Realiser A16 User Manual

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User Manual v2.0 for A16 firmware v2.05
10/20/21

The Realiser A16 firmware may be updated from time to time to fix bugs, make improvements, and add new features. This manual conforms to firmware revision 2.05 (20 October 2021). Please check regularly at: www.smyth-research.com/downloads for firmware and user manual updates going forward. Please note there may be significant differences between the operation described here and that for other firmware versions.

Contents

1 Safety

IMPORTANT SAFETY INSTRUCTIONS

READ BEFORE OPERATING EQUIPMENT

• Read these instructions.

• Keep these instructions.

• Heed all warnings.

• Follow all instructions.

• Do not use this apparatus near water.

• Clean only with a dry cloth.

• Install in accordance with the manufacturer's instructions.

• Do not install near any heat sources such as radiators, heat registers, stoves or other apparatus (including amplifiers) that produce heat.

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

• Only use attachments/accessories specified by the manufacturer.

• Unplug this apparatus during lightning storms or when unused for long periods of time.

• 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.

• Never expose the equipment to rain or a high level of humidity. For this reason do not install it in the immediate vicinity of swimming pools, showers, damp basement rooms or other areas with unusually high atmospheric humidity.

• Do not use the device outside. To reduce the risk of fire or electric shock, do not expose this/these device/s to rain or moisture.

• Never place objects containing liquid (e.g. vases or drinking glasses) on the equipment. Liquids in the equipment could cause a short circuit.

• Lay all connection cables so that they do not present a trip hazard.

• Check whether the specifications comply with the existing mains supply. Serious damage could occur due to connecting the system to the wrong power supply. An incorrect mains voltage could damage the equipment or cause an electric shock.

• Never place open flames near the equipment.

• If the equipment causes a blown fuse or a short circuit, disconnect it from the mains and have it checked and repaired.

• Do not open the equipment without authorisation. You could receive an electric shock. Leave all service work to authorised expert personnel.

• Do not hold the mains cable with wet hands. There must be no water or dust on the contact pins. In both cases you could receive an electric shock.

• The mains cable must be firmly connected. If it is loose there is a fire hazard.

• Always pull out the mains cable from the mains and/or from the equipment by the plug, never by the cable. The cable could be damaged and cause an electric shock or fire.

• If the power cable is connected, avoid contact of the unit with other metallic objects.

• Do not insert objects into openings. You could damage the equipment and/or injure yourself.

• Do not use the equipment if the mains plug is damaged.

• When installing the device into a 19" rack, make sure that the mains switch, mains plug and all connection on the rear of the device are easily accessible.

• When connecting the headphone do not place the headphone on your head until you can establish the volume of the sound being played.

• When connecting the headphone, ensure that the volume is turned down to minimum. Adjust the volume after putting on the headphone. Keep the headphone volume at moderate levels. Hearing damage accumulates over a lifetime of listening at even slightly excessive volumes.

• Connecting and disconnecting cables, choosing menu items, and any adjustments should be done at a low volume setting and with the headphones off your head, to avoid being exposed to high level sounds that could cause hearing damage.

• With wired headphones you should avoid sharp movements, which could cause the headphone to fall off your head. You could be seriously injured especially if you are wearing pierced earrings, spectacles etc. The cable could wind around your neck and cause strangulation.

• Take the headphones off when changing presets, until you are familiar with the presets.

2 Introduction

2.1 Realiser A16 operational design

2.1.1 The Problem: Listening to multichannel audio over headphones.

The Realiser A16 has been designed primarily to allow multichannel immersive audio to be heard accurately through stereo headphones.

Today, almost all immersive audio content is monitored over loudspeakers during production, not through headphones, and therefore loudspeaker reproduction, in an acoustically controlled room, is the preferred method for listening to immersive audio.

Since sound is essentially heard through our two ears, headphone playback is theoretically capable of reproducing the same loudspeaker listening experience. For headphones to accurately mimic loudspeakers, digital signal processing must be used to create multiple virtual loudspeakers in a virtual acoustic environment. The individual audio signals are filtered through these virtual loudspeakers and room, and then summed together to create a binaural signal that is played over regular stereo headphones.

If the virtual loudspeakers filters are personalised to an individual, the final reproduction through headphones is remarkably accurate when compared directly to the loudspeaker reproduction: the spatial positioning of each source is maintained, stereo imaging between the virtual speakers is preserved, and the reverberation of the room and the overall timbre of the sound is the same.

One problem with headphone reproduction is that the sound stage rotates with the listeners head causing significant aural confusion and loss of spatialisation. A solution to this rotation problem is the use of head tracking technology. The head tracker tracks the listeners head movement, adjusting the virtual speaker processing such that the virtual speakers move in the opposite direction to the listener's head. This results in a very stable and natural listening experience causing the headphones to effectively disappear.

2.1.2 The Solution: Realiser A16

The Realiser A16 implements the personalised-head-tracked Smyth Virtual Surround (SVS) technology for headphone rendering. The A16 renders up to 24 virtual speakers in single-user operation or up to 16 virtual speakers for dual-user operation. The A16 also includes a suite of routines that allow users to create personalised measurements of real loudspeakers using the supplied microphones, which can then be used to create virtual listening rooms for almost any presentation format. Factory default measurements are also included for out-of-the-box operation. Headphone equalisation routines are also included to maximise the performance of the listeners headphones.

Internal Dolby Atmos immersive audio bitstream decoding up to 24 channels is supported as standard. DTS:X decoding for up to 12 channels is available as an upgrade. Decoded audio can either be sent to the SVS processor for rendering over headphones, or it can be sent to the auxiliary outputs (AV mode). Multi-channel audio can be input directly and rendered to headphones - multi-channel PCM via the auxiliary AES-EBU (16ch), Dante (16ch), USB (16ch), HDMI (8ch), stereo (2ch) or SPDIF (2ch) interfaces, or multi-channel analogue via the auxiliary single-ended or balanced interfaces.

Optional auxiliary 16ch balanced 2U version

Optional auxiliary 16ch AES-EBU 2U version

Optional auxiliary 16ch DANTE 2U version

2.1.3 Realiser A16 specifications

SVS loudspeaker virtualisation with integrated headtracking (16ch 32bit floating-point processing@48kHz sampling rate, processing latency 32ms, maximum reverb length 750ms). All source signals above a sampling rate of 48kHz are down sampled to 48kHz.

Virtualisation sources:

1. HDMI inputs (1-4): Dolby Atmos, DD+, DD, DTS*, DTS-X*, 8ch LPCM (24bit@48/96/192kHz)

2. SPDIF input (coaxial or optical): Dolby Digital, DTS*, 2ch LPCM (24bit@48/96/192kHz)

3. USB 2.0 input: 16ch LPCM (24bit@96kHz)**

4. Auxiliary Input/Output options (only one can be installed):

a) 16ch single-ended line-level (2Vrms, -115dB DR)

b) 16ch balanced line-level (+18dBu, -115dB DR)

c) 16ch digital AES-EBU (24bit @96kHz)

d) 16ch digital Dante (24bit @96kHz)

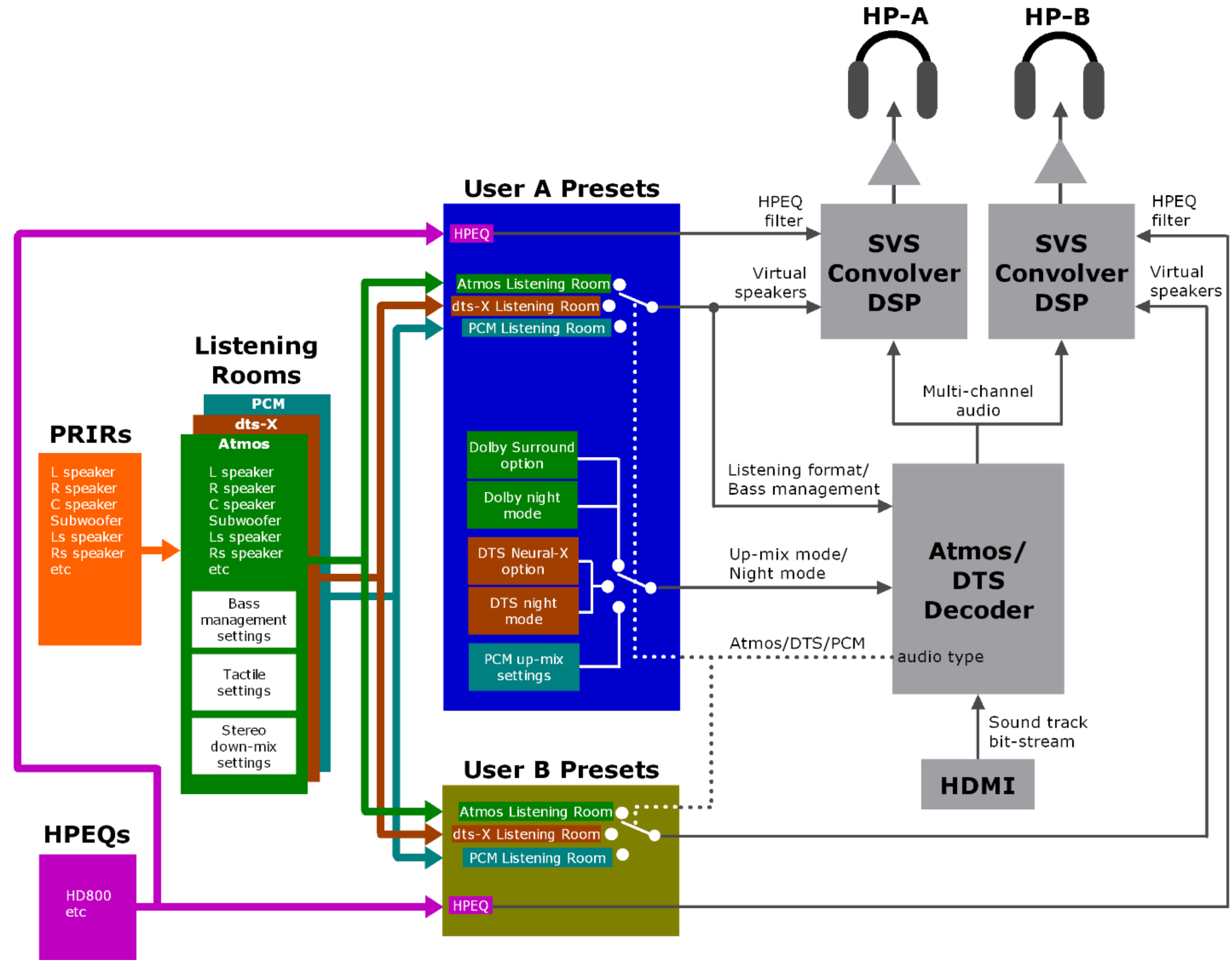
5. Stereo line inputs: 2ch (2Vrms, -105dB DR)

*Where the user has purchased the DTS-X decoding upgrade.

**The USB interface only functions with Mac OS 10.12 or older The USB interface only functions on Windows PCs in conjunction with the supplied ASIO driver.

2.2 Realiser A16 operational overview

- *The A16 runs on **Presets** – each preset contains three different **Listening Rooms.***
- ***Listening Rooms** are designed for specific audio decoding formats, and for specific loudspeaker arrangements within these formats. Listening rooms contain a maximum of 24 virtual loudspeakers.*
- ***PRIRs** contain the binaural room impulse measurement data from real loudspeaker sources that is used to generate virtual loudspeakers. PRIRs can contain measurements of up to 64 virtual loudspeakers.*



2.2.1 Presets

User presets contain three types of virtual listening rooms, Atmos, dts:X and PCM. Atmos rooms are intended to be used when a Dolby bitstream is being decoded, or when PCM audio is being up mixed by Dolby Surround. Dts:X rooms are intended to be used when a DTS bitstream is being decoded, or when PCM audio is being up mixed by Neural:X. PCM rooms are intended to be used when PCM is being received. The A16 processor can either switch dynamically between the different rooms as the input audio type changes, or the rooms can be selected manually by the user.

2.2.2 Listening Rooms

A virtual listening room can contain up to 24 virtual loudspeakers arranged in a wide range of speaker layouts. The virtual loudspeakers that make up the virtual listening room are derived from personalised room impulse response (PRIR) files held in the A16 PRIR directories. During Atmos and dts:X* bitstream decoding, the speaker layout specified by the listening room sets the rendered listening mode in the object orientated decoder. Similarly, when PCM is being up mixed by either Dolby Surround or DTS Neural:X*, the speaker layout of the virtual listening room sets the up-mixed audio format.

*Where the user has purchased the DTS-X decoding upgrade.

2.2.3 PRIRs

PRIRs contain room impulse response measurement data of loudspeakers within a sound room captured either by an individual or using a dummy head. PRIRs typically comprise several speaker positions, for example left front, centre, right front, etc. To accommodate head-tracking each PRIR speaker position typically consists of binaural measurements for multiple head orientations (described herein as ‘look angles’). A single PRIR can contain up to 64 ‘virtual’ loudspeakers and up to 23 look angles for each.

2.3 *Unpacking and parts assembly*

2.3.1 Unpacking

The Realiser A16 package contains the items below.

Main processor components:
1. Realiser A16 processor (either the 2U 19” rack-mountable version or the HS version)
2. Power Supply (input 100-240V AC, 50/60 Hz, output 12V DC @ 3A) *
3. IR remote control

Set-top head-tracking components:
4. Set-top IR reference unit (for head-tracking)
5. Set-top cable (3.5mm plug to 3.5mm plug, 4-pole)
6. Set-top extension cable (3.5mm socket to 3.5mm plug, 4-pole)

Head-top head-tracking components:
7. Head-top head-tracker
8. Clip for mounting the head-top to headphones
9. Rubber bands (for connecting head-top clip to headphones - 3 sizes)
10. Head-top cable (2.5mm plug to 2.5mm plug, 4-pole)
11. Head-top extension cable (2.5mm socket to 2.5mm plug, 4-pole)
12. Cable clips (to attach the head-top cable to the headphone cable)

Measurement microphone components:
13. Lanyard (orange neck strap)
14. Clip for lanyard (for supporting in-ear microphones during PRIR measurements)
15. In-ear microphones (one pair)
16. Foam earplugs (for microphones - 3 sizes)
17. Grounding wrist strap (for earthing body during PRIR measurements)

18. Headband for mounting a Head-top device during PRIR measurements

For the Realiser A16 2U processor there is an optional accessory.

* The power supply is designed for any mains voltage and frequency and is provided with a mains cord appropriate for the market to which the Realiser is shipped.

Optional accessories:
19. 2U 19” rack-mount ears

2.3.1.1 Unpacking: Main processor components

Realiser A16 2U version

Realiser A16 HS version

Universal power supply (100-240V, 50/60Hz)

Remote control (IR)

2.3.1.2 Unpacking: Set-top head-tracking components

Set-top IR reference for head-tracking

Set-top cable (*3.5mm plug to 3.5mm plug, 4-pole)*

Set-top extension cable *(3.5mm socket to 3.5mm plug, 4-pole)*

2.3.1.3 Unpacking: In-ear measurement microphone components

Lanyard *(for supporting microphones during measurements)*

Microphone cable support clip *(connects to the lanyard and provides strain relief to the microphones when inserted in the ear canals)*

In-ear SVS measurement microphones *(one pair)*

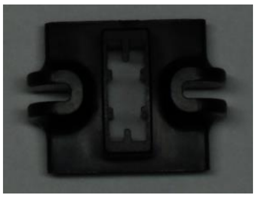
Ear foam *(seals the SVS microphones when inserted in the ear canal – 4 pairs in 3 sizes)*

Grounding wrist-strap *(used during SVS microphone use to protect against static discharge damage)*

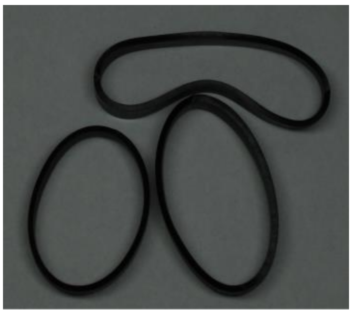
Head tracker Head-band

2.3.1.4 Unpacking: Head-top head-tracking components

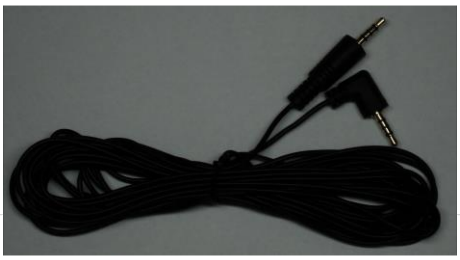
Head-top head-tracking device



Clip for mounting the head-top device *(connects to the headphone head-band)*



Rubber bands (3 sizes) *(connects the clip to a headphone head-band)*



Head-top cable (*2.5mm plug (RA) to 2.5mm plug, 4-pole*

Head-top extension cable *(2.5mm socket to 2.5mm plug, 4 pole)*

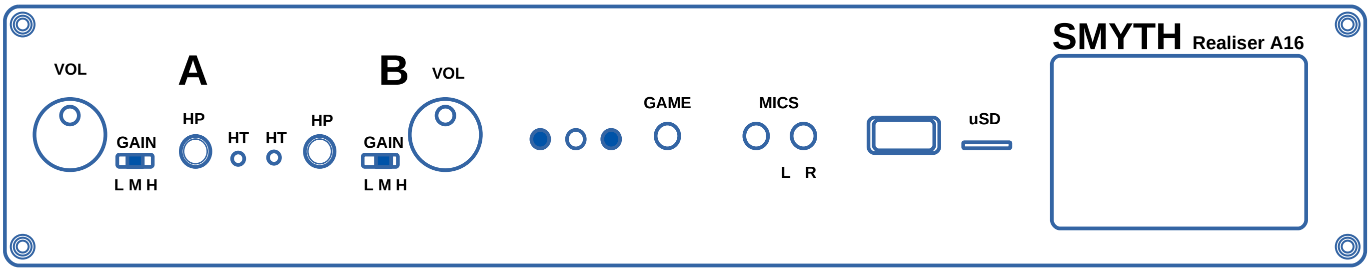
Cable clips to connect the head-top cable to the headphone cable *(3 sizes for circular headphone cables, 2 sizes for flat headphone cables)*

2.3.1.5 Unpacking: Optional accessories

19" rack-mount ears *(for the 2U version of the Realiser A16)*

2.4 *Part Names and Functions*

2.4.1 Front panel of the Realiser A16-2U



Front panel of the Realiser A16-2U

The Realiser A16-2U front panel has the following elements (from right to left):

LCD display – a 480 x 320 pixel full colour LCD panel

micro SD card slot and activity light – for external storage of PRIR and HPEQ files, and for firmware upgrades. The activity light is in the left-hand corner of the uSD slot opening.

USB OTG port – digital headphone output (not currently operational)

MIC jacks (L and R) – for in-ear binaural measurement microphones (2 x 3.5mm 4-pole sockets)

GAME port – for a stereo headphone output signal and mono microphone input signal (3.5mm 4-pole socket)

IR receiver window – for receiving remote control commands

Ambient light detector window – detects the intensity of ambient light in the room to reduce or increase the brightness of the LCD display

Indicator LED – indicates power up (steady green), standby (steady red), IR command receive (single green blink) and firmware update mode (continuous blinking green)

VOL knob for User B headphone output – a digital gain control for the headphone B output. This knob also operates as a momentary push switch.

GAIN switch for User B headphone output – can be set to L(ow) for IEMS, M(id) for normal headphones or H(igh) for less sensitive headphones.

HP headphone socket for User B – for a 1/4" stereo headphone plug

HT head-tracker input for User B – connects to the head-top device from User B (2.5mm 4-pole socket)

HT head-tracker input for User A – connects to the head-top device from User A (2.5mm 4-pole socket)

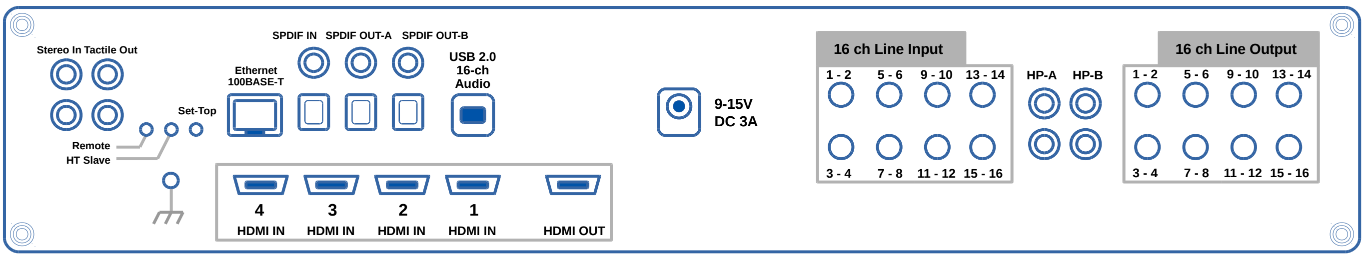
HP headphone socket for User A – for a 1/4" stereo headphone plug

GAIN switch for User A headphone output – can be set to L(ow) for IEMS, M(id) for normal headphones or H(igh) for less sensitive headphones.

