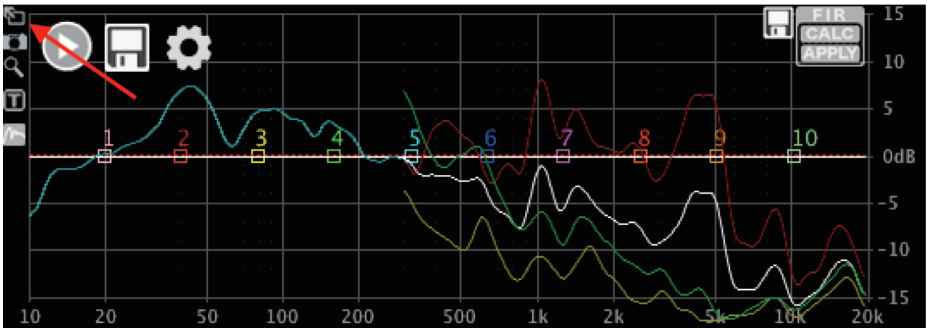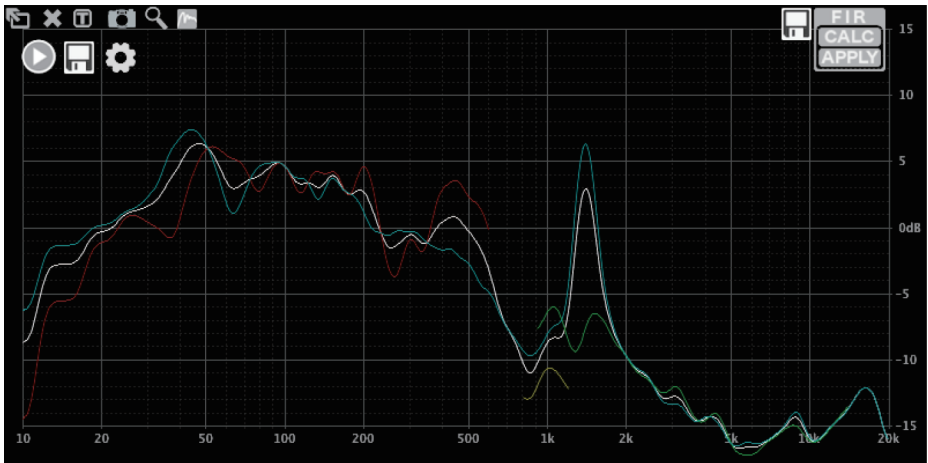

The graphical window will fill the size of the unit panel. Click the same button again to use the whole display area for maximum detail.
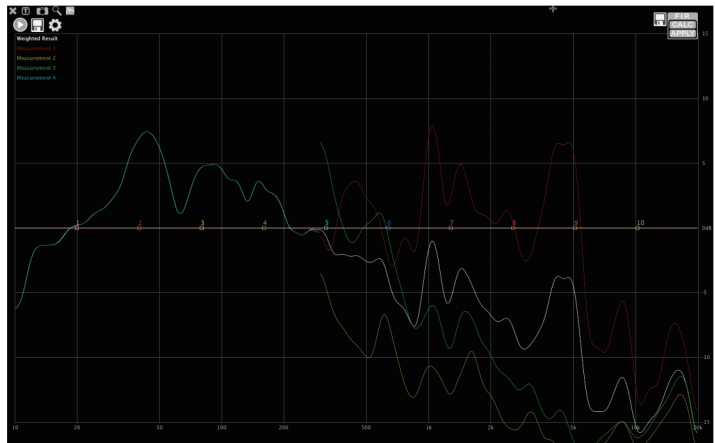
You can play around with different frequency and weight settings to reach the desired weighted result. One more example: Settings:
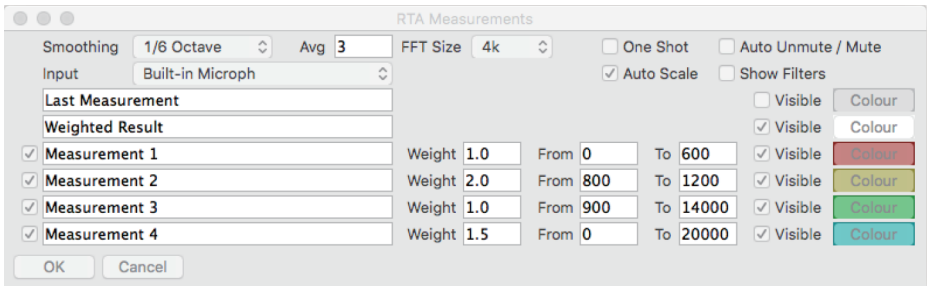

And the result:

Using FIR filters

There are 3 ways to define a FIR filter in ALLCONTROL: Load from a file, create by using PEQs and crossovers to draw a target response, and by inverting the RTA measurement.