VOL knob for User A headphone output – a digital gain control for the headphone A output. This knob also operates as a momentary push switch.

NOTE: The Realiser A16-HS front panel has the same elements listed above, in a vertical and horizontal orientation.

2.4.2 Rear panel of the Realiser A16-2U (standard auxiliary 16ch single-ended)



Rear panel of the Realiser A16-2U

The Realiser A16-2U rear panel has the following ports (from right to left):

Line outputs - Sixteen channels of line-level outputs on 8 x 3.5mm stereo sockets. These should be connected to loudspeaker amplifiers for PRIR measurements or in the AV presentation mode (max output level 1.2Vrms)

HP-B – alternative headphone output for User B on 2 x RCA sockets (identical signal to front panel 1/4" headphone output socket)

HP-A – alternative headphone output for User A on 2 x RCA sockets (identical signal to front panel 1/4" headphone output socket)

Line inputs - Sixteen channels of line-level inputs on 8 x 3.5mm stereo sockets. All of these inputs can be used as sources to create virtual loudspeakers (max input level 1.2Vrms).

Power jack - 9-15V DC, 3A external power supply unit

HDMI inputs - 4 x HDMI 2.0 inputs for digital audio inputs (8-ch LPCM and bitstream)

HDMI output - 1 x HDMI 2.0 output for video pass-thru

USB 2.0 - 16-ch digital audio input, 2-ch digital audio return (24-bit 96kHz max)

SPDIF outputs - optical and coaxial SPDIF outputs for User A and User B stereo headphone signals (variable level)

SPDIF input - optical and coaxial SPDIF input for 2-ch LPCM and bitstream audio signals

Ethernet - 100BASE-T ethernet port for remote control of the A16 via a web-browser interface (not currently available)

Set-top - connects to the set-top device (4-port 3.5mm socket)

HT Slave - connects to the HT Slave port of another A16 allowing one head-tracker to control two A16 units (3.5mm 4 pole socket) (not currently available)

Remote – connects to the Remote port of another A16 allowing one remote control to control two A16 units (3.5mm 4 pole socket)

Earth – for use with the wrist grounding strap (supplied) during PRIR measurements to reduce body induced hum

Tactile Out – line-level low-frequency output signals for connection to seat-shakers or stereo subwoofers (2 x RCA sockets)

Stereo In – analogue line-level stereo audio inputs (2 x RCA sockets)

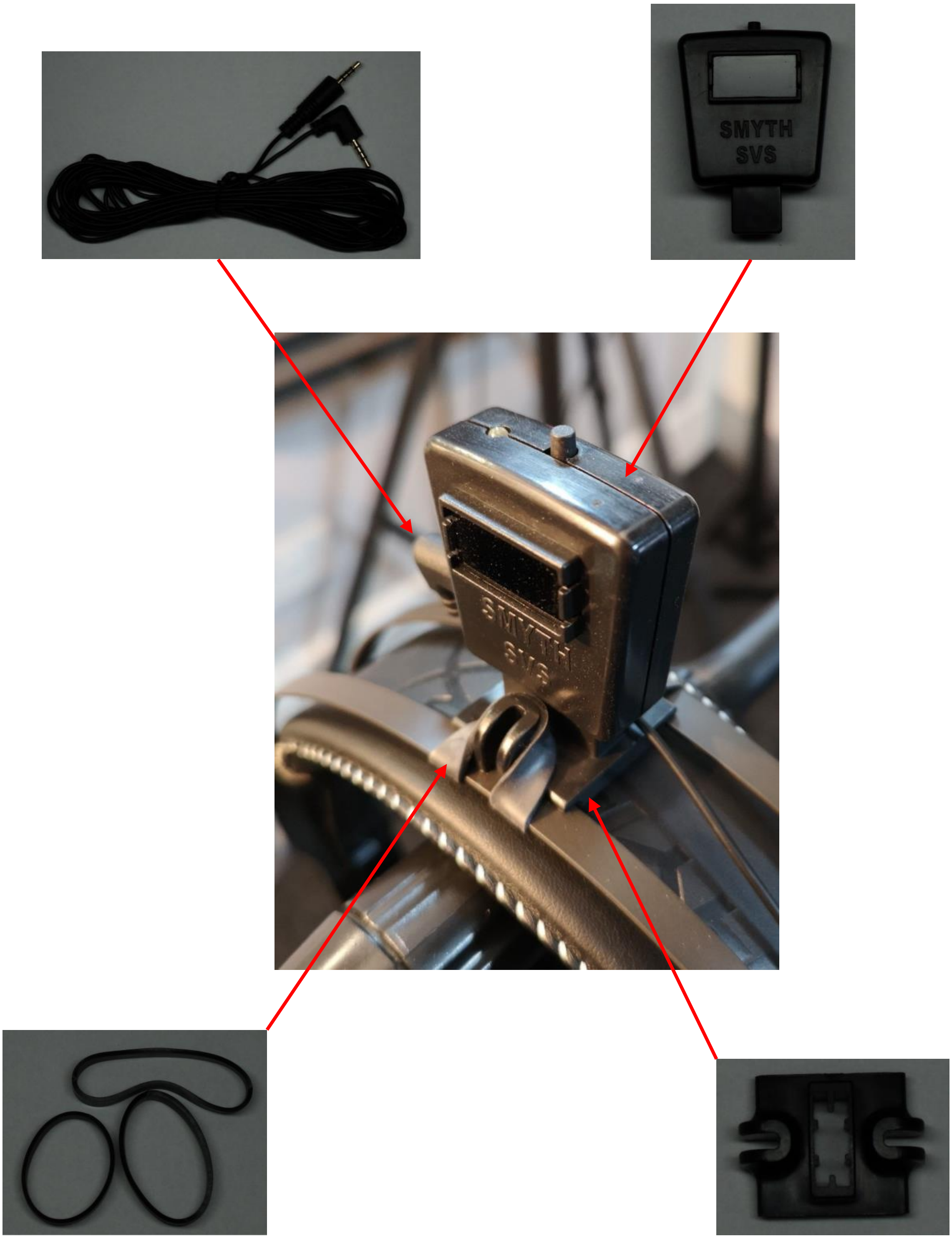
NOTE: The standard Realiser A16-HS rear panel has the same elements listed above, in a vertical and horizontal orientation.

2.4.3 Realiser A16 Remote control

Realiser remote control showing the function of the buttons.

2.4.4 Head-tracker: Head-top and IR Reference Set-top

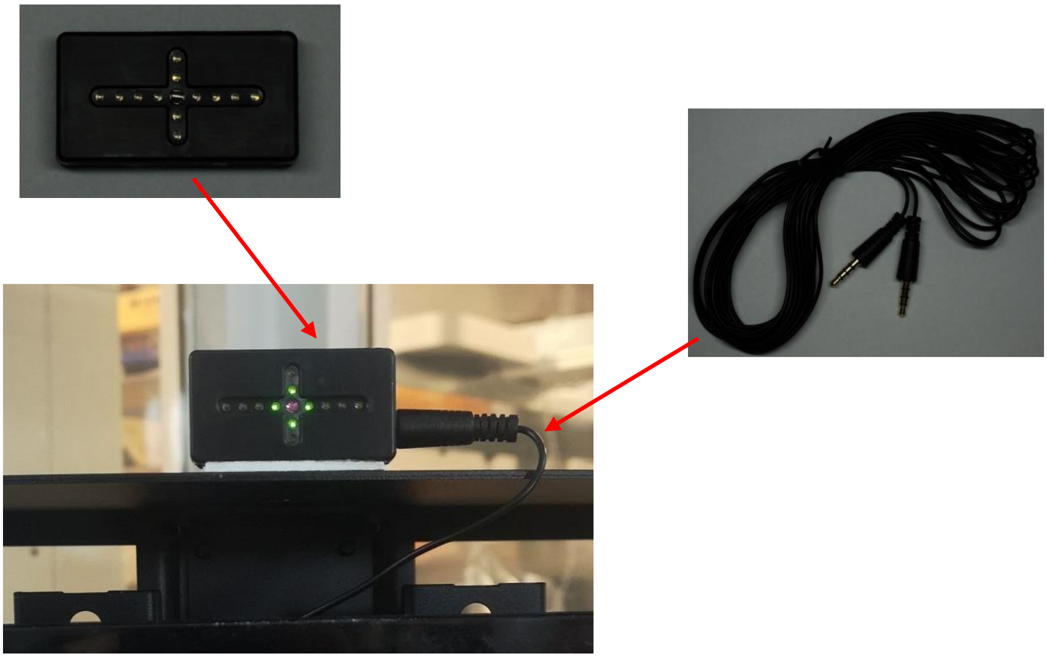
2.4.4.1 Head-Top



The Head-Tracker Head-Top connected to a pair of headphones.

The Headphone and head-tracker head-top connected to the Realiser via the ports in the front panel.

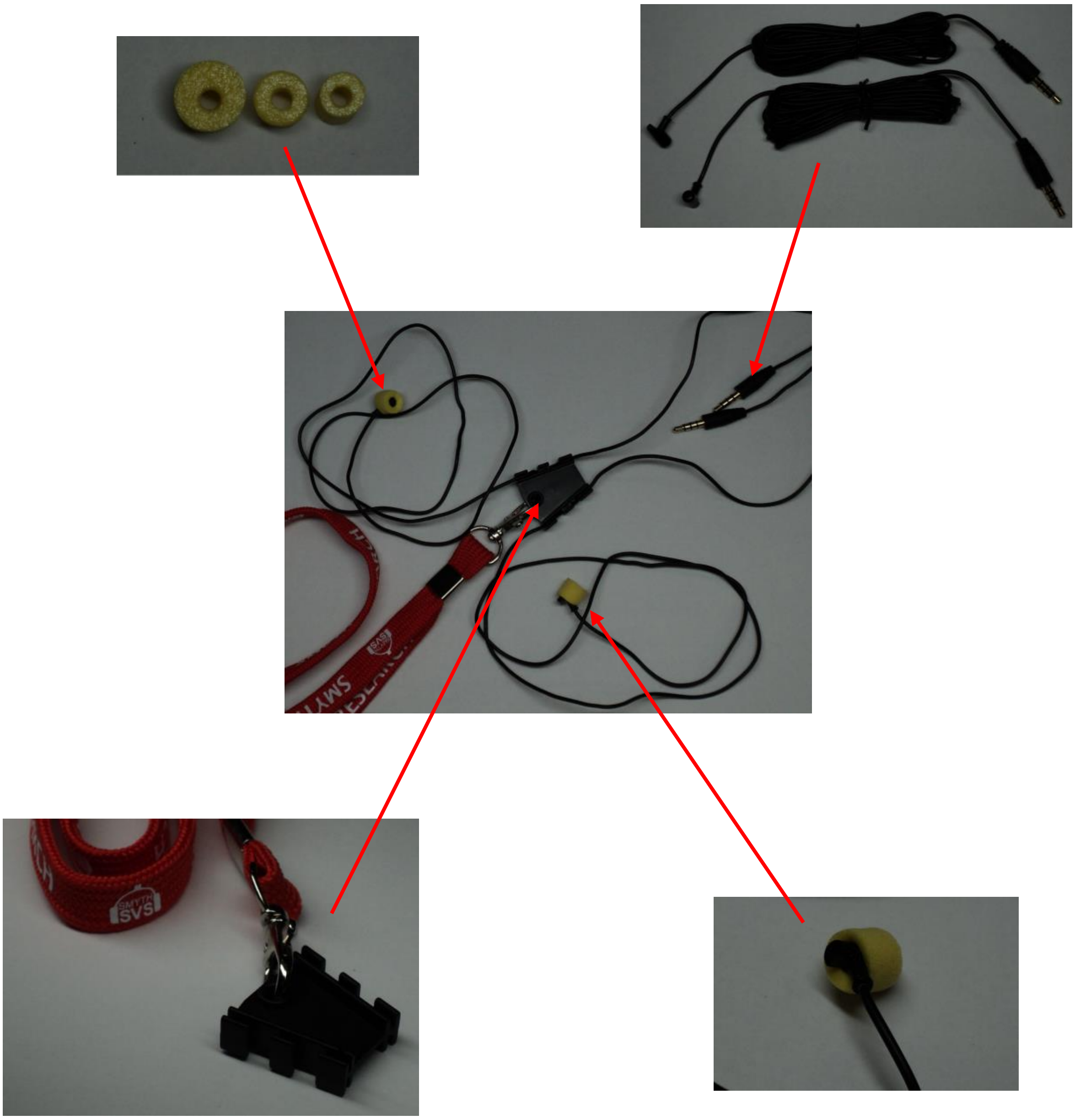
2.4.4.2 IR Reference Set-top



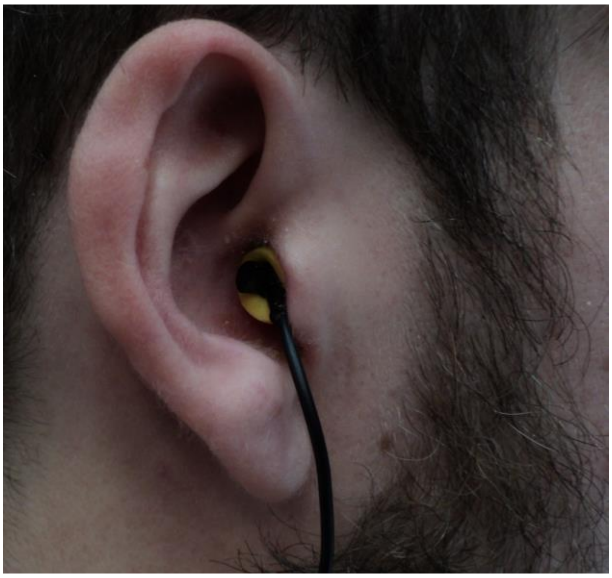
Set-Top powered up

Set-Top connected to the Realiser via the Set-top port on the rear panel.

2.4.4.3 Binaural microphones



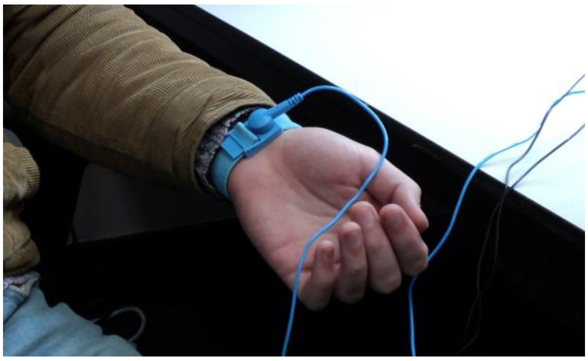
The binaural microphones used to create a PRIR



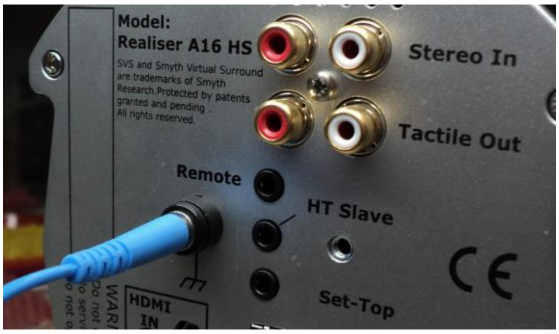
Binaural Microphone in ear

Binaural Microphones in use.

The Microphones connected to the Realiser



The wrist strap worn during a binaural reading.



The wrist strap connected to ground.

2.5 *Quick start*

1. Connect an BD or DVD player (or other HDMI source) to HDMI Input 1 and a TV monitor to the HDMI Out

2. Connect the 9-12V DC power supply to the A16.

3. Power up the A16, using the remote control if necessary.

   The A16 will initially display a splash screen, then show presets loading, and finally display a preset Speaker Map for User A. The factory default preset for both user A and user B will be 9.1.6ch for Atmos, 9.1.2ch for dts:X and 9.1.6ch for PCM audio. The audio source selector should be set to input HDMI-1. If not, navigate to this line and change the input source to HDMI-1.

4. Set the BD or DVD player (or another HDMI source) to output BITSTREAM audio to the A16, and start playing a DVD or BD disc.

   If a Dolby bitstream is detected by the A16, the Speaker Map will display Dolby Atmos as the source, and audio signals will be visible on some of the speakers icons in the Speaker Map – the actual speakers will depend on the format of the Dolby bitstream.

   If the HDMI source is set to AUTO output, and an audio CD is played, a PCM bitstream will be detected by the A16, the Speaker-map will display PCM as the source and will switch to the default 9.1.6ch PCM configured room. Audio signals will be visible on only the left and right speaker icons in the Speaker-map when stereo audio is being received.

   Currently the A16 cannot detect or decode any DTS bitstream.

5. Set the GAIN of headphone A to L(ow) and set the headphone volume of A to 50 using either the VOL knob or the remote control.

6. Connect headphones to the headphone output for User A and check that the volume is not excessive before putting the headphones on.

   The headphones will now be playing an SVS headphone rendered version of the decoded audio signals.

   Details of all the connections on the Realiser are described in an Appendix at the end of this document.

3 Initial power up

3.1 Power On Sequence

During power up the A16 goes through a sequence of hardware and firmware tests that are shown on the LCD display. The sequence for a successful power up is:

1. Splash screen display
2. Hardware verification
3. Loading and activating the User A and User B presets
4. Displaying the Speaker Map page for User A

3.1.1 Splash screen

The splash screen will be changed periodically to indicate major revisions of the firmware.

Rev 2.04 splash screen on Realiser power-up.

3.1.2 Hardware tests

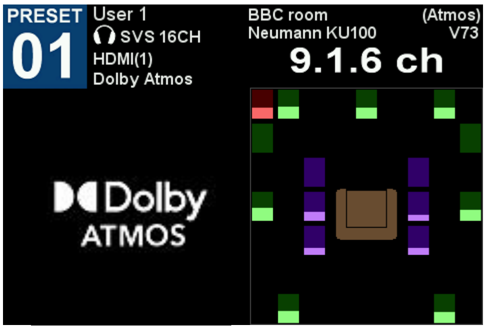
As the splash screen is being displayed the A16 runs internal tests on the DSP memories, CPU memory, FPGAs, HDMI interface board and audio decoding module and verifies the revision of hardware present in the unit.

3.1.3 Loading and running Presets

After the hardware tests have been completed successfully, the A16 will attempt to load and run (or activate) the presets for User A and User B that were used during the previous session. For dual user mode the preset for User A will be loaded and activated first, followed by the preset for User B. For single user mode only the User A preset will be loaded.

3.1.4 Displaying the Speaker Map for User A preset

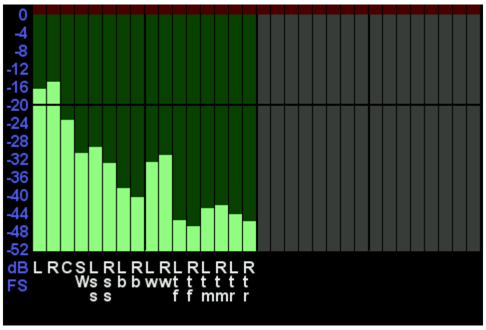
Once the presets have been loaded and activated successfully the A16 will automatically display the speaker map for the active preset for User A. The speaker map is a pseudo-birds-eye-view of the loudspeakers in the virtual sound room associated with the preset. Each speaker doubles up as a miniature level meter that indicates the signal level present at that speaker. In this screen shot User A is currently listening over headphones to a Dolby Atmos soundtrack where the audio tracks are being rendered by the 16ch SVS processor to a 9.1.6ch virtual sound room labelled the BBC room.



Speaker Map for User A

3.1.5 Displaying the Input Level Meters for User A

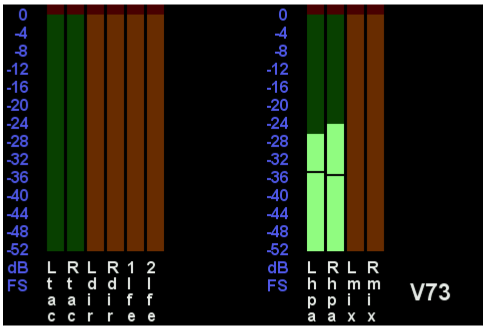
Pressing the green PA key on the remote control displays more typical input level meters showing the digital signal levels that are present at the inputs to the virtual speakers. In this screen capture all sixteen audio signals are active.



Input signal levels for User A

3.1.6 Displaying the Headphone, Tactile and Bass Level Meters for User A

Pressing the green PA key again displays the output level meters showing the digital signal levels that are present on the SVS processor outputs. In this screen shot only the headphone outputs are active.



Output signal levels for User A

Pressing the PA button for the third time causes the display to return to the speaker map. You can also return to the speaker map at any time by pressing the BACK key on the remote control.

3.1.7 Listening to the Internal Audio Test Loop

From the Speaker Map page, it is possible to do a simple headphone sound check using an internally generated musical loop signal.

1) Set the GAIN switch of headphone A output to L(ow) – the GAIN switch is located on the front panel beside the HP A socket.

2) Set the VOL of headphone A to 50 – use the VOL knob for headphone A on the front panel of the A16. The volume can also be controlled from the remote control – the A side and B side volumes are set independently.

3) Connect headphones to the HP A socket to listen to the musical test loop.

4) Toggle ON/OFF the musical loop with the TEST key on the remote control.

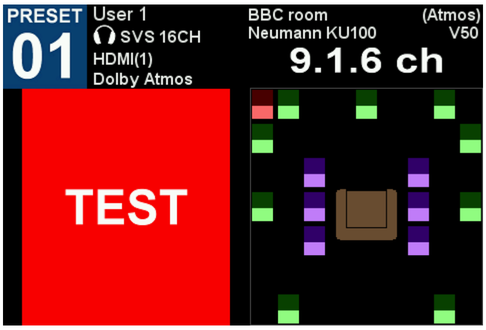
> The word TEST should now be displayed prominently in the Speaker Map display, and all the speaker icons should indicate some signal. Essentially a monophonic music test signal is sent to each virtual speaker and the combined signals from all the virtual speakers are sent to the headphone outputs.
>
> To listen to individual virtual loudspeakers, use the L, C, R, etc keys on the remote control to activate the SOLO mode. (The six overhead speakers are soloed using the keys 1,4,7 and 3,6,9.) A white box will outline the soloed virtual speaker on the Speaker Map display. (These virtual speakers are a factory installed default that were measured using a dummy head binaural microphone). Press the ALL key again to deactivate the SOLO mode and return to listening to all the virtual speakers.
>
> To listen to a group of virtual speakers press the ALL key to activate the MUTE mode. This mutes all the virtual speakers (muted speakers are outlined in red), and the L, C, R etc keys are then used to unmute/mute any group of speakers. Press the ALL key to deactivate the MUTE mode and return to listening to all the virtual speakers.

5) Push the TEST key again to stop the musical loop – the TEST key toggles the musical loop ON/OFF

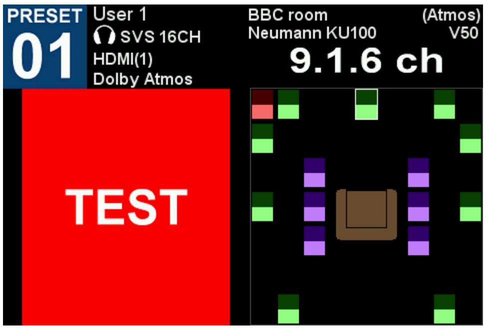
6) Turn all the virtual speakers back on using the ALL key.

This removes the white or red boxes surrounding individual speaker icons from the display. The SOLO and MUTE modes can also be activated during normal listening modes.

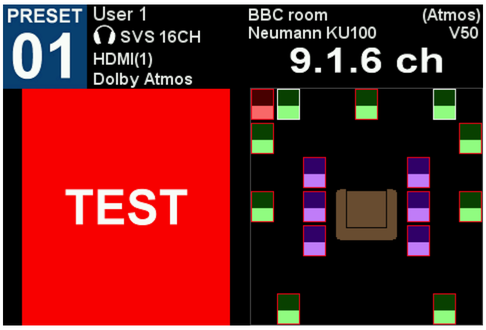
The TEST, MUTE and SOLO modes are useful diagnostic tools for checking the normal operational modes of the Realiser A16. The music loop can be changed to a pink noise signal under the audio settings menu.



Test Mode: using an internally generated musical loop to listen to all the virtual speakers.



Solo Mode: listening to individual virtual speakers. The soloed speaker is outlined in white. The Centre speaker is currently being soloed.

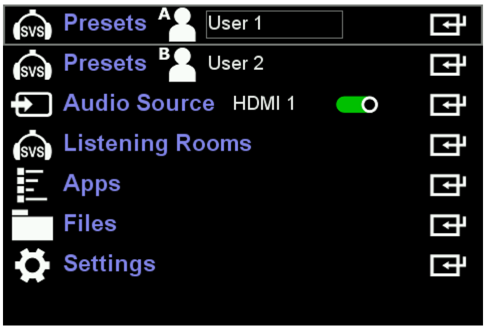


Mute mode: listening to groups of virtual speakers. Muted speakers are outlined in red, unmuted speakers in white. The Left and Right speakers are currently un-muted.

4 Menu Navigation

4.1 The Home Page menu

The Home Page is the root menu for the A16 and gives access to all functions and configuration pages of the A16. The Home Page can be accessed from any other menu using the BACK key repeatedly.

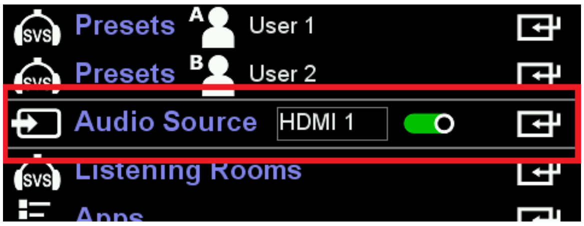


The Realiser A16 home page.

4.2 Navigating the menus, selecting options and changing values with the remote control

4.2.1 Menu option selection:

The outer grey box indicates the menu option currently being selected, Audio Source in this example. Use the ▲ and ▼ keys on the remote control to move the selection box between different menu items.



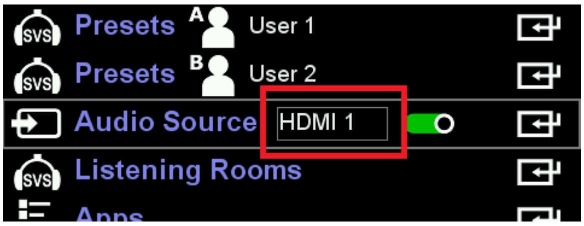
Menu option selector

4.2.2 Selecting values in a menu option:

The inner grey box within the menu option selector indicates that the value may be changed using the ADJ + and ADJ – keys on the remote.

Values may be numerical (a list of numbers), graphical (an on/off toggle switch) or textual (a text list or text entry).

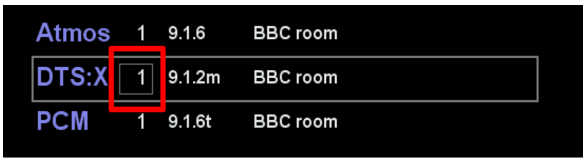
If there are multiple variables on a line then move the selection box left and right using the ◄ and ► keys on the remote control.



Value selector within a menu option

4.2.2.1 Values from a numeric or text list:

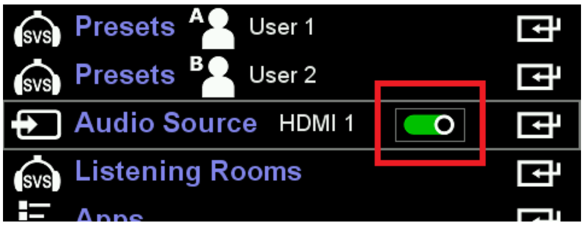
Use the ADJ+ and ADJ- keys to select a numeric or text value from a list.



Selecting a numeric value in a menu option

4.2.2.2 Graphical values:

Use the ADJ+ or ADJ- keys to toggle a switch ON or OFF



Value selector using an on/off toggle button

4.2.2.3 Text entry values:

1. Use the ENTER key to create a text-entry cursor in the value select box.
2. Use the ▲ and ▼ keys to change the cursor from lowercase entry (red cursor) to uppercase (green cursor).
3. To move the cursor use the ◄ and ► keys.
4. To delete a character use the CANCEL key.
5. Input numbers and text using the alpha-numeric keys.
6. To save the new text and exit the text box use the ENTER key.

Text entry - uppercase with a green cursor

4.3 *Moving between menu levels using the ENTER and BACK keys*

4.3.1 The ENTER symbol:

New menu or activate symbol

This symbol at the end of a menu option line indicates that pressing the ENTER command key on the remote control will either give access to another level of menus, or will start some processing, activation or updating function.

Menu continuation symbol for multi-page menus indicating another page below the current one.

4.3.2 Menu continuation symbols: ↓ and ↑

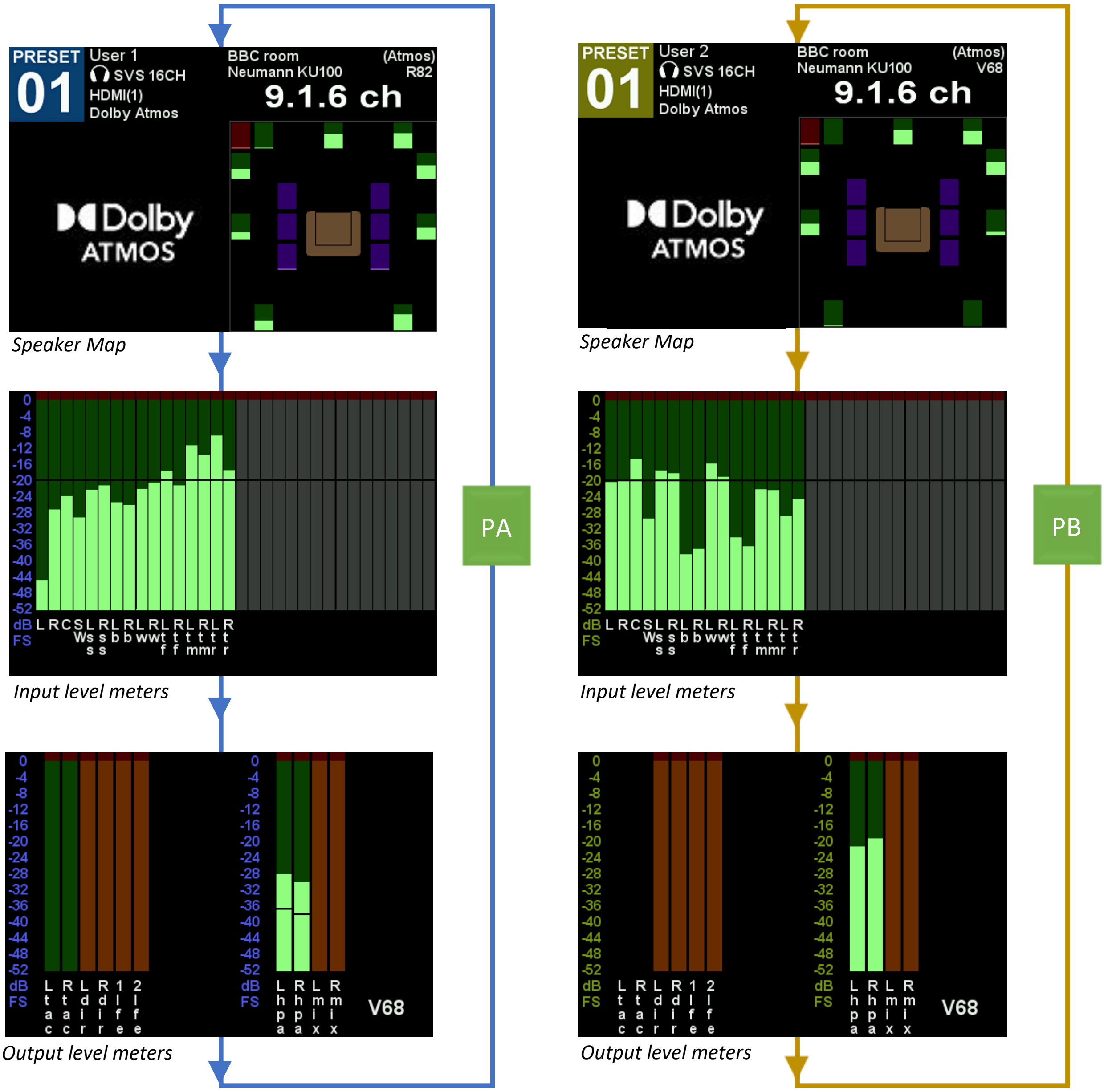
The green ↓ and ↑ symbols, at the bottom or top of a menu page, indicate that the visible screen is only showing part of the full menu. Use the ▲ and ▼ keys to scroll up and down between multiple pages in a menu.

4.3.3 The BACK key:

The BACK key on the remote control is used to simultaneously save configuration data and move back to either the previous menu or to the previous page of a multi-page menu.

4.3.4 The PA and PB key

If the presets for User A and User B have been successfully loaded and activated, the Speaker Map of the active preset for each user can be displayed using the PA or PB keys on the remote control. For User A the preset number shows a blue background while User B has a greenish background. Pressing the PA/PB key for a second time brings up the input level meters, while the output levels meters are shown on the third press. Further presses rotate between these three displays. Whilst in the Speaker Map, Input Level Meter or Output Level Meter pages, either the MENU key or the BACK key will return the user to the previously displayed menu.



4.3.5 Changing Presets from the Speaker Map page:

Whilst in the Speaker Map page the ADJ + and ADJ – keys will cycle through the 16 presets for the currently displayed user.

4.3.6 Displaying head tracking angles using the HT key:

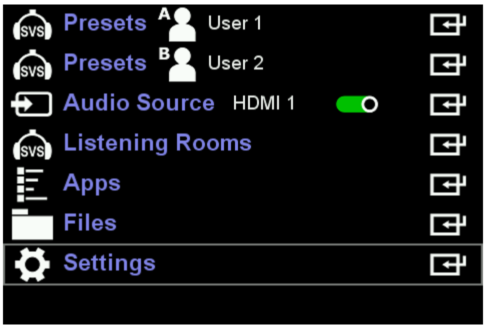
Whilst in the Speaker Map page the HT key toggles between the real time head tracker angle display and the speaker map page.

Real time head tracking angle display.

The azimuth angles for both User A and User B head trackers are shown. The current head tracker firmware revision and the hardware revision is also shown for both. Press HT key to return to the speaker map display.

5 Settings

The Settings menu is accessed from the Home Page menu and is used to set or view configuration data that seldom bb changes.

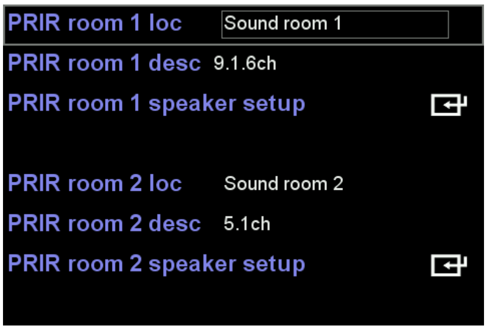


Settings option in the Home Page menu

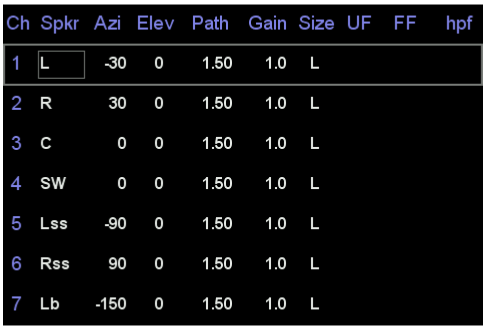
Settings menu

5.1 PRIR Sound Rooms

PRIR sound rooms describe rooms that you intend to capture during a PRIR (ALL) measurement. Two are provided in case there is more than one setup to be captured. If the user does not intend to make PRIR measurements, the factory settings can remain. The information entered is used as a short-cut to populate the PRIR information fields during PRIR measurements. Rather than entering the information manually at the time of the measurement, the routines instead take the information from one of these sound rooms, thereby speeding up the measurement process. It is therefore advisable to describe the room with reasonable accuracy since this information will form the basis of any resulting PRIR files. The specific sound room selected by the measurement routines is selected in the applications page.

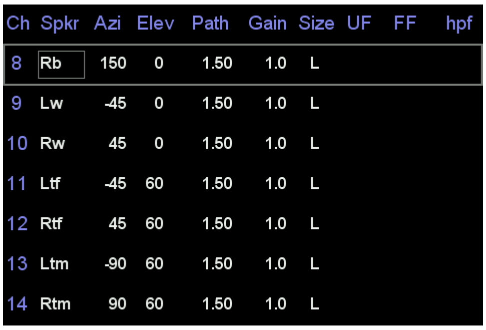


PRIR Sound Rooms menu



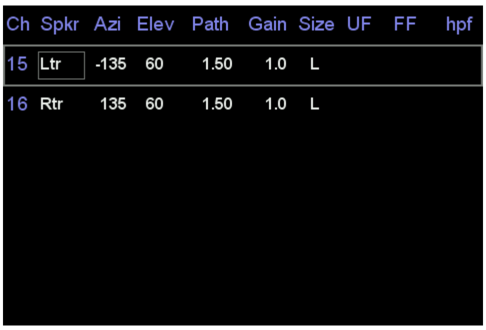
Ch	Spkr	Azi	Elev	Path	Gain	Size	UF	FF	hpf
1	L	-30	0	1.50	1.0	L			
2	R	30	0	1.50	1.0	L			
3	C	0	0	1.50	1.0	L			
4	SW	0	0	1.50	1.0	L			
5	Lss	-90	0	1.50	1.0	L			
6	Rss	90	0	1.50	1.0	L			
7	Lb	-150	0	1.50	1.0	L			

PRIR room 1 speaker setup page 1



Ch	Spkr	Azi	Elev	Path	Gain	Size	UF	FF	hpf
8	Rb	150	0	1.50	1.0	L			
9	Lw	-45	0	1.50	1.0	L			
10	Rw	45	0	1.50	1.0	L			
11	Ltf	-45	60	1.50	1.0	L			
12	Rtf	45	60	1.50	1.0	L			
13	Ltm	-90	60	1.50	1.0	L			
14	Rtm	90	60	1.50	1.0	L			

PRIR room 1 speaker setup page 2



Ch	Spkr	Azi	Elev	Path	Gain	Size	UF	FF	hpf
15	Ltr	-135	60	1.50	1.0	L			
16	Rtr	135	60	1.50	1.0	L			

PRIR room 1 speaker setup page 3

5.1.1 PRIR room 1 loc

Room location can be a basic description of where the room is. This name text will become part of the PRIR file.

5.1.2 PRIR room 1 desc

Room description is often the speaker layout and becomes part of the PRIR file.

5.1.3 PRIR room 1 speaker setup

Speaker setup includes assigning speaker labels (Spkr) to the analogue outputs (Ch) at the back of the A16, the speaker azimuth (Azi) and the elevation (Elev) angles, and the distance in metres from the speaker to the listener (Path). All other input fields are ignored at this time. This menu is spread over 3 pages.

5.1.3.1 Ch

Channels 1 to 16 refer to the physical multichannel line outputs on the back panel of the A16. These physical outputs must be connected to the amplifiers of the matching loudspeakers in the room and cannot be re-configured.

5.1.3.2 Spkr

The ADJ+ and ADJ- keys are used to cycle through all the available speaker labels for each output channel. The actual names of these speakers, and their approximate physical location in a room, is given in Table 4 in Appendix B.

The speaker label chosen for each output channel should match the actual physical loudspeaker in the room being measured. For example, if the room is configured for Dolby Atmos then the Dolby Atmos naming conventions should be used.

Tables 1 and 2 in **Appendix A: Listening Rooms Loudspeaker Configurations** list the names of the loudspeakers used for Dolby Atmos and DTS:X layouts up to 16 channels. However, any names may be used for other formats, such as Ambisonics.

> ***NOTE:*** *Care should be taken when choosing speaker labels for a PRIR Sound Room. The use of some speaker names may restrict the ability of the measured virtual speakers to become associated with a particular format – in effect the virtual speaker in a PRIR file may become locked out of a particular format due to its name. These restrictions will only apply for bitstream and PCM audio that is input through HDMI, and are summarised in Appendix B: Table 4: Loudspeaker names and labels.*
>
> *For example, if a virtual speaker is labelled as Lh (Left height) it can be seen from Appendix B: Table 4 that this name is not used in any Dolby Atmos configuration that can be rendered by the Realiser A16. Therefore, a virtual speaker labelled as Lh in any PRIR file will not be matched to a decoded audio output channel when listening to any Dolby Atmos encoded bitstream.*
>
> *Alternatively, if a virtual speaker is labelled Ltf (Left top front) this name is used by Dolby Atmos, DTS:X and PCM, and therefore can be matched in at least one configuration of either a Dolby Atmos, DTS:X or PCM Listening Room.*
>
> *Notwithstanding, the composite PRIR builder application can override virtual speaker labelling if the user realises the problem after the measurement has been made.*

5.1.3.3 Azi

This describes the azimuth angle of the loudspeaker with respect to the PRIR measurement position – normally the central listening position. Negative angles are used for left side loudspeakers, positive values for right side speakers.