When a unit is connected, the filter that is loaded will be shown on the display as a white line. In this example, no FIR filter has been loaded yet. Let's change that, shall we?

CREATING A FILTER FROM PARAMETRIC EQUALISERS

First, set some PEQs to create a target response, then click "CALC" in the FIR corner:
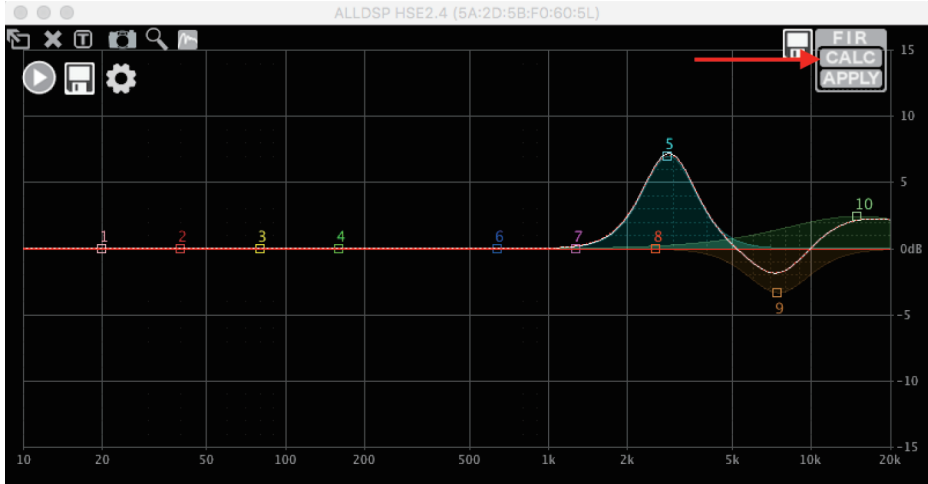
A pop-up window will appear with some options:

- Number of taps: You can select how many taps of the FIR filter are used. More taps means more accuracy in the low end, but also means that the filter will cause more latency. The filters we generate are all linear phase with a symmetrical impulse response, and the latency is therefore equal to half of the filter length.

- Currently active filter: You can add to the filter that is currently active in the hardware, or you can start from scratch by deselecting this option.

- HPF, LPF and PEQs: You can choose which filter will be converted to FIR and which will be ignored. It is common practice to use IIR for low frequencies and FIR for higher frequencies to keep the latency down to a minimum.

- Track changes: The FIR calculation will be updated automatically as you change PEQs.

- Invert RTA measurement: We will come to that later.

- Data from File: We will come to that later.
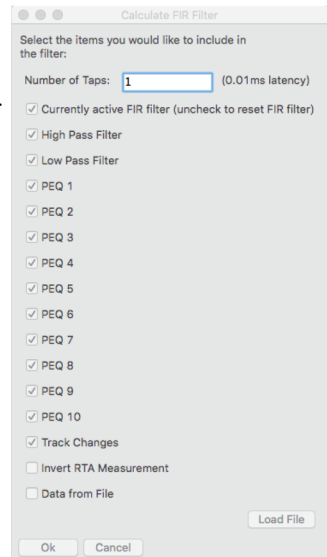
Select 256 taps (the maximum number for the hardware that is used for this example), deselect "Currently Active FIR filter" and click OK. In the graphical screen you will see a dotted red line indicating the calculated FIR response. As you can see in the screenshot above, it is perfect as far as the display resolution allows us to see. The red dotted line is the calculated response from the FIR coefficients; so any deviations will be visible here. This becomes clear when we try to use this FIR filter to change the low frequency response:
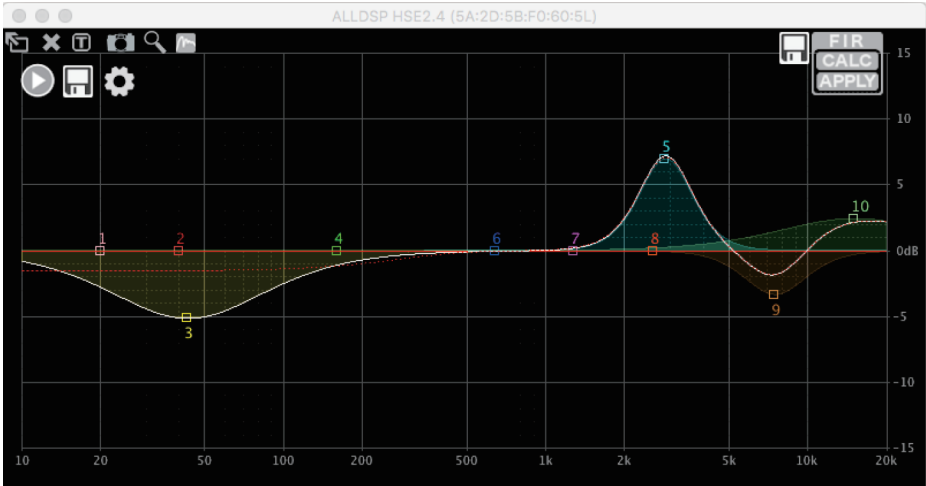
As you can see, the deviation below 200Hz is quite serious. In other words: A 256 tap FIR filter is useless in this frequency region. Which is to be expected, it is caused by the "finite" in the name FIR. You can use the red dotted line to see the result of the approximation.