5.1.3.4 Elev

This is the elevation angle of the loudspeaker, with respect to ear height, from the listening position. Positive angles are used for speakers above ear level, negative for below ear level.

5.1.3.5 Path

Path describes the distance (in metres) of each speaker from the listening position.

5.1.3.6 Gain

Gain is not presently used and can remain at the factory setting.

5.1.3.7 Size

Size is not presently used and can remain at the factory setting.

5.1.3.8 UF and FF

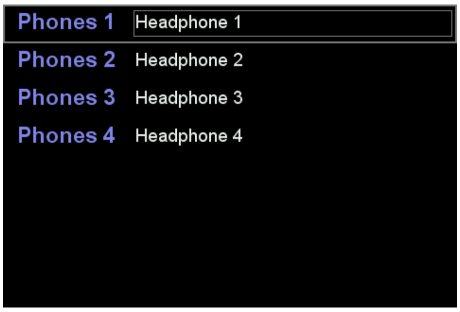
UF and FF are not presently used and can remain at the factory setting.

5.1.3.9 HPF

HPF is not presently used and can remain at the factory setting.

5.2 Headphones

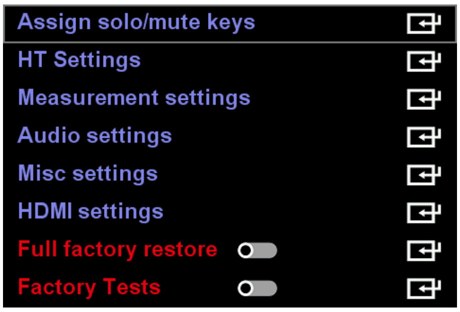
Phones 1-4 describe headphones that you intend to equalize during a HPEQ measurement. Four are provided in case the user wishes to measure more than one model on a regular basis. This menu allows the user to set descriptors for the four different headphones using the alpha-numeric keyboard to insert text. These headphone descriptors are added to any HPEQ file measured for these headphones, allowing particular HPEQ files to be readily identified. Any unused descriptors can retain their factory name.



Headphone menu: the names of four headphones can be stored.

5.3 System

System settings are parameters that may need to be changed for different listening arrangements and conditions.

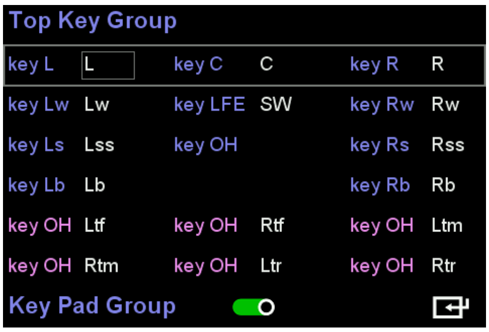


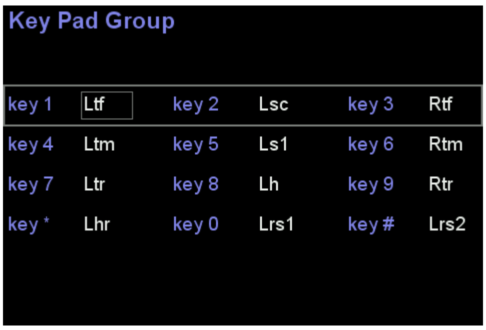
System Menu

5.3.1 Assign solo buttons

During test procedures individual speakers, both real and virtual, can be soloed using several keys on the remote control. This menu allows the user to change the assignment using the ADJ+ and ADJ- keys on the remote.

CAUTION: *Any of the solo keys can be re-assigned. The example below illustrates the key labelled Ls on the remote control has been re-assigned to solo a Lss speaker. Therefore, after re-assignment, the actual label names on the remote-control keys may not reflect the speaker names that they control.*

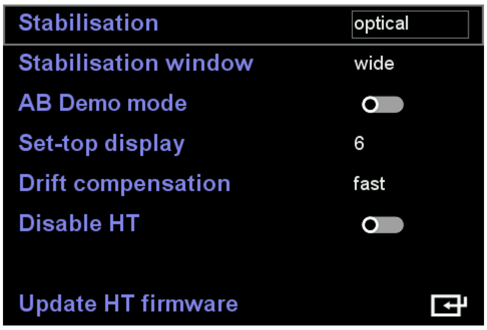




Assign solo buttons menu. In this example the Ls/Rs keys have been configured to control the Lss/Rss speakers.

5.3.2 HT Settings

The Head-tracker consists of a head-top device (placed on the headphones) and a set-top device that normally sits on top of a monitor or TV, centrally. The head-top device has three tracking elements: inertial (gyro), magnetic and optical. The inertial element is the primary means of tracking the rotation of the listener's head through a full 360 degrees, with either the magnetic or optical elements being used to stabilise the inertial tracking – i.e. to correct for inertial drift. The inertial tracker can also be used with no stabilisation. Either the OPTICAL or OPTICAL A8 stabilisation requires a reference pulse of IR light from the set-top device, whilst NONE or MAGNETIC stabilisation does not specifically require the use of the set-top device.



HT Settings menu

5.3.2.1 Stabilisation

Stabilisation refers to the method of correcting for long-term drift of the inertial tracking element in the head-tracker.

If NONE is selected then the inertial tracking built in to the head tracker determines the head angle at all times - there is no reference to the magnetic or optical elements. The inertial heading must first be set to the listening frame i.e. the zero degrees angle must be set using the push-button switch on top of the head-tracker when looking at zero degrees. Fixed heading leakage is used to overcome the natural inertial drift, pulling the heading slowly to zero degrees and can be set to either FAST or SLOW. Essentially this means that if the headphones are held static in any direction, this static direction will eventually become zero degrees – i.e. the virtual centre speaker will eventually move to this direction.

If OPTICAL is selected then the position of the set-top IR reference determines zero degrees. The inertial tracking is stabilised by the optical heading, that is, the optical heading is used to eliminate drift in the inertial tracking. The optical sensor has already been calibrated at the factory. Drift elimination is limited to when the optical sensor is operating within the stabilisation window. A wide window is +/- 60 degrees either side of the set-top reference. A narrow window is +/-30 degrees either side. Outside this window the inertial drift elimination process is suspended.

If MAGNETIC is selected then the inertial tracking is stabilised by the magnetic heading. That is, the magnetic heading is used to eliminate drift in the inertial tracking. The magnetic heading must first be set to the listening frame i.e. the zero degrees angle must be set using the push-button switch on top of the head-tracker when looking at zero degrees. Drift elimination is limited to when the magnetic sensor is operating within the stabilisation window. A wide window is +/- 20 degrees either side of the current zero degree direction. A narrow window is +/-10 degrees either side. Outside this window the inertial drift elimination process is suspended. The magnetic sensor should be calibrated if magnetic stabilisation is to be used – see Appendix C.

If OPTICAL A8 is selected then the position of the set-top IR reference determines zero degrees. However, in this mode the head tracking calculation is based entirely on the optical sensor (as was the case for legacy A8 operation). For reliable tracking the head tracker must point towards the set-top IR reference and there should be a clear path between the two. For the optical A8 mode the maximum head angle is fixed 65 degrees left or right of the set-top. Outside this, the heading will freeze at the last valid angle until the head tracker moves back into range.

***NOTE:** The optical tracking sensor can be fooled by IR light from regular incandescent light sources and sunlight, causing the head tracker to pull its zero degrees reference towards these extraneous sources. Please ensure such lighting is keep well away from the set-top to avoid this confusion. If possible, operate the system in subdued lighting for best results.*

5.3.2.2 Stabilisation window

This sets the window size to either WIDE or NARROW within which OPTICAL and MAGNETIC stabilisation operates. Outside the stabilisation window the inertial sensor drift compensation is suspended. Within the stabilisation window the angle measured by the magnetic or optical sensors is used to eliminate any drift in the inertial sensor.

5.3.2.3 AB Demo mode

Normally set OFF. When set ON the tilt angle of the head-top can be used to trigger the A16 to switch between headphone and speaker outputs (AV mode). This is useful for comparing virtual and real loudspeakers immediately after a PRIR has been measured, but during normal playback this can be distracting.

5.3.2.4 Set-top display

This sets the light intensity of the green LEDs on the set-top device. It can also be set OFF. This does not affect the intensity of the IR source in the set-top used for optical stabilisation.

5.3.2.5 Drift compensation

This lets the current angle, measured by the inertial sensor, leak away exponentially to zero either in SLOW or FAST mode. This exponential leakage is only applied for the NONE stabilisation mode.

5.3.2.6 Disable HT

This forces the head tracker angles for user A and user B to be held at zero. This would typically be used where the user wishes to dispense with head tracking.

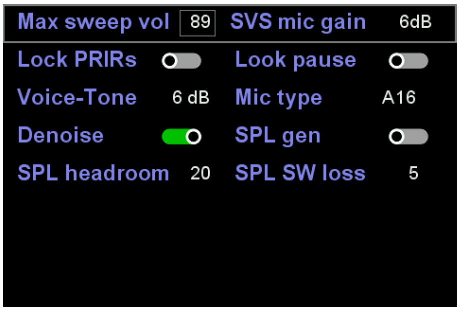
5.3.2.7 Update HT firmware

Used to update the head-tracking firmware in the head-top **- the head-top must be connected to the HT port of User A.**

New head-tracking firmware may from time-to-time be included within a new A16 firmware updates. Please consult the firmware update documentation. However, it is not possible to update the head tracker firmware automatically and requires the user to initiate the process following an A16 firmware update. Details on updating the HT firmware are given in Appendix F. The current head-tracking firmware version can be viewed in the Updates/About menu.

5.3.3 Measurement Settings

This option sets parameters that may need to be changed during a PRIR or HPEQ measurement.



Measurement settings menu

5.3.3.1 Max sweep Vol

Sets the maximum volume of the sine wave sweeps output from the Multichannel Line Outputs during the PRIR measurement. This will over-ride the value set in the Max Vol line option of the System menu (**Home Page** menu: **Settings** menu: **System** menu). In other words, during a PRIR measurement the maximum volume of the sine sweeps can be set to a higher (or lower) value than the maximum line out volume set for normal audio playback.

5.3.3.2 SVS Mic gain

Sets the gain of the binaural microphone during the PRIR and HPEQ measurements. For example, the gain may need to be increased to boost the microphone signals if the sine sweeps from the loudspeakers are too low in volume. A lower mic gain will normally increase the signal-to-noise ratio of the recorded sine sweeps, and is generally preferred.

5.3.3.3 Lock PRIRs

A locked PRIR can only be used by the A16 that was originally used to measure it.

> *Normally set OFF.*
> *When set ON a measured PRIR is locked to the host A16.*
> *When set OFF a measured PRIR can be used by all A16 units.*

5.3.3.4 Auto save

Normally set OFF. When set ON a measured PRIR is automatically saved to the SD-card

5.3.3.5 Voice-Tone rel gain

This changes the relative loudness of the voice prompts compared to the sine sweeps during PRIR measurements. Because the voice prompts are emitted from all active speakers during a PRIR measurement whilst the sine sweeps are emitted from either one or four loudspeakers, in certain situations it is useful to be able to reduce or increase the loudness of the voice prompts. This control does not affect the loudness of the sine sweeps – only the loudness of the voice prompts.

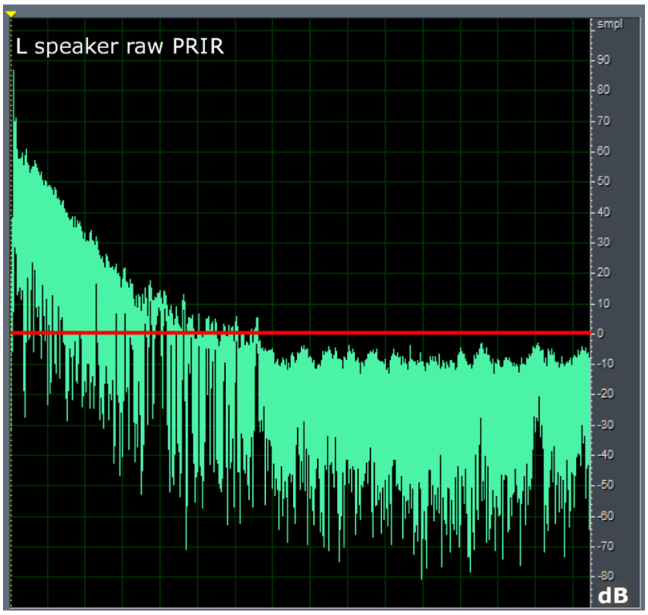
5.3.3.6 Mic Type

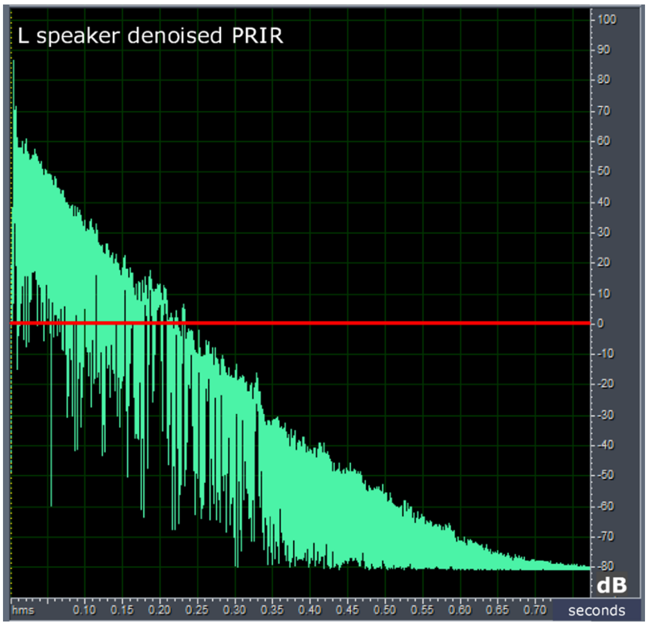
Normally set to A16. The A16 and A8 binaural microphones have slightly different frequency response characteristics, and this menu option partially compensates for this difference. It does not compensate for the electrical differences between the microphones.

Caution: *The A8 microphones CANNOT be used directly with the A16 and will be damaged if this is attempted. Please contact the company if you wish to connect A8 microphones to the A16.*

5.3.3.7 Denoise

Denoise is a signal processing routine that is useful for reducing the effects of background noise in impulse response measurements. Denoise defaults to on. The noise floor of a PRIR measurement is a function of the ambient noise level of the sound room being measured, the noise level of the in-ear microphones, the energy of the sinewave sweeps and the sweep duration. In studio conditions the 12-second sweep measurements can attain signal-to-noise ratios of around 112dB (a 4-second sweep will on average be 5dB lower at 107dB). Noise in a PRIR is not perceived as noise in the conventional sense but as low-level reverberation. As such, it would be more accurate to describe the quality of the PRIR as having a peak-to-residual reverberation ratio of 112dB. This means that in a virtual room the reverberation will at best decay to 112dB below peak listening level, whereas in a real room the reverberation decays to zero. The left-most trace of the diagram below illustrates this residual reverberation phenomenon which shows the raw PRIR reverberation energy as a function of time over the Realiser's 0.75 second convolution window. The measured room has a reverberation time of around 0.25 seconds and the final 0.4 of the PRIR is just residual noise, albeit 100dB below peak. Fortunately, low level reverberation of this order is difficult to perceive under normal listening conditions and therefore a good measurement made under ideal conditions will essentially be indistinguishable from the real room. Nonetheless, ideal conditions are often not attained in practice and the use of 12 second sweeps can make measurement sessions lengthy occasions. The denoising feature results in a virtual room reverberation decay that more accurately follows that of the room under measurement, as depicted in right-most trace in the diagram below.





5.3.3.8 SPL gen

SPL gen enables or disables the internal noise generator used to measure sound pressure levels (SPL). This setting is only relevant during the SPL measurement procedure.

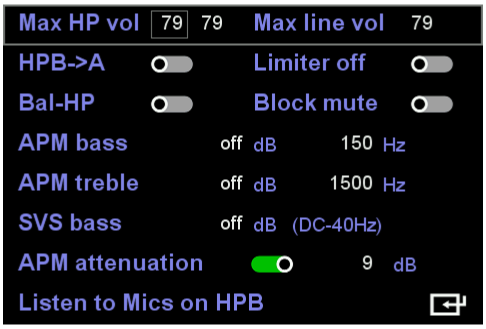
5.3.3.9 SPL headroom

SPL headroom sets the desired headroom above the SPL reference level. The default is 20dBu. The SPL headroom value is used to evaluate if there is sufficient gain in the headphone circuit to guarantee the headroom requirement. This setting is only relevant during the SPL measurement procedure.

5.3.3.10 SPL SW loss

SPL SW loss sets the typical reduction in peak loudness that occurs when a full-band noise signal is reproduced through a 80Hz bandlimited subwoofer. The default is 5dB. This setting is only relevant during the SPL measurement procedure.

5.3.4 Audio settings



Audio settings menu

5.3.4.1 Max HP Vol

These set the maximum headphone output volume for User A and User B.

5.3.4.2 Max Vol line

Sets the maximum Multichannel Line Output volume for audio playback during AV playback mode or during A/B switching mode. This does not affect the volume of the sine-wave sweeps output during a synchronous PRIR measurement which are set by the Max Sweep Vol set in the Measurement Settings menu

5.3.4.3 HPB->A

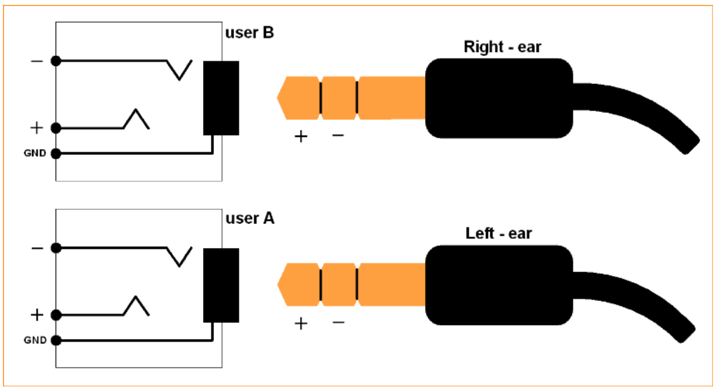
This option, when enabled, allows the headphone output for User B to be switched almost instantaneously to the headphone output of User A. This can be very useful when comparing small variations in either PRIR measurements or HPEQ adjustments. This mode is only available when in the User A Speaker Map display. Further details on this feature are given elsewhere.

5.3.4.4 Limiter off

The digital headphone volume is automatically reduced in real time each time a digital clipping event is detected at the PCM headphone output. This limiter acts to ensure that the listener is not subjected to sustained clipping distortion due to the presence of high-level input signals. The automatic limiting function can be disabled by enabling Limiter off.

5.3.4.5 Bal-HP

Enabling Bal-HP switches the user A headphone output to balanced mode. In this case the user A headphone jack (or the user A rear RCA jacks) carries the balanced signals for the left ear and the user B headphone jack (or the user B rear RCA jacks) carries the balanced signals for the right ear (see schematic below). Only headphones that are wired with a separate return wire to each driver can make use if this mode. Typically, a custom balanced headphone cable will be required. Since the user B headphone jack operates as the R-ch for user A headphone, user B audio can only be heard through the user B SPDIF output. In balanced output mode the voltage across the headphone drivers is twice that of the unbalanced mode and, as a result, will sound twice as loud for the same volume setting. Please ensure your headphone volume is reduced appropriately before engaging this mode. Please also ensure that the user A and user B gain sliders on the front panel are set to the same position. Moreover, we also recommend using the lowest slider gain settings for both until you become more familiar with this mode of operation. Note that the 'Bal-HP' switch is implemented in real time and does not require the reloading of Presets.



5.3.4.6 Block mute

Block mute is a legacy function that bypasses the muting function of the APM89L decoder board. Due to an error in the A16 muting logic, when changing tracks (BluRay, DVD, CD) a sharp click can sometimes be heard in the headphones/line output just as a track is interrupted and just before a track begins to play. This problem affects all A16 manufactured prior to May 2020. It does not affect APM110 based A16s. Please be warned that although Block Mute is effective at avoiding track-skip clicks, it could theoretically allow other audible artefacts to pass through to the headphones and/or loudspeakers. Please exercise caution by maintaining a low volume setting until you gain experience as to what artefacts, if any, occur during your normal operations.

5.3.4.7 APM bass

APM bass is a Bass shelving filter that can be deployed when listening to HDMI and SPDIF sourced audio, both over the headphones and via the line outputs in AV mode. The Bass gain can be adjusted +/-12dB while the shelf frequency can be adjusted from 50 to 500Hz. Adjustments to the filter occur in real time and affects all audio channels for both User A and User B equally.

5.3.4.8 APM treble

APM treble is a Treble shelving filter that can be deployed when listening to HDMI and SPDIF sourced audio, both over the headphones and via the line outputs in AV mode. The Treble gain can be adjusted +/-12dB while the shelf frequency can be adjusted from 500 to 5000Hz. Adjustments to the filter occur in real time and affects all audio channels for both User A and User B equally.

5.3.4.9 SVS bass

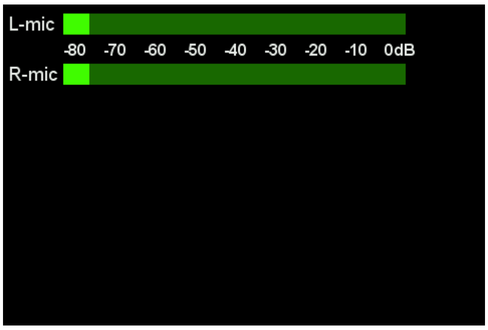
SVS bass is a sub-sonic Bass shelving filter that can be deployed when listening to any audio source over the headphones. The Bass gain can be adjusted +/-12dB while the shelf frequency is fixed 0 to 40Hz. Adjustments to the filter occur in real time and affects all audio channels for both User A and User B equally.

5.3.4.10 APM attenuation

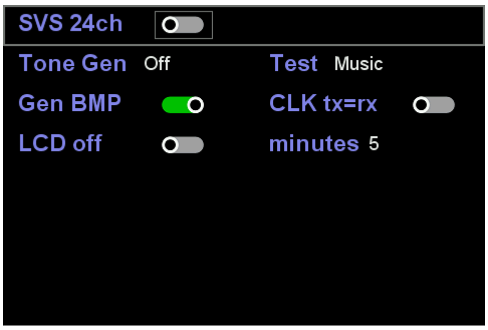
APM attenuation is the reduction in the output signal level of the APM89L/110 decoder. The default attenuation is 9dB and this should not be altered unless the user has a specific application that requires a different level. The decoder output level must be attenuated for normal operation in order to provide sufficient headroom for any additional bass managed LFE signals that occur within the decoder. When listening to decoded audio over the headphones, the APM attenuation is reversed within the SVS processors, thereby making the level shift invisible to the user. However, where the decoded audio is heard through the line outputs using AV mode, this audio level remains attenuated by the APM attenuation setting.

5.3.4.11 Listen to Mics on HPB

Listen to Mics on HPB allows the user to verify the operation of the measurement microphones using the L-mic R-mic level meters and to listen to the microphones over the user B headphone output. This step is often undertaken just prior to beginning a PRIR measurement in order to confirm that the L-mic is in the left ear and the R-mic is in the right ear.

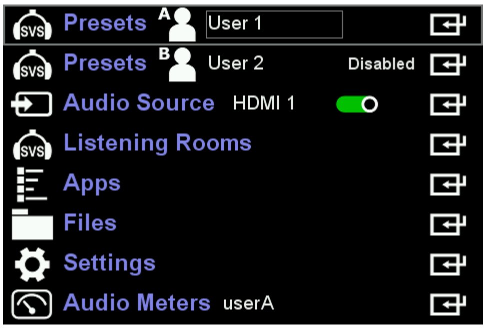


5.3.5 Misc Settings

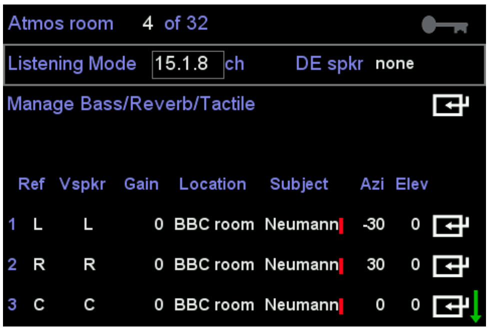


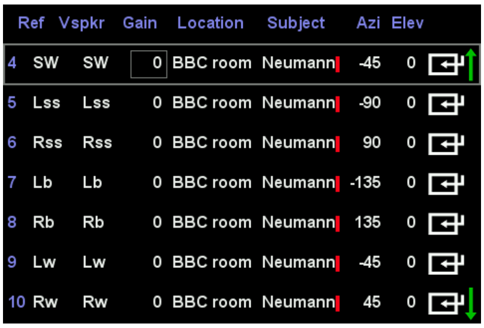
5.3.5.1 SVS 24ch

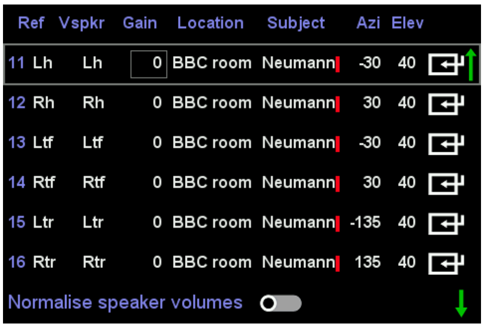
For A16 models that incorporate the APM110 decoder sub-assembly a single user 24 channel Atmos decoding, Dolby Surround up-mixing and SVS rendering mode is supported. Please note that PCM decoding over HDMI remains limited to 7.1 speakers, but can, if desired, be up mixed to a higher channel format using the Dolby Surround up mixer included in the APM110. Also note that the AV mode is disabled while the A16 is running in 24-speaker mode due to the fact that the A16 only supports 16 line outputs. In dual user mode the A16 runs separate 16 speaker SVS rendering operations for users A and B using a DSP for each. 24ch SVS rendering requires the use of both DSPs for user A, with user B headphone output disabled. Hence 24-speaker decoding and rendering drops the functionality of the A16 down to a single user. Furthermore, due to a limitation in the A16 motherboard design, 24 speaker Atmos decoding can only input HDMI sourced audio streams (HDMI(1), HDMI(2), HDMI(3) or HDMI(4). Specifically, it is not possible to decode SPDIF sourced bitstreams while the 24ch Atmos decoding operation is running. To enable the 24-speaker Atmos decode and SVS rendering mode, first ensure the audio source is set to an HDMI input, then enable the slider on SVS 24ch. The A16 will immediately implement the change, jump back to the home page, and reload the User A Preset.

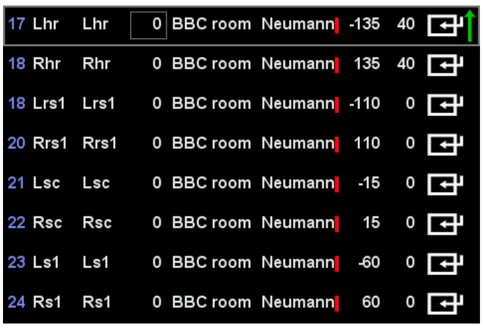


Once completed a ‘disabled’ notice is added to the User B Preset line to remind the user that the user B side is no longer available. To make use of the 24ch mode it is necessary to select an Atmos listening room with more than 16 speakers. Building Atmos rooms beyond 16 speakers is undertaken in the normal way except (see chapter on building listening rooms) that a third page for the additional 8 speakers is included at the end. In the example below a 15.1.8ch Atmos room has been selected in the Listening Mode. The virtual speaker list has then been populated from the BBC factory room.

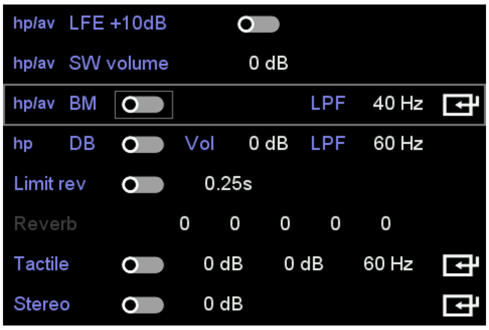




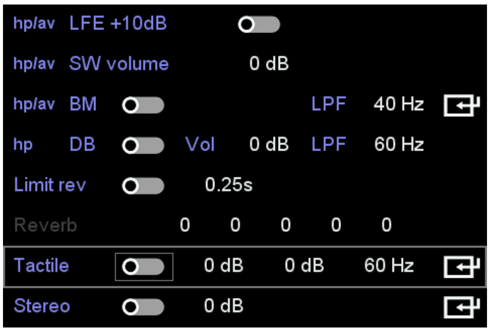


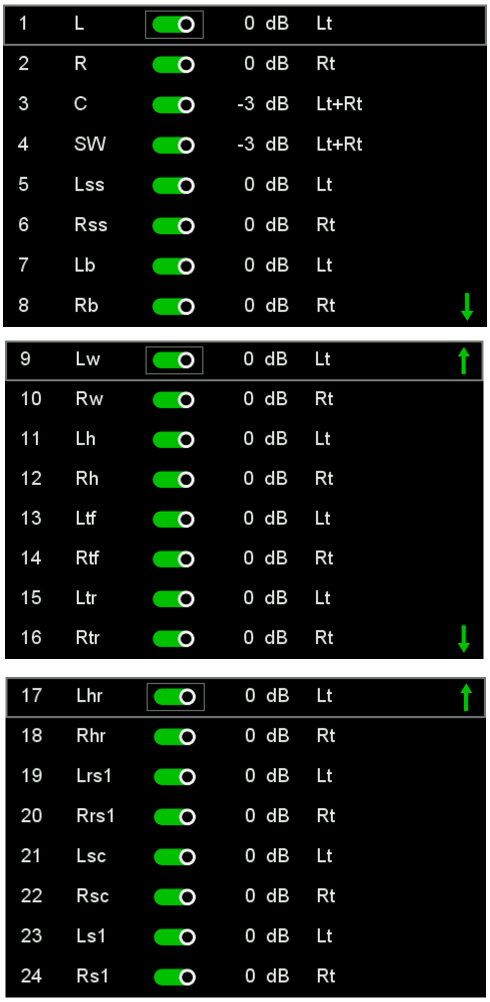


Bass management for 24-ch rooms also includes an additional 4 pairs of speakers.

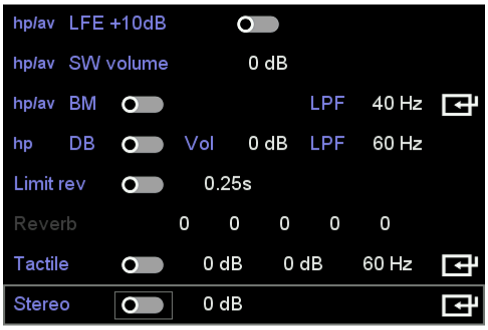


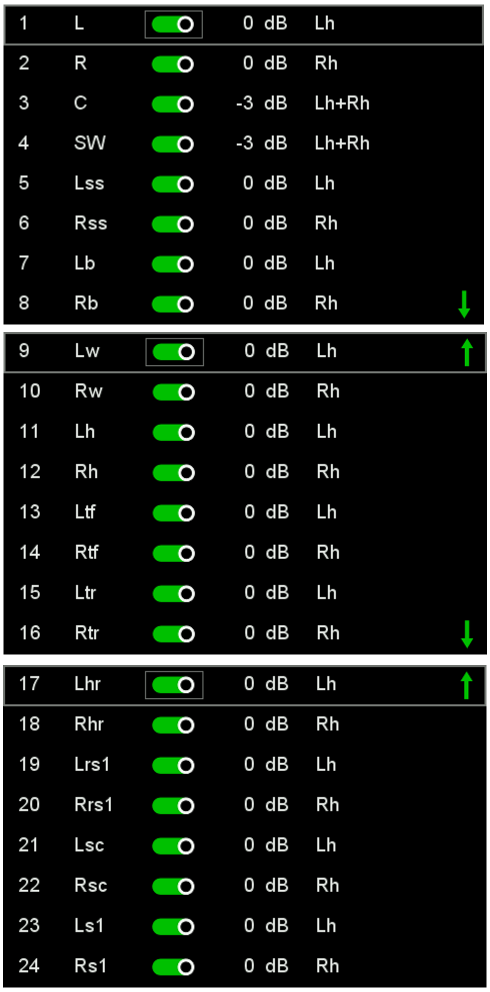
Tactile configuration for 24-ch rooms also include 8 additional speakers.



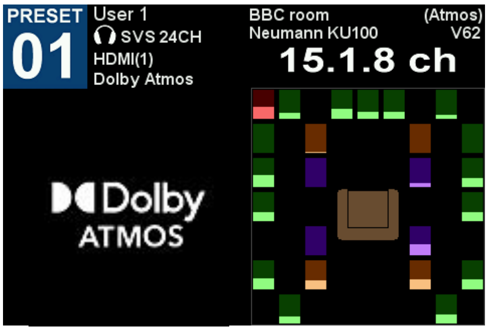


Also for Stereo Mix Down configuration.

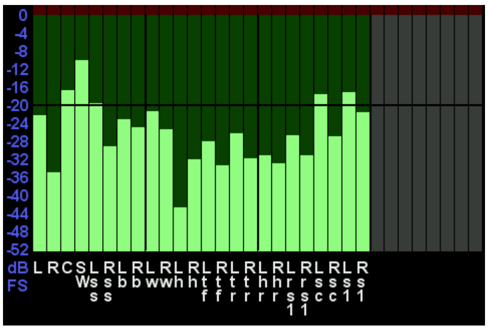




The speaker map for 24 ch mode.



The 24 ch input level meter display.



5.3.5.2 Tone Gen

To aid decoder speaker identification the Tone Gen feature of the APM89L/APM110 decoder subassembly can substitute decoder audio with known tones. This is particularly useful for verifying high order decoder modes of operation. Presently the tone generator will only function when an HDMI input is receiving active content and will mute anytime this HDMI stream is interrupted. Hence it is necessary to select HDMI(1-4) before enabling this feature and then play any content with a 48kHz soundtrack to this same HDMI input to active the tones.

The tone generator function has two modes of operation.

1) ID Tones are tones with a unique frequency for each listening room speaker and are output at a level of -20dBFS to both SVS headphone rendering DSPs when using headphones or to the line outputs when in AV mode.
2) 1kHz Tones are the same 1kHz sinewave tone for each listening room speaker and are output at a level of -20dBFS to both SVS headphone rendering DSPs when using headphones or to the line outputs when in AV mode.

Note that the tone generator function runs only on the APM89L/APM110 decoder boards and therefore cannot be used to verify listening rooms that input their audio via the USB or Line inputs. Also note that for the Tone Gen function to work it is necessary to reload the user A Preset after the function is selected in the MISC settings page. Likewise, to turn off the tone generator it is necessary to return to the MISC settings page, turn Tone Gen off and then reload user A Preset.

5.3.5.3 Test

The TEST key is used to activate a built-in audio loop to aid verification and debugging. The user can select between a short music loop and a pink noise loop (-20dBFS). Note that this selector is implemented in real time and does not require Presets to be reloaded. However, there is a 10 second pause on changing the test selection as the new test signal is uploaded to the rendering DSPs.

Note that the tone generator function runs only on the APM89L/APM110 decoder boards and therefore cannot be used to verify listening rooms that input their audio via the USB or Line inputs. Also note that for the Tone Gen function to work it is necessary to reload the user A Preset after the function is selected in the MISC settings page. Likewise, to turn off the tone generator it is necessary to return to the MISC settings page, turn Tone Gen off and then reload user A Preset

5.3.5.4 Gen BMP

When Gen BMP is enabled, pressing the # key on the remote captures the current LCD image and writes it as a 480x320 pixel Bitmap file under the ‘realiser’ directory on the external SD card.

5.3.5.5 CLK tx=rx

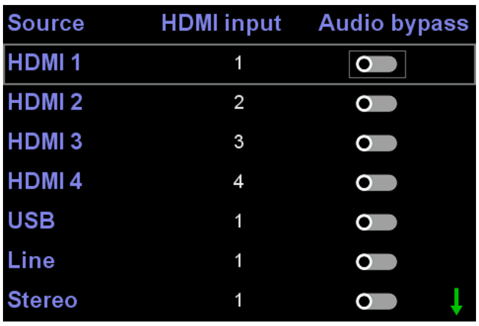
CLK tx=rx is relevant only to the AES-EBU auxiliary interface card. When disabled, the AES-EBU output (tx) clock is derived from the internal A16 clock (48kHz). When enabled, the AES_EBU output (tx) clock slaves to the AES-EBU input (rx) clock.

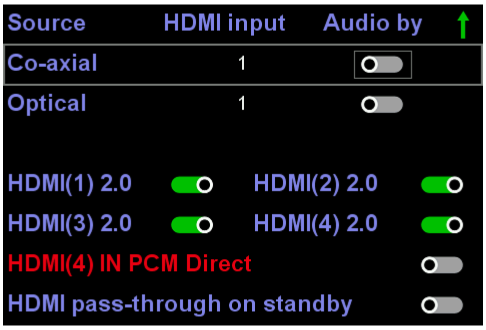
5.3.5.6 LCD off

When enabled, LCD off causes the LCD display will automatically turn off after a pre-determined period of remote-control inactivity. The inactivity period can be set from 5 minutes to 30 minutes in step of 5 minutes.

5.3.6 HDMI settings

HDMI audio signals are physically linked to their corresponding video signal and therefore, when an HDMI audio source is selected the correct video source is automatically switched to the HDMI output and is seen on the video monitor. This is not the case for non-HDMI audio signals, and this menu allows a linkage to be created for the purpose of switching the correct HDMI video signal to the HDMI output.





Default HDMI input menu: linking non-HDMI audio input sources to HDMI inputs for the purpose of switching and viewing the correct video signal

5.3.6.1 Source

All non-HDMI audio sources are listed, and each can be given a different assignment.

5.3.6.2 HDMI input

Each source should be allocated an HDMI input. The default assignment is HDMI 1.

5.3.6.3 Audio bypass

When enabled, this switches the HDMI audio signal as well as the HDMI video signal.

5.3.6.4 HDMI(4) Direct PCM input

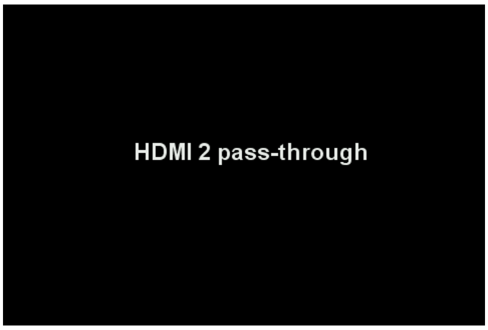
When enabled PCM audio input over HDMI (4) can be routed directly to the SVS processing, bypassing the APM89L/110 decoder board entirely. The PCM audio latency in this mode is fixed at 30ms. Supported layouts are PCM 2ch, 5.1ch and 7.1ch (24-bits, 44.1 to 192kHz). Please note that the EDID tables within the A16 have not been altered in this mode and still advertise all Dolby decode capability to upstream devices as well as PCM. Hence for Dolby/DTS content it may be necessary to configure the source player to decode locally to ensure PCM is sent over HDMI. The A16 will keep the audio mute if it receives anything other than PCM. To use this mode first ensure the active user A Preset configuration under PCM Audio management, HDMI up mixer is set to Direct.

The HDMI (4) PCM bypass mode will use the PCM room assigned in any preset. Raw HDMI PCM channels assignments are non-standard in that the C and SW channels are swapped compared to industry norms. Hence PCM rooms that will be used in this bypass mode must be built with this in mind. The HDMI channel assignments covered in 2.0, 5.1, and 7.1 are shown below.

Ch	*SPKR*
1	*L*
2	*R*
3	*SW*
4	*C*
5	*Lss*
6	*Rss*
7	*Lb*
8	*RB*

5.3.6.5 HDMI pass-through on Standby

When enabled, HDMI pass-through (audio and video) allows the current HDMI input to remain connected to the HDMI output while the A16 operates in standby. Both HDMI audio and video signals are routed from input to output in this mode. On entering standby (pressing remote control power button) the HDMI output is connected to the currently selected HDMI input and the HDMI source and sink re-establish a new AV connection. At this time, the HDMI input currently being passed through to the HDMI output is displayed on the LCD.