INVERTING THE RTA MEASUREMENT

We can also create a filter by inverting the RTA measurement. Using the measurement we did in the previous chapter, the result looks like this:
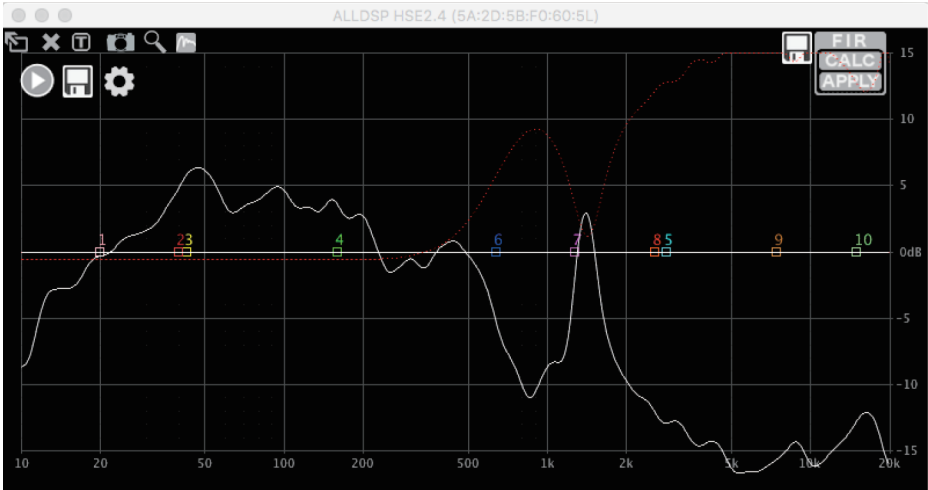
As you can see, it only works above appr. 300Hz. Below that, the response is automatically reduced, taking the amount of taps into account. The red dotted line again shows the expected response.

LOADING A FILTER FROM A FILE

Click CALC, then "Load File". Select your file. If the number of taps in the file fit in the FIR filter, it will be loaded automatically.   If the number of taps exceeds the available filter size, the filter will be truncated. ALLCONTROL supports 24 and 32 bit fixed point files, and floating point files. Any lines that do not contain a number (such as comments) will be ignored. You can generate FIR files with many 3rd party programs, or even online. Here is a 1k HPF,   generated with ScopeFIR:
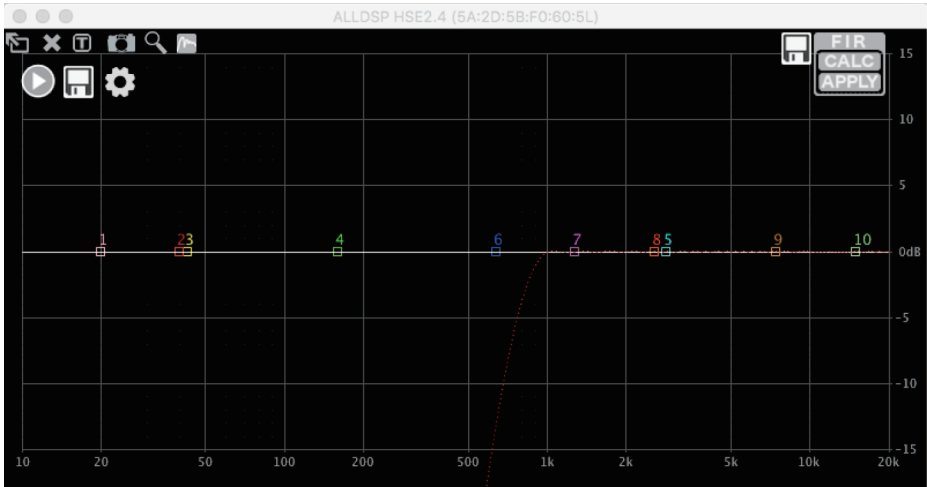