In HDMI pass-through mode the A16 must remain powered to keep the HDMI sub-assembly active. However, all unnecessary A16 processing is halted to minimize power consumption. To exit HDMI pass-through the user can either press the remote-control power button or momentarily depress the user A or user B volume knobs.

5.3.7 Full factory restore

When enabled, this will restore the A16 to the factory default settings. This includes all the configuration data, all the Listening Rooms and all Presets for both users. It also erases any PRIR and HPEQ measurement data from the circular memory buffers. However, PRIR and HPEQ files in permanent memory are not erased.

CAUTION: *This command may permanently erase important PRIR and HPEQ measurement data and other settings.*

5.3.8 Factory Tests

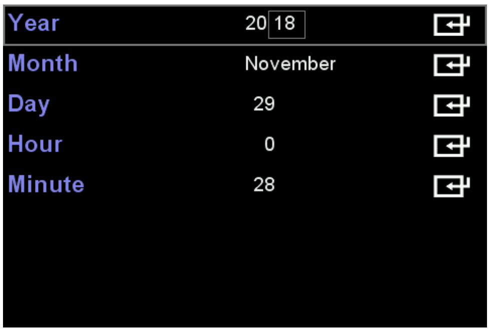
Factory tests are used during post assembly testing to confirm the performance of each A16 and are not required during the normal operation of the A16. As such, we do not recommend the user rerun these tests.

CAUTION: *These tests output full range analogue and digital signals which can damage hearing and audio equipment.*

5.4 Time

Used to set the date and time using the ADJ+ and ADJ- keys. The current date and time are also added to PRIR and HPEQ measurements for identification purposes.

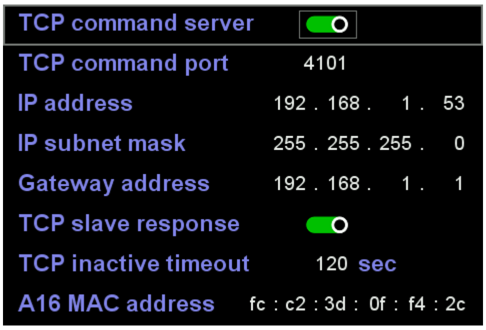
NOTE: *The ENTER key must pressed prior to leaving time menu to save the new date/time values*



Time menu: setting the current date and time

5.5 Network

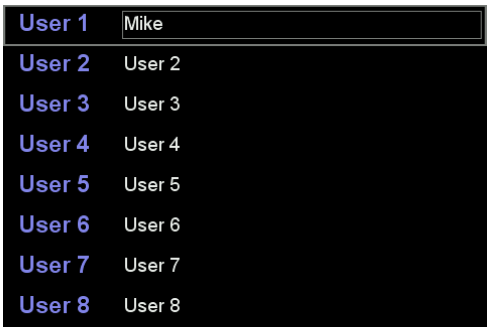
The A16 implements a simple TCP based command and response protocol that allows a remote device to control and/or monitor certain operational aspects of the A16 in real time over a home network. Please refer to the TCP Command Server chapter for more information.



Network settings for the TCP command server.

5.6 Users

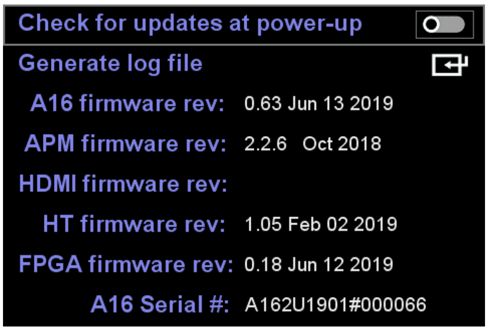
Eight different user names can be added using the alpha-numeric keyboard to insert text. These names are then added to measured PRIR and HPEQ files allowing these files to be more easily identified. These names are also used to save and select presets.



Users menu: the names of eight users can be stored

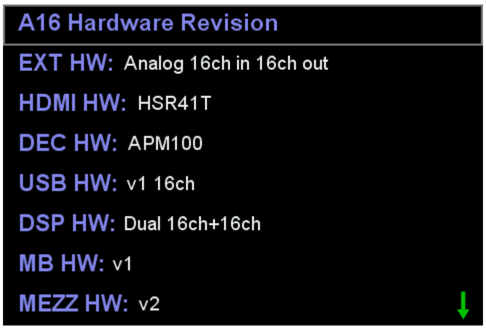
5.7 Updates/About

The first window in Update/About provides information about the firmware running on the various internal sub-systems. It also shows the serial number of the specific A16 unit.



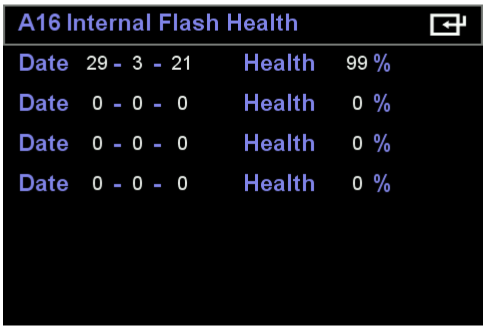
A16 Firmware revision numbers

The second window provides information about the installed hardware sub-systems which is used by the factory for verification.



A16 Hardware versions

The third window optionally provides health status information about the A16 internal flash storage memory. A history of the last four 1% changes is shown (this feature is operational only for certain A16 hardware configurations).



A16 Internal flash health status (where applicable)

5.7.1 Check for updates at power-up

Normally set OFF. When set ON the unit will scan the SD card and the internal permanent memory for any new firmware revisions that may have been downloaded previously and will update each programmable part if a newer version is detected.

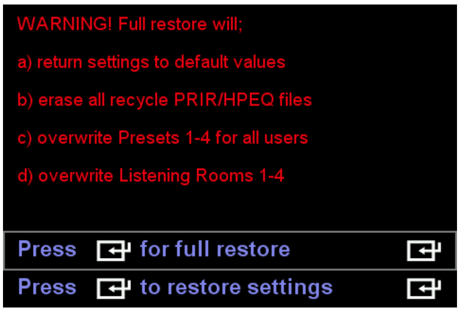
NOTE: *Full details for updating the A16 with new firmware are given in the appendices:* ***Updating the Realiser A16 Firmware****.*

5.7.2 Generate log file

This generates a small 1 kbyte file and writes it to the Realiser folder of an SD-card. The log file consists of information that uniquely identifies the A16, such as the serial number and can be used to create an account on the Realiser Exchange website – check website for details.

5.8 Restore factory setup

This option returns some of the core A16 settings to the factory default condition. It is intended to allow users to get the system working again. It uses pre-installed factory default PRIRs and HPEQ files to generate default Listening Rooms, and then creates default Presets based on these Listening rooms. This function also erases PRIR and HPEQ files in the recycle buffers, overwrites Listening Rooms 1 to 4, and overwrites Presets 1 to 4. However, restore never overwrites PRIR or HPEQ files in the A16 internal storage areas. Therefore, users should save any important PRIR and HPEQ files that may still reside in the recycle buffers to SD card, or to internal storage, before proceeding.

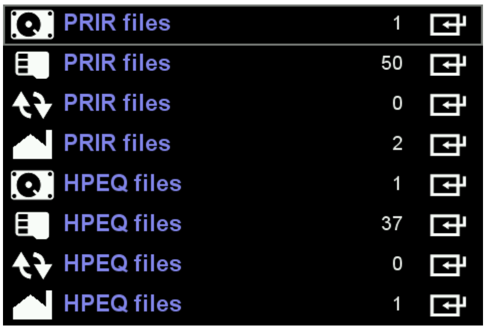


Restore factory setup menu: Warning message

6.1 Files menu (PRIR and HPEQ)

On the Home Page menu, the PRIR/HPEQ file menu is accessed through the Files option.

Home Page menu: **Files** menu



File menu showing the number of PRIR and HPEQ files in each location.

6.2 Memory locations

Permanent internal storage for PRIR and HPEQ files. Files can be moved into permanent memory from the recycle buffer or micro-SDcard. Files can also be deleted from permanent memory. These files are unaffected by factory restore.

External micro-SD card storage for PRIR and HPEQ files. Files can be copied to/from the SD card from/to the permanent internal memory. Files cannot be deleted from the micro-SD card. These files are unaffected by factory restore.

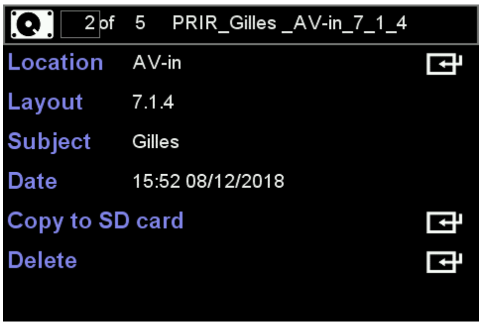
Internal recycle buffer used for storing measured PRIR and HPEQ files. There are sixteen (16) slots in the buffer and the last saved measurement is always stored in slot 1. Measured PRIR and HPEQ files must be moved to permanent internal storage to avoid being over-written. Files can be copied from the recycle buffer to permanent internal storage and to the micro-SD card. Files cannot be deleted from the recycle buffer but will eventually be over-written once the buffer is filled. These files are deleted during a factory restore.

Factory installed PRIR and HPEQ files. These files cannot be copied, deleted or modified. In the event that the firmware of the A16 unit must be reset, these PRIR and HPEQ files are used to re-create factory-default listening rooms for Dolby Atmos, DTS:X and PCM formats. These files are unaffected by factory restore.

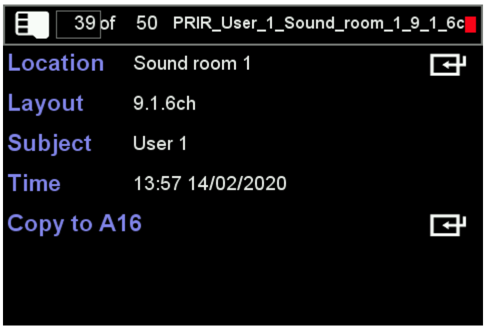
6.3 PRIR files menu

Home Page menu: **Files** menu: **PRIR files** menu

The PRIR files menus for all four memory locations are similar but differ in the options for moving or deleting the files.



Selecting a PRIR file from permanent storage.



Selecting a PRIR file external SD card.

6.3.1 Location

This is the name of the room where the PRIR was measured. If a photograph of the room is attached to the PRIR it can be viewed using the ENTER key.

Image of the room in which the PRIR was measured.

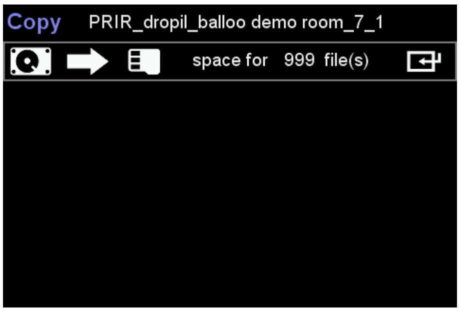
Image of the room in which the PRIR was measured.

6.3.2 Layout, Subject, Date

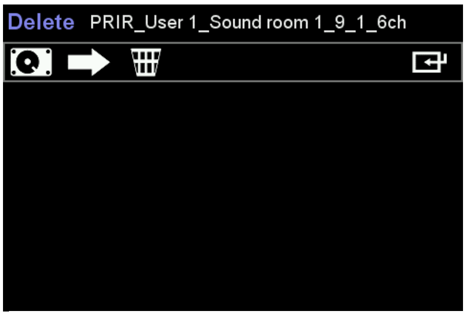
Information relating to the PRIR measurement to assist in identifying a given PRIR. The layout and subject fields for internal PRIR files can also be edited using the standard text entry method up to a maximum of 32 characters for each.

6.3.3 Copy to SD card menu and Delete menu

PRIR files may be copied from permanent storage or the recycle buffer to an external SD card (when available). PRIR files may also be deleted from permanent storage – but cannot be deleted from the external micro-SD card or the recycle buffer.



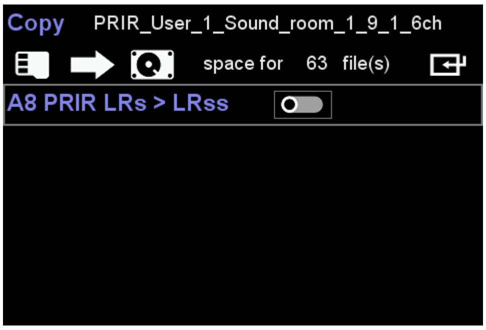
Copying a PRIR file from permanent memory to an SD-card

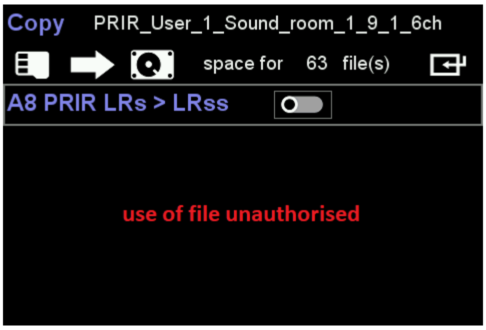


Deleting a PRIR file from permanent memory.

6.3.4 Copy from SD to internal file

PRIR files located in a /realiser/PRIRs directory of an external SD card may be copied into 1 of 64 internal storage slots. The slot number is automatically allocated by the system. If all 64 slots are already occupied, then it will be necessary to delete an internal file to free up space. The SD card reader can also parse Realiser A8 rev 2.0 PRIR files (PRIR20xx.SVS) and convert them to A16 PRIR files, prior to storing them into internal storage. When A8 PRIR LRs>LRss is enabled, the A8 PRIR parser will relabel surround speakers as side-surround speakers in the new A16 version of the PRIR. This is useful where you wish to use an A8 5.1ch PRIR to build rooms that only specify side-surround speakers.





The SD card reader always checks if the user has permission to use the PRIR file being copied into their A16. If the PRIR is locked to another A16, the file transfer is abandoned, and an authorisation warning displayed.

6.3.5 Changing the PRIR filename, subject and layout

The filename, layout and subject fields of internal and recycle PRIR files can be edited using the standard text entry method up to a maximum of 32 characters for each.

6.4 HPEQ files menu

Home Page menu: **Files** menu: **HPEQ files** menu

The HPEQ files menus for all four memory locations are similar, but differ in the options for moving or deleting the files.

HPEQ files menu

6.4.1 Phones, Subject, Time

Information relating to the HPEQ measurement to assist in identifying a particular HPEQ file.

6.4.2 Content

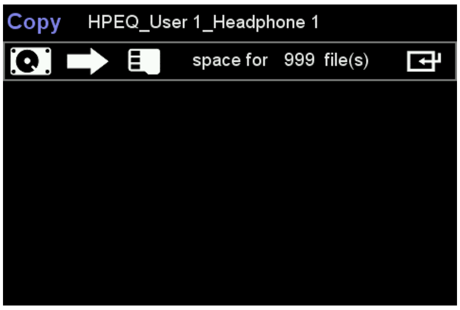
This describes the EQ information available in the HPEQ file. Currently there are four sets of data, one set (autoEQ) taken with the automatic HPEQ measurement procedure, one set (flatEQ) generated at the same time as the autoEQ, and two optional sets that are typically manual adjustments to the autoEQ or flatEQ data. The flatEQ filter was designed to be used as the base filter for IEM-type headphones.

autoEQ*:* *All HPEQ files contain autoEQ data, measured using the automated EQ procedure.*
flatEQ*:* *This EQ filter is flat and is typically used as the base filter for IEM-type headphones.*
manLOUD*:* *Manual EQ adjustments determined using the Equal Loudness EQ measurement procedure.*
manSPKR*:* *Manual EQ adjustments determined using the External Speaker EQ measurement procedure.*

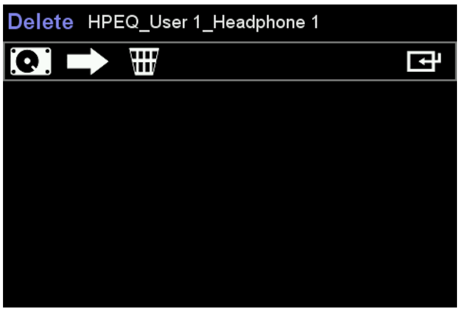
A HPEQ file in the recycle buffer showing three filters, autoEQ, manLOUD and manSPKR. Any of these filters can be chosen as the HPEQ filter within a preset.

6.4.3 Copy to SD card menu and Delete menu

HPEQ files may be copied from permanent storage or the recycle buffer to an external SD card (when inserted). PRIR files may also be deleted from permanent storage – but cannot be deleted from the external micro-SD card or the recycle buffer. Use a computer to delete files from a micro-SD card.



Copying a HPEQ file from permanent memory to an external SD-card



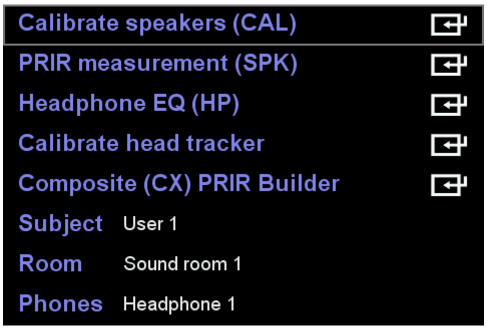
Deleting a HPEQ file from permanent memory

6.4.4 Changing the HPEQ filename, subject and layout

The filename, layout and subject fields of internal and recycle HPEQ files can be edited using the standard text entry method up to a maximum of 32 characters for each.

7 APPs (applications)

APPS or applications are routines that generally run on the A16 in place of the regular decoding/SVS rendering system. This means that the normal operation of the A16 may be suspended while any application program is running.



7.1 Subject-Room-Phones

The ‘Subject’ field can select from any of the eight optional usernames, the ‘Room’ field can select from either of PRIR sound rooms, while the ‘Phones’ field can select from any of the four optional headphone model names. These fields are used to populate the files generated by the ‘PRIR measurement’ and ‘Headphone EQ’ apps.

7.2 Calibrate speakers (CAL)

Calibrating the speakers is an application used as part of the PRIR measurement procedure. Please consult the PRIR measurement chapter for further details.

7.3 PRIR measurement (SPK)

Please consult ‘Measuring a new PRIR in a sound room using the Synchronous method’ chapter for further details.

7.4 Headphone EQ (HP)

Please consult the ‘Measuring personalised HPEQ filters’ chapter for further details.

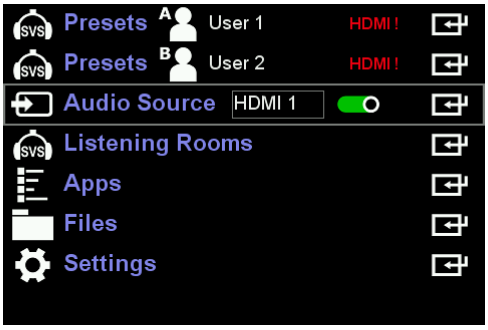
7.5 Calibrate head tracker

Please consult the ‘Calibrating the magnetic sensor in the head tracker device’ appendix for further details

7.6 Composite (CX) PRIR Builder

Please consult the ‘Building a composite PRIR’ chapter for further details.

8 Audio Source Selection

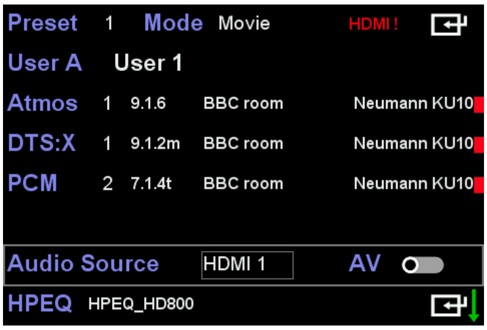


Audio Sources that can be selected are as follows;

1) *eARC (where the HDMI board is present)*
2) *HDMI 1 input*
3) *HDMI 2 input*
4) *HDMI 3 input*
5) *HDMI 4 input*
6) *USB*
7) *Line (or AES-EBU where the digital auxiliary board is present)*
8) *SPDIF Co-axial*
9) *SPDIF Optical*

The audio source selection is controlled in the home page when the global slider is enabled.

When disabled, control of the audio source moves to the individual User A active Presets (example below).



9 Configuring a Preset for SVS headphone or AV mode

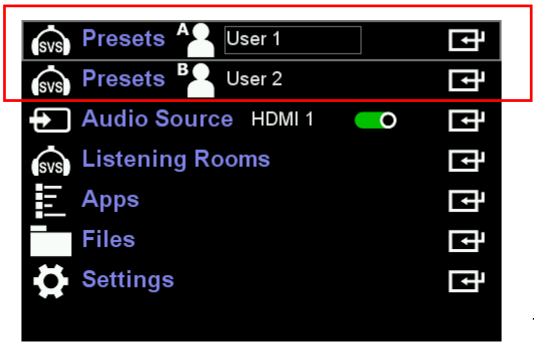
The primary purpose of a Preset is to select personalised Listening Rooms and Headphone EQ filters for an individual listener. Presets bring together Listening Rooms (which contain PRIR data), configuration data and user information and are the gateway for running the headphone virtualisation process of the A16. Presets contain Listening Rooms for each of the major listening formats (Dolby Atmos, DTS:X and PCM) and can automatically switch between these rooms when the incoming bitstream changes. The selected Listening Rooms also configure the rendering /listening mode of the bitstream decoder. For example, one Preset may select to decode and render Dolby Atmos in a room with a 9.1.4ch configuration, and to decode DTS:X in a different room with a 7.2.4ch configuration. The A16 can then automatically switch between these different configurations if the bitstream changes. The loudspeaker configuration of a room may also be called the Listening Mode in this manual.

NOTE: *Presets are configured and stored independently for each user, A and B. However, the design of the A16 requires that the same Listening Mode be operating for both users A and B, and this is set from the Preset for User A. For example, if the current Preset for User A demands a 7.1.4ch listening mode, while the current Preset for User B is requesting a 2.0ch listening mode, the active Listening Mode will be set to 7.1.4ch but User B will only render the first 2 channels of the full 7.1.4ch track.*

NOTE: *For the correct operation of the A16 it is vital that Presets for both User A and User B are loaded and active, since much of the real-time functionality of the A16 (e.g. head-tracking) requires that both DSPs are running correctly.*

9.1 The Home Page Menu

The Home Page Menu is the starting point for navigation through the menus and provides access to all functions and features of the Realiser A16. The total number of different presets that can be stored is 256. This number comes from: 2 (User A and B) x 8 (User names) x 16 (presets per user name) = 256 presets.

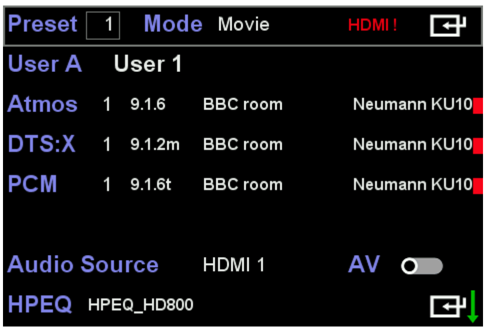


The Home Menu page with the User Presets highlighted

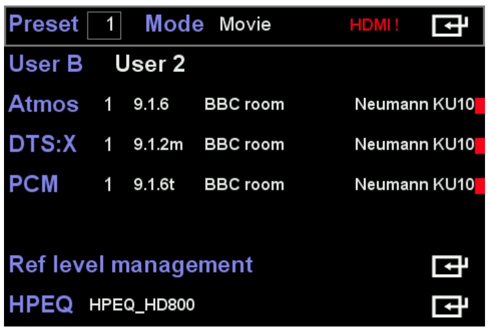
1. *Select which User Preset to configure – choose User A or User B using the UP and DOWN arrow keys*
2. *Select the User Name for the preset using the ADJ+ and ADJ- keys*
3. *Move to the Preset Menu page using the ENTER key*

9.2 The Preset Menu

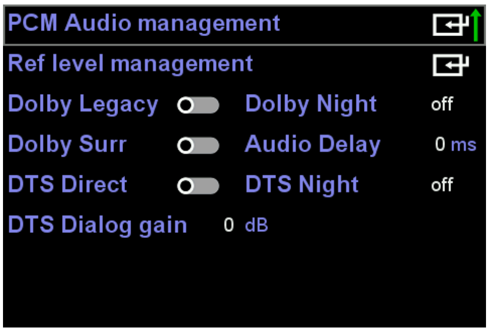
The Preset configuration menu for User A is spread over two pages, while the configuration menu for user B has fewer options and is contained within one page. Presets allow individual users to choose their preferred listening room for different audio bitstreams and allows the configuration of these listening rooms to be modified. For example, a single user could configure two presets to have the same listening rooms for Dolby and DTS bitstreams but configure each preset to use a different up mixer when rendering PCM audio streams. The Preset menu is also used to select a personalised Headphone EQ measurement for an individual listener.



User A Preset Menu page 1



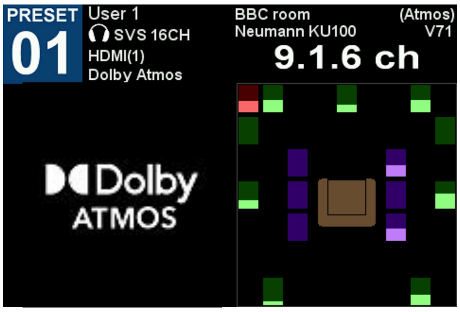
User B Preset Menu page 1



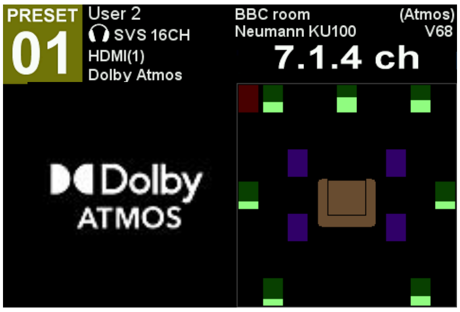
User A Preset Menu page 2

9.2.1 Select the Preset number

For each username sixteen different presets can be configured and saved internally. The factory default initially stores the same 16 presets for Users 1 thru 8 for both listener A and B. Use the ADJ+ and ADJ- keys to select an individual preset, for activating or to change the configuration of the preset. The ENTER key loads and activates the selected Preset. Once the preset is active the PA key will show the Preset Speaker Map page for User A (blue background to the preset number) and the PB key will show the Speaker Map page for User B (green background)



Preset Speaker Map for User A



Preset Speaker Map for User B

9.2.2 Set the SVS Rendering Mode

The Mode selector for the Preset is currently fixed to SVS Movie mode. Other modes may be available in future firmware versions.

9.2.3 Verify the User

The User name is informational only. It is selected in the previous menu (Home Menu page) by hovering the cursor over the preset option and using ADJ+ and ADJ- buttons to select a user.

9.2.4 Select the Atmos, DTS:X and PCM listening rooms for this preset number

These are the Listening Rooms that have been previously created for each bitstream format. Up to 32 different listening rooms in each format are available for selection in a Preset. For example, in the User A Preset menu page above User 1, has configured Preset 1 to use Atmos Listening Room #1, DTS:X Listening Room #1 and PCM Listening Room #1. These listening rooms have been created from dummy head measurements (Neumann KU100) but would more typically be created from room impulse response data (PRIR data) personalised to the User 1.

9.2.5 Audio Source

This selects the audio input source for this preset only when global audio source selection in the home page has been disabled. This allows the audio source to change between presets.

9.2.6 Toggle the AV mode ON or OFF

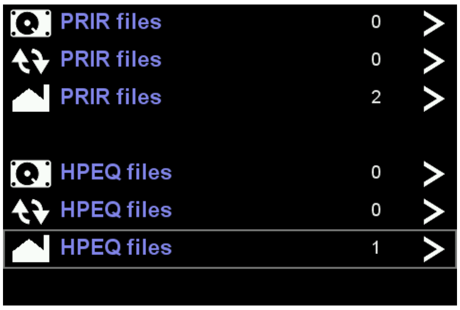
Normally set OFF. When set OFF this Preset, when active, will not allow audio to be output to the 16-ch line outputs, and can only be used for SVS headphone rendering of decoded audio. When set ON this Preset, when active, allows decoded audio to be output to either the 16-ch line outputs or to be rendered to the SVS headphone outputs. In AV mode the A16 is being used as an AV receiver and the decoded audio signals are routed directly from the phono outputs to loudspeaker amplifiers.

When the AV mode is enabled for a preset the loudspeaker icon and headphone icon keys on the A16 remote-control are used to toggle between the AV loudspeaker mode and SVS headphone mode.

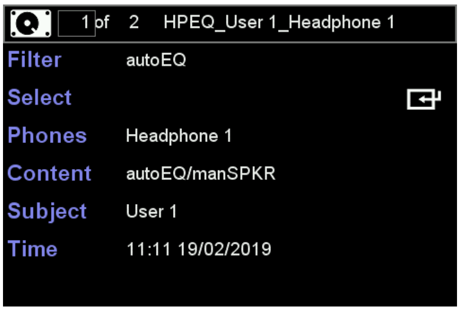
NOTE: *The AV switching will not function unless A/B Demo mode under HT settings is disabled.*

9.2.7 HPEQ menu

The HPEQ file is intended to store the impulse response data of a particular pair of headphones calibrated for an individual user, often using binaural microphones mounted in the user's ear canals. The HPEQ file will typically contain inverse filter coefficients generated automatically from the measured impulse response data (autoEQ) and may also contain a flatEQ filter and filters generated manually (manLOUD and manSPKR). In addition, the HPEQ file contains identifying information such as the subject's name, the model name of the headphones and the time/date of the HPEQ measurement. The HPEQ option shows the currently selected HPEQ file for this Preset. To select a different HPEQ file use the ENTER key to bring up the file menu for HPEQ files, then navigate to one of the three available sources of HPEQ files, and finally SELECT an individual HPEQ file using the ENTER command key. The display will return automatically to the Preset configuration menu, with the newly selected HPEQ file now displayed in the HPEQ option for the preset.



Navigate to the source of the HPEQ files



Choose the correct HPEQ file, and the correct filter within this file, and then select it. All available filters are shown under Content.

Select a particular HPEQ file from the source location using the ADJ+ and ADJ- keys.

9.2.7.1 Filter

Select the required inverse filter; autoEQ, flatEQ, manLOUD or manSPKR (if available). All available filters within the HPEQ file are shown in the Content line. AutoEQ will always be available as a filter option.

9.2.7.2 Select

Finally, use the ENTER key to select this HPEQ inverse filter into the preset. The display will automatically revert to the preset configuration page.

9.2.7.3 Phones, Subject, Time

Descriptive information about the measured headphones, the measured subject and the date and time of the measurement. These are all set during the actual HPEQ measurement.

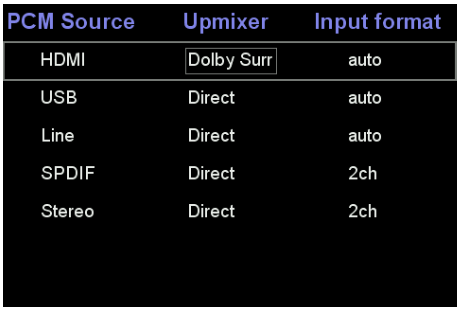
9.2.7.4 Content

Displays the valid inverse filters contained within the measured HPEQ file. AutoEQ will always be displayed and available as an option, whilst the two manual EQ filters (manLOUD and manSPKR) will only be displayed if they are available. *The manual EQ filters are normally based on the automatically generated HPEQ autoEQ filter or flatEQ filter and details of how these are generated are found in the 'Measuring personalised HPEQ' chapter.*

NOTE: *A valid HPEQ file must be* ***ALWAYS*** *be selected for User A and User B for SVS headphone rendering. The factory default HPEQ files can always be used.*

9.2.8 PCM Audio management

PCM Audio management is an option found on the second page of the Preset configuration page for User A. This option leads to the PCM Audio Management menu and configures the A16 to optionally either up-mix or pass-thru PCM audio signals from the HDMI or SPDIF inputs. For all other input types the up mixer option is not available.



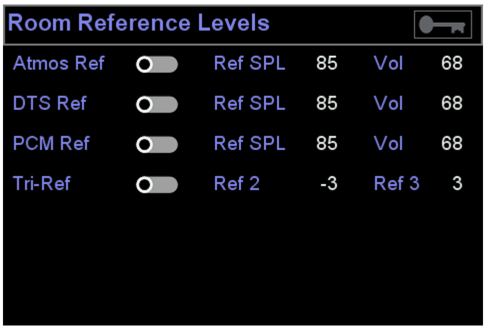
PCM Audio Management Menu: selecting the upmixer for PCM audio sourced from HDMI and SPDIF inputs

Digital PCM signals from any of the HDMI inputs can be mono, stereo, or multichannel, while digital PCM signals from the SPDIF inputs (co-axial or optical) are limited to mono or stereo. For each of these inputs the user can select to either use the direct (or unmodified) digital PCM signal, whether mono, stereo or multichannel, or can use an up mixer. Either Dolby Surround or DTS Neural:X up mixers can be selected as the up mixer for either source. The BACK key saves these settings are moves back to the previous preset configuration menu.

***Note:** this is only selecting the upmixer for PCM signals – not for bitstream audio data.*

9.2.9 Ref Level management

Ref Level management is an option found on the second page of the Preset configuration page for both User A and User B. Room Reference Levels enables or disables the use of reference level indexed volume display for Atmos, DTS:X and PCM listening rooms specified in the Preset. It also enables or disables the use of a fixed three level volume mode for any room type. Details of these functions are found elsewhere in this manual.



Reference Level Management Menu for both User A and User B presets

9.2.10 Dolby Legacy decode

When enabled, decoded Dolby Atmos audio beyond 7.1 is muted. This is intended for users who have a legacy speaker layout, such as stereo, 5.1ch or 7.1ch, and who do not have any height speakers. When disabled, this instructs the Dolby Atmos decoder to operate normally.

9.2.11 Dolby Night mode

This option limits the dynamic range of the decoded Dolby Atmos output signals, and can be set ON, OFF or Auto.

9.2.12 Dolby Surround

When enabled, this instructs the Dolby Atmos decoder to up mix, using Dolby Surround, any decoded legacy Dolby bitstreams to the selected listening mode for this preset, for any detected Dolby bitstream. When disabled, this instructs the Dolby Atmos decoder NOT to up mix any legacy Dolby bitstreams.

9.2.13 DTS direct

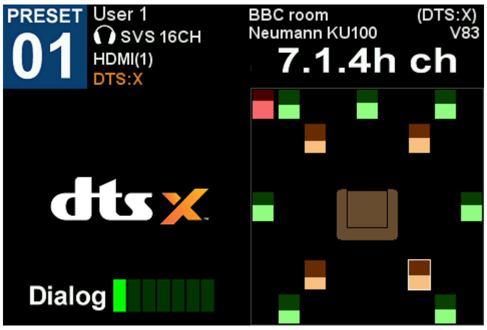
When enabled, decoded legacy DTS bitstreams are output in their native format and are not subject to Neural:X up mixing. This mode is used when the user wishes to hear the audio exactly as the producer intended. When disabled, this instructs the DTS:X decoder to operate normally and where the number of speakers in the listening room is different to that of the decoded bit stream, up or down mixing will be applied automatically.

9.2.14 DTS Night mode

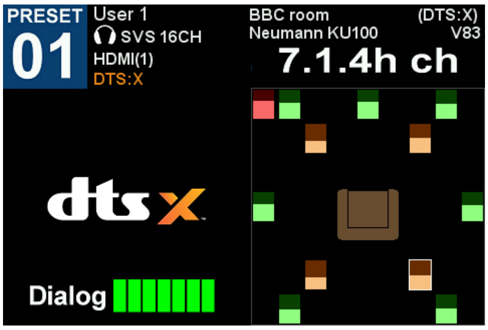
This option limits the dynamic range of the decoded DTS:X output signals. When disabled full dynamic range is preserved. When enabled the dynamic range of the audio is reduced.

9.2.15 DTS Dialog gain

DTS dialog gain sets the gain of the dialog object between 0 and 6dB. It is only activated on receiving a DTS:X bitstream where dialog control is flagged. In this case the speaker map display will include a graphical dialog gain meter below the main DTS:X logo that indicates the gain.



DTS dialog gain meter (0dB gain)



DTS dialog gain meter (6dB gain)

A gain of 0dB illuminates just the left most segment. A gain of 6dB illuminates all 7 segments.

Note: To save the complete preset configuration move BACK to the main preset configuration page and then move BACK to the Home Menu Page. The Preset configuration will be saved but will not be loaded or activated.

9.3 Load and Activate presets for User A and User B

1. **Select the audio source:** In the Home Page menu navigate to the Audio Source option, and use the ADJ+ and ADJ- keys to select the audio source.
2. **Select a user name for Preset A, then load and activate the Preset**
   a. In the Home Page menu navigate to the Presets menu option for User A, then use the ADJ+ and ADJ- keys to select a user name for Preset A.
   b. Press ENTER to move to the Preset Configuration menu.
   c. Scroll through the available Presets for this user name using the ADJ+ or ADJ- keys.
   d. To load a selected Preset use the ENTER key. A short “Loading” message will be displayed in the top line during this operation, followed by a short “Active” message to indicate that the Preset is now running.
3. **Navigate BACK to the Home Page, select a user name for Preset B, then load and activate the Preset**
   a. In the Home Page menu navigate to the Presets menu option for User B, then use the ADJ+ and ADJ- keys to select a user name for Preset B.
   b. Press ENTER to move to the Preset Configuration menu.
   c. Scroll through the available Presets for this user name using the ADJ+ or ADJ- keys.
   d. To load a selected Preset use the ENTER key. A short “Loading” message will be displayed in the top line during this operation, followed by a short “Active” message to indicate that the Preset is now running.
4. **Display the Speaker Map for the active Presets for User A or User B**
   a. Once a Preset is loaded and activated the Preset Speaker Map for either User A or User B can be displayed using the PA or PB keys (Preset A or Preset B) on the remote control.
   b. The ADJ+ and ADJ- keys scroll up and down through the 16 presets for each user. (Note: it takes a few seconds to load each new preset.)

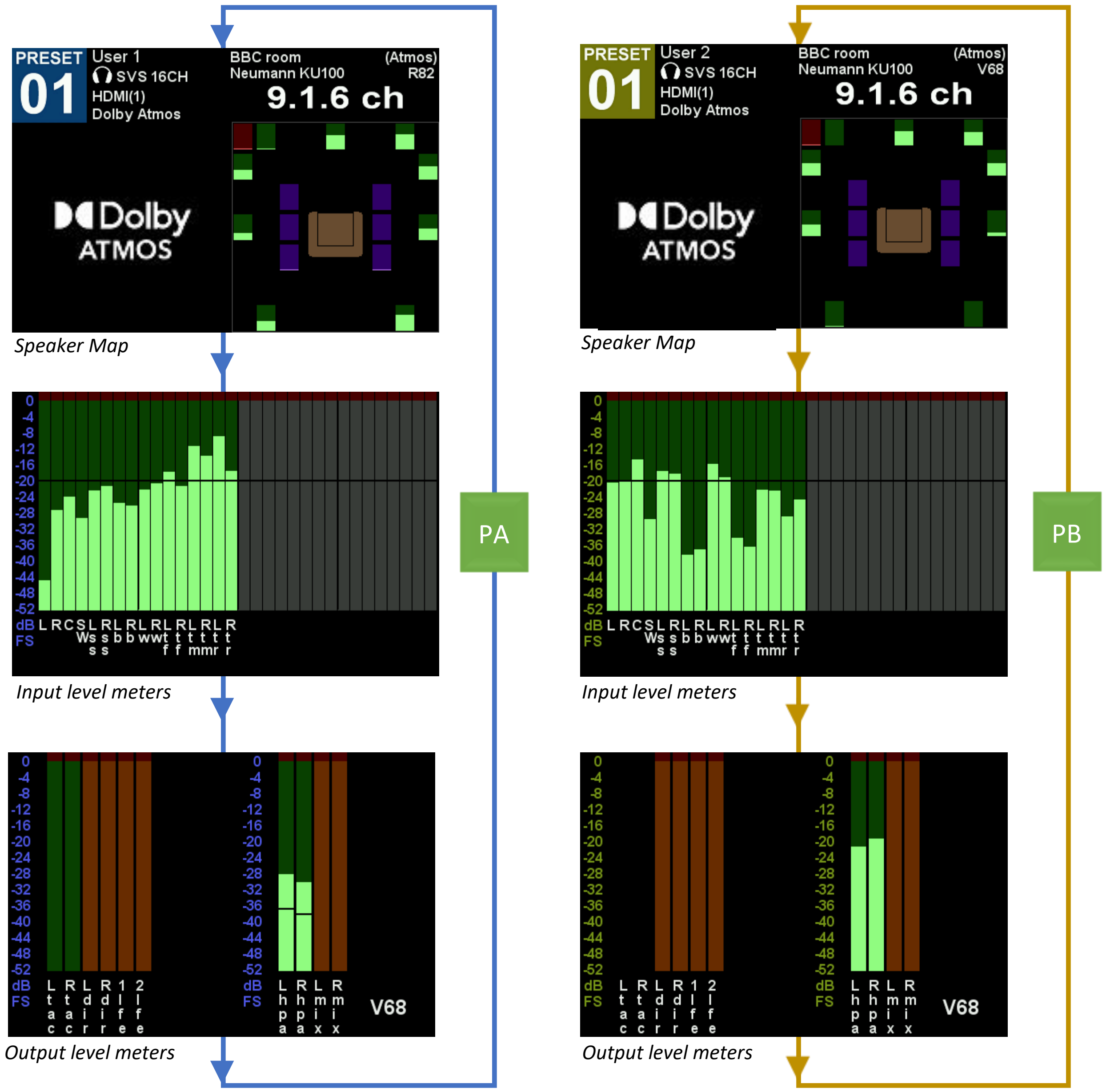
The background colour around the Preset number indicates that the preset is either for User A (blue) or User B (green). In the graphic panel each speaker icon also displays the decoded signal level for that speaker, whether in AV loudspeaker mode or SVS headphone mode. Information relating to the names of the speakers displayed in the graphic panel can be found in **Appendix B Graphical representation of loudspeakers in the Speaker Map display of the A16.**

Note: The Preset Speaker Map shows the current ACTIVE preset for users A and B. When a preset is selected in the Preset Menu, a copy of the preset is loaded into the DSP and activated. Therefore, it is possible to return to the Preset Menu and change some parameters of the original preset, without changing the active preset. The changes will only take effect when the modified preset is loaded and activated. If changes are made in the currently active preset the user is prompted to **RELOAD** the preset.