Again, the dotted red line indicates the calculated frequency response.

## SENDING A FILTER TO THE HARDWARE

When you are satisfied with the response indicated by the red dotted line, click APPLY to send the filter to the hardware. A window will appear with two options:
- Reset converted IIR filters: When selected, the software will automatically disable the IIR filters that were converted to FIR.
- Do not show this dialog again: Speaks for itself,doesn't it?
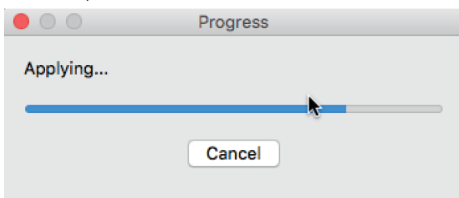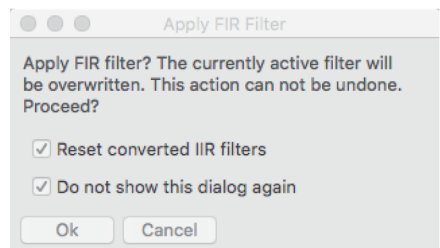
The communication can take some timeto complete,depending on filter size and connection speed.

If no other filters are active, the response will be shown as a white line:
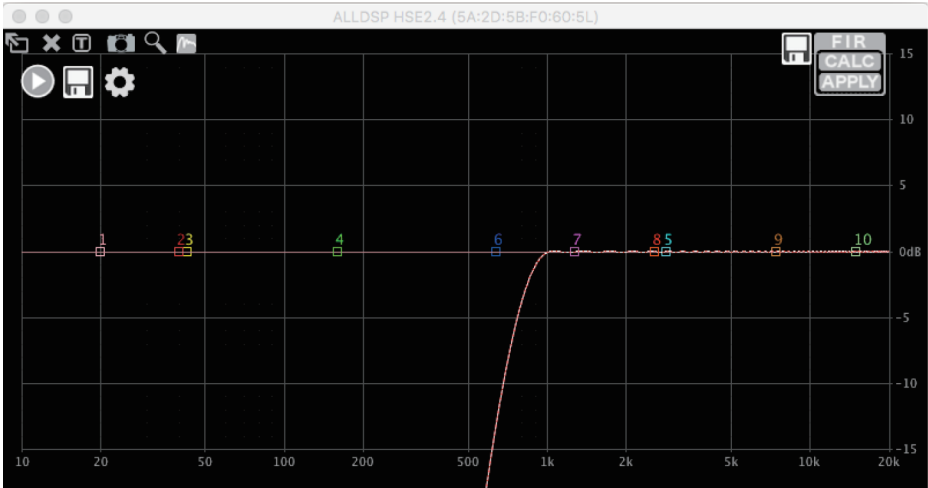
When you add PEQs now, you will see the white line will follow the PEQs, as well as the red dotted line; the red dotted line is the newly calculated FIR filter, following the changes you are doing with PEQs, and the white line is the overall frequency response. If some PEQs are not taken into account in the FIR calculation, the white line will include them (because that is the overall frequency response) but the red dotted line will not. There will also be a solid red line, which is the response of the FIR filter that is currently loaded in the hardware:
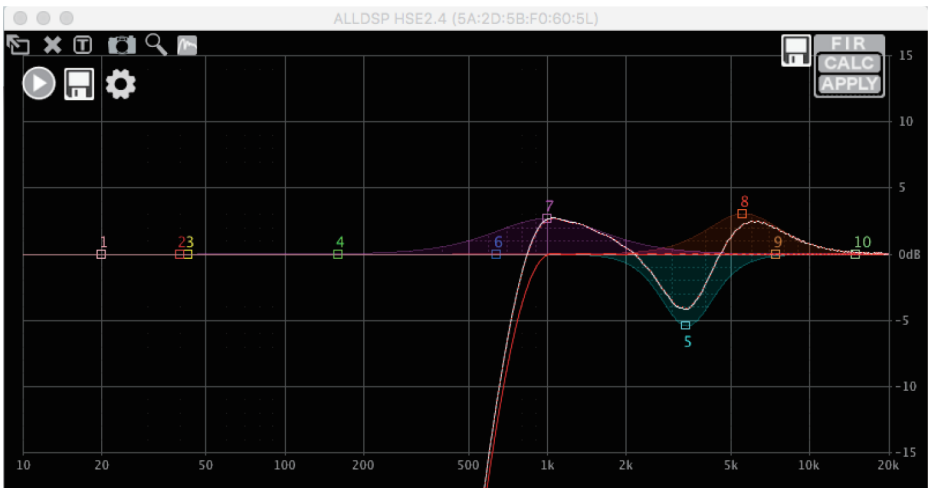
SAVING FILES

Click the disk icon next to the FIR block to save a filter as a text file. The filter that is loaded in the hardware will be saved to disk, and can be loaded into other channels or saved for later use.
Here is the FIR file of this example (omitting most coefficients):
FIR filter generated by LAX DSP control software. Sample rate: 48828Hz, 255taps

-0.00251117721317
0.000136785209243
0.000140666030413
0.000148858875106
0.000161494128481
0.000177634880029
0.000197263434621
0.000219509005649
0.00024409592163
0.000270059332377
0.000297261402149
0.00032448116705
0.000351532362565
0.000377209857282
0.000401332974621
0.000422636978525
0.000440970994737
0.000454998575363
0.000464653596498
0.000468616374055
0.000467024743774
0.000458568334793
0.000443564728109
0.00042071472873
0.00039056222904
0.000351847149596
0.000305717811131
0.000250644981978
0.000188274309127
0.000117114745135
3.95923853105e-05
-4.697032275e-05
-0.000135075300995
-0.000237385742477
-0.000340500846664
-0.000444556586651
-0.000553787686189
-0.000663964078139
-0.000775675289694
-0.000885844231064
-0.000994163565801
-0.00109791569463
-0.00119658745881

Conclusion

We believe that these functions provide you with powerful tools to use FIR filters and RTA measurements conveniently.   Development is ongoing (as always), so functionality will be added in the future.

LAX®

Warranty Card

锐丰企业 RUIFENG CORPORATION

Customer Hot Line:
400-707-6800

LAX C.V. products which produced by RUIFENG Corporation(RUIFENG in brief) are ensured with highquality by strict quality inspection and contr -ol before delivery.our will provide warranty according terms mentioned in this warranty card. The warranty is viod if warranty card or purchasing invoice is missed.The service is suggested to start by calling our service No.:400-707-680 in advance.

一、This warranty is valid within one year form date of sale (except sone occasions specially mentioned as below),and covers for manufacturing defects and all matters concerning the workmanship of LAX&C.V. products。

二、Warranty period for particular item。
　　　1、offering one month free change service and one year warranty for amplifiers,mixers or processors.
　　　2、Offering three months warranty for which categorized as spare parts:
　　　1.)The voice coil and diaphragm of microphone.
　　　2.)The voice coil and diaphragm of speaker.

C、RUIFENG willnot be responsible for any warranty if/for：
　　1、Non-this company proxy product；2、Fake products；
　　3、Products without intact number or serial number is changed,deleted or transferred or has not been recorded in the company s record。

D、Thewarranty becomes void if product problems result from；
　　　1、improper operation and misuse(overloading or bumed with feedback)；
　　　2、Careless moving.
　　　3、Terrible conditions(over-higt input voltage and humidity)
　　　4、Irresisitible situation such as STORMor EARTH QUAKE…etc；
　　　5、repaired or restructured by unofficial technicians outside RUIFENG ；
　　　6、Failure assemble/installation from the instructions of manufacturer。

E、The warranty applies only on the cost of repair or replacement and does not include the relevant freight charges。

F、RUIFENG is not responsible for the loss caused by equipment or component damaging。

G、RUIFENG charges reasonably if warranty period expired。

H、The warranty card is subject to change without prior notice。

Purchaser Name:	Manufacturer:

Dealer Name:	Dealer Address:
	City　　State　　ZIP

Purchase Date:	Model:	Serial No.:	Invoice No.:
(d)　(m)　(y)			

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CERTIFICATION

Model

QA

Date

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RUIFENG CORPORATION

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