Note: Changes to a preset involving operation of the head tracker (and some other real-time parameters) will take effect instantly, without needing to reload the preset.

9.4 *The PA and PB key*

If the presets for User A and User B have been successfully loaded and activated, the Speaker Map of the active preset for each user can be displayed using the PA or PB keys on the remote control.



For User A the preset number shows a blue background while User B has a greenish background. Pressing the PA/PB key for a second time brings up the input level meters, while the output levels meters are shown on the third press. Further presses rotate between these three displays.

9.4.1 *Input Level Meters*

The input level meters indicate the peak PCM audio level of any channel input to the SVS rendering DSP. The speaker names are taken from the currently active preset (either A or B). The levels shown are identical in value to those displayed in each speaker icon in the Speaker Map display. Clipping (red segment) on the inputs is also indicated and can be reset using the red CANCEL key on the remote control.

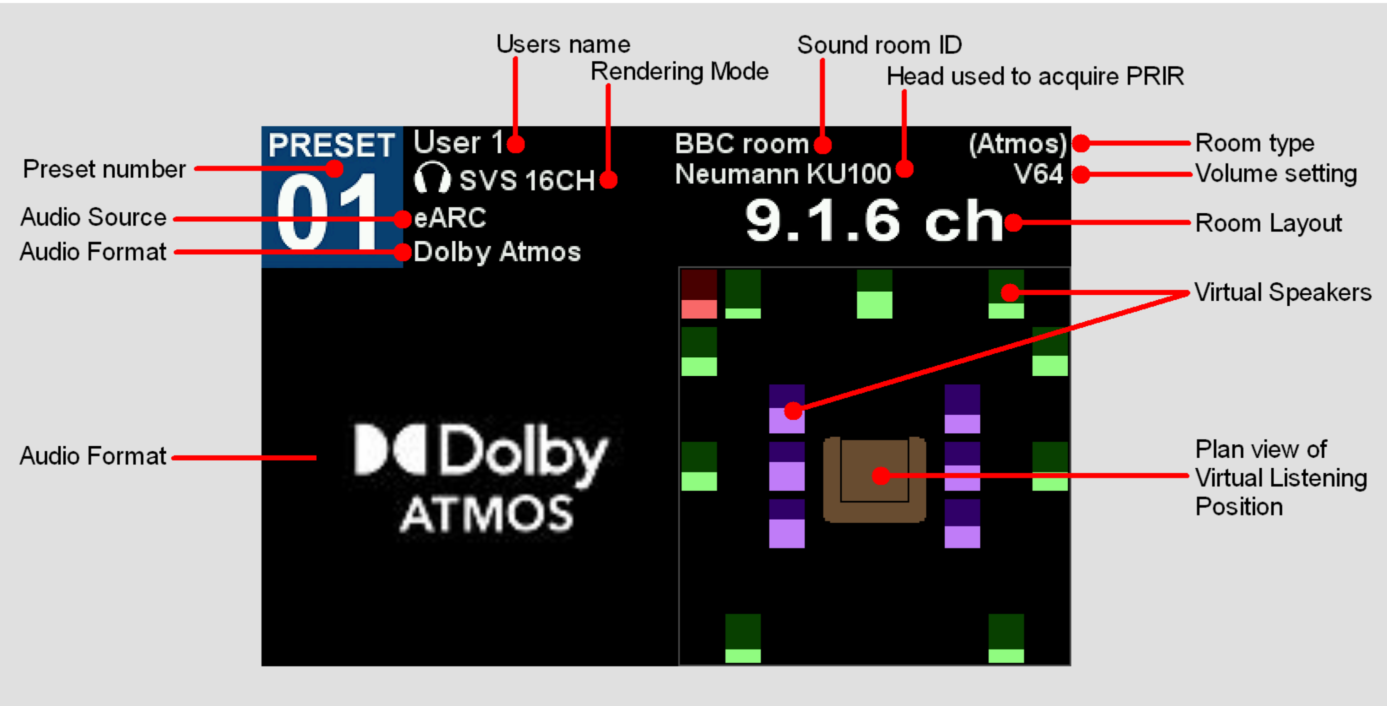
9.4.2 *Output level Meters*

The output level meters also indicate the peak PCM audio levels internal to the SVS rendering DSPs. Green levels meters signify these are signals that are output from the SVS rendering DSPs. Brown level meters signify these signals remain inside the SVS rendering DSPs but can be mixed in to form part of an output signal.

a) Lhp and Rhp are the headphone signal levels.
b) Ltact and Rtact are the tactile signal levels, when enabled. These signals are generated only by User A
c) Ldir and Rdir are the redirected direct bass signal levels, when enabled.
d) Lfe1 and lfe2 are the redirected virtual subwoofer bass signal levels, when enabled.
e) Lmix and Rmix are the stereo mix down signal levels, when enabled.

9.4.3 Elements of the Speaker Map display for any preset

1. **Preset number** and user A (blue background) or B (greenish background)
2. **Users name** – programmed by the user
3. **Rendering Mode** – either SVS 16ch, SVS 24ch, AV (line out) or 2ch (Stereo mix down)
4. **Audio Source** – either eARC, HDMI (1,2,3 or 4), USB, Line, AES-EBU, Stereo, Co-axial or Optical
5. **Audio Format** – either none, Dolby Digital, Dolby Digital Plus, Dolby Atmos, Dolby Surround, DTS:Headphone:X, DTS:X, DTS, PCM, 2ch, PCM 6ch or PCM 8ch
6. **Room Type** – either Atmos, DTS:X or PCM
7. **Sound Room ID** – this refers to the PRIR data (Ch-1) that was used to create this listening room.
8. **Head** – this refers to the PRIR data (Ch-1) that was used to create this listening room.
9. **Room Layout** – valid listening modes for each format as listed in the Appendix B
10. **Virtual Speakers** -a visual representation of the speakers in the Listening Room – further details are listed in Appendix B
11. **Volume Setting** – current volume setting



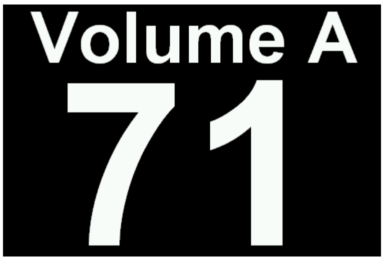
NOTE: *The ADJ+ and ADJ- keys scroll up and down through the 16 presets for each user. It takes a few seconds to load each new preset.*

9.4.4 Controls associated with the Speaker Map display

9.4.4.1 Headphone volume control in SVS Movie mode

The volume of the headphone output is controlled via dedicated volume control rocker keys on the remote control or via the physical volume knobs on the front panel of the A16. The volume control changes the headphone output when the A16 is operating in SVS Movie mode. When operating in AV mode the volume control changes the level of the analogue line outputs which would typically be connected to loudspeaker amplifiers. The volume is displayed momentarily on the LCD display. Both User A and B can set their headphone volumes independently. In

SVS Movie mode changing the volume for either user also changes the Speaker Map display to show the active preset for that user. The headphone outputs of User A and User B can be muted independently using the MUTE toggle keys.



Volume for User A headphone output

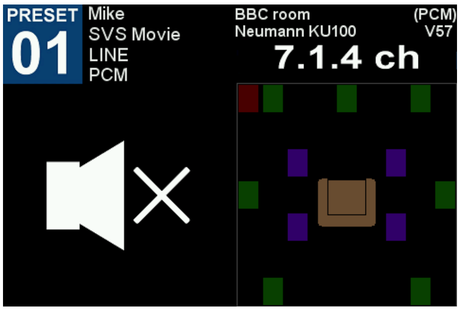
Volume for User B headphone output

9.4.4.2 Line output volume control in AV mode

In AV mode the volume control sets the level of the multichannel analogue output signals that are fed to the loudspeaker amplifiers.

***Note**: In AV Mode only the Speaker Map for User A is displayed – the PB key is not valid – and the line output volume can only be controlled using the user A volume rocker switch or the User A volume knob.*

Line output volume in AV loudspeaker mode



Muting the headphone output for User A using the MUTE key on the remote

9.4.4.3 Mute outputs

Muting the audio output is toggled ON and OFF by the MUTE keys on the remote control. The headphone outputs for User A and User B can be muted independently using the MUTE-A and MUTE-B keys. The line output level is muted using either key.

9.4.4.4 Switch between SVS Movie (headphone) mode and AV (loudspeaker) mode

To engage the AV (loudspeaker) mode use the speaker icon key  on the remote control. To return to SVS Movie (headphone) mode use the SVS headphone icon key  on the remote. In SVS Movie mode the audio is rendered to headphones as virtual loudspeakers. This is the default mode of operation when a preset is loaded and active, and can be engaged using the SVS headphone icon key  on the remote. In AV mode the audio is intended to be rendered to real loudspeakers and is sent directly to the multi-channel line outputs. SVS headphone rendering is switched off.

NOTE: AV mode can only be engaged if AV mode has been enabled in the active preset configuration for user A and the A/B demo mode has been disabled under HT settings. If AV mode cannot be engaged then the active preset will need to be re-loaded.

Move to: **Home Page** menu: **Preset User A/B** menu: **AV mode option**

9.4.4.5 Display head-tracking angles

Whilst in the Speaker Map page the HT key toggles between the real time head tracker angle display and the speaker map page.

Real time head tracking angle display.

The azimuth angles for both User A and User B head trackers are shown. The current head tracker firmware revision and the hardware revision is also shown for both. Press HT key to return to the speaker map display.

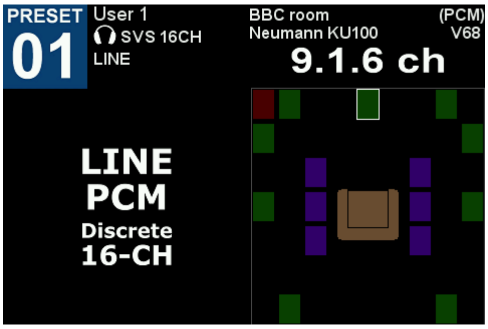
NOTE: Headtracking is only active if presets for user A and user B are loaded and active.

9.4.4.6 SOLO Mode: solo individual speakers in SVS Movie (headphone) or AV (loudspeaker) modes

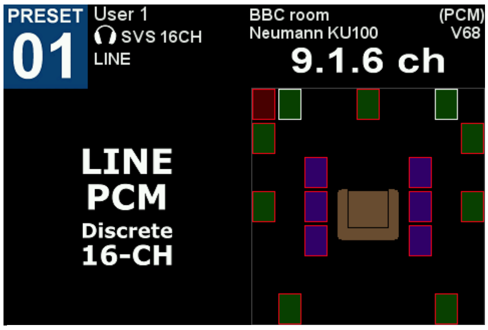
Whilst in the Speaker Map display individual loudspeakers can be soloed, using the individual speaker keys and the alpha-numeric keys on the remote to solo individual speakers. When in the Speaker Map display to SOLO a speaker simply push the key assigned to that speaker. A white box is placed around the speaker icon in the speaker map. Only one speaker at a time can soloed. To disengage the SOLO mode and listen to all the active speakers press the ALL key on the remote. The Assign solo buttons menu is used to change the solo speaker key mapping on the remote control. Any speaker label can be assigned to any of the available keys. (Home Page: Settings: System: Assign solo buttons)

9.4.4.7 MUTE mode: mute individual speakers in SVS Movie (headphone) or AV (loudspeaker) modes

Whilst in the Speaker Map display groups of loudspeakers can be muted, using the individual speaker keys and the alpha-numeric keys on the remote to select/de-select individual speakers. When in the Speaker Map display to enable MUTE mode press the ALL key. A red box is placed around all the muted speaker icons in the speaker map. To toggle mute ON or OFF for an individual speaker use the solo speaker keys. All speakers can be muted or un-muted. To disengage the MUTE mode and listen to all the active speakers again press the ALL key on the remote.



Soloing the Centre virtual speaker while listening over headphones



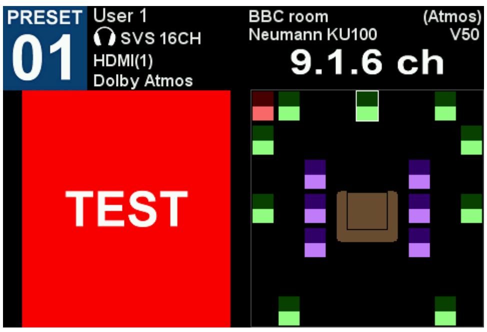
Muting all virtual speakers except for Left and Right speakers.

NOTE: To change the solo speaker key assignment, navigate to the **Assign solo buttons** menu:

Home Page menu: **Settings** menu: **System** menu: **Assign solo buttons** menu

9.4.4.8 TEST mode: play an internal music loop

The TEST key on the remote toggles ON or OFF the playing of an internally generated musical test signal, as a continuous loop, through all active loudspeakers. In the speaker map each loudspeaker icon will show a varying level as the test signal is played. The audio test signal will play in both rendering modes – SVS Movie (headphone) and AV (loudspeakers). While in TEST mode individual speakers can be soloed or muted using the appropriate keys on the remote control. A soloed speaker is indicated by a white border around the speaker icon, and a muted speaker is indicated with a red box. For example, in the example below the centre speaker is being soloed. To turn OFF the SOLO or MUTE modes and listen to ALL the speakers use the ALL key on the remote control. To turn OFF the TEST mode and return to normal operation press the TEST key.



Soloing the Centre virtual speaker over headphones while listening to a looped jingle in Test mode

9.4.4.9 Increment or decrement the Preset Number

While in the Speaker Map display the preset number for user A or B can be changed using the ADJ+ and ADJ- keys to increment or decrement through the list of 16 presets allocated to the currrent named user. This operation takes a few seconds to complete while the new preset is being loaded.

9.4.4.10 Re-route Headphone B signal to Headphone A output (HPB->A)

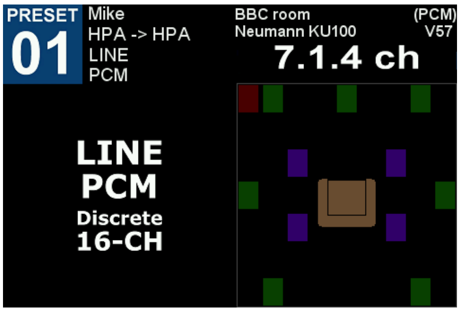
When enabled under Audio settings, and while in the Speaker Map display for User A, the headphone signal for User B can be re-routed (switched) to the headphone output of User A by pressing the RIGHT ARROW icon key on the remote control. To switch back to the normal headphone routing press the LEFT ARROW icon key or navigate away from the Preset Speaker Map display. The HPB→A mode, while enabled, also correctly switches the headtracking input signal and the volume setting from User A to User B. The main purpose of this mode is to allow two presets to be compared almost instantly in an A/B type comparison. This is possible because the presets for user A and user B are both active simultaneously and are rendering the headphone audio independently.

The HPB→A option must be enabled in the System menu.

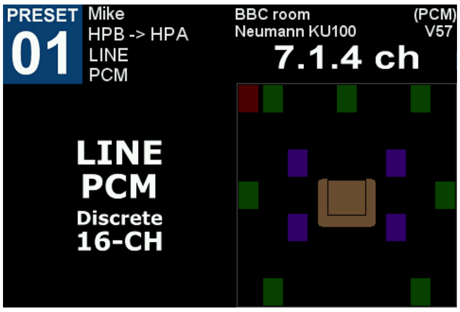
Move to: **Home Page** menu: **Settings** menu: **System** menu: **Audio Settings** menu: **HPB→A option**

NOTE: This mode is only valid when the listener is listening to headphone A output.

NOTE: If the A/B demo mode is also engaged while in HPB->HPA mode, the system returns in HPA->HPA mode.



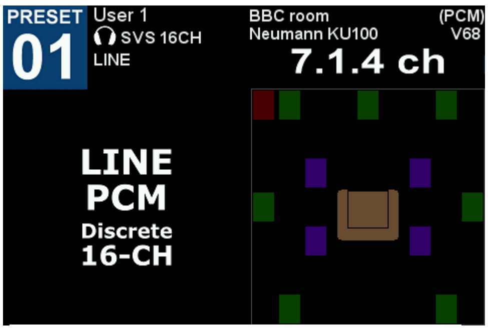
HP B to A mode: allows an A/B comparison to be made between two presets



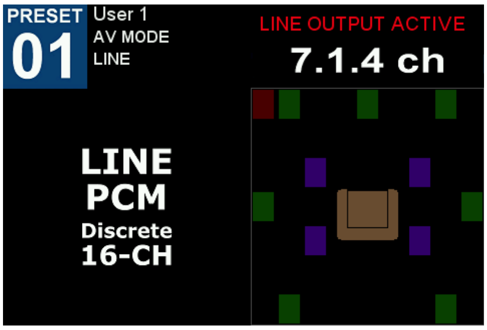
Reverting back to the normal listening mode using the LEFT arrow key

9.4.4.11 A/B demo mode

When enabled under HT settings, the decoded audio signals can be switched from the SVS Headphone to AV line outputs, using the tilt of the head tracker as the A/B switch. For this mode to work, the AV mode in the active Preset must be disabled. When the head tracker is vertical the SVS headphone mode is active, and when the head tracker is tilted forwards the AV mode is active – this facilitates an individual user using their headphones to conduct an A/B listening test between the SVS virtual speakers rendered through headphones and the real loudspeakers in a room. An alternative switching mechanism is to disable A/B demo mode under HT settings and then to enable AV mode in the active Preset and use the SVS headphone icon  key and AV speaker icon  key on the remote control for manual switching.



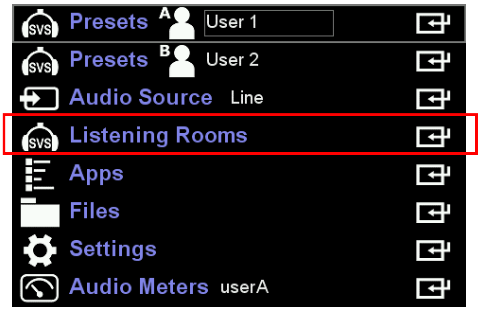
The normal SVS headphone mode.



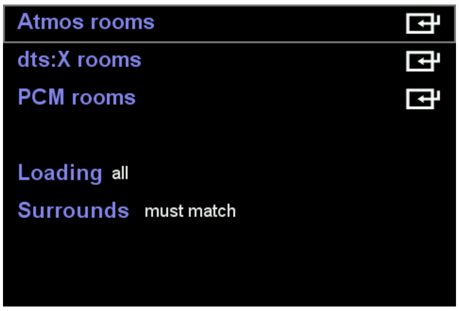
AV mode engaged using the A/B switch activated by the Head-top tilt detector.

10 Building a Listening Room from one or more PRIRs

Listening rooms are configured to reflect the varying loudspeaker setups of different audio decoding formats, such as Dolby Atmos and DTS:X, and other formats. The individual loudspeakers that make up these formats are matched to personalised room impulse response (PRIR) data that contain virtualised versions of the same loudspeakers. A completely virtualised listening room of any format is thereby created suitable for headphone rendering. Listening rooms also contain parameters relating to bass management, reverberation control, stereo mix-down control, and for generating tactile output signals suitable for driving bass-shakers.



Accessing the Listening Rooms menu from the Home Page menu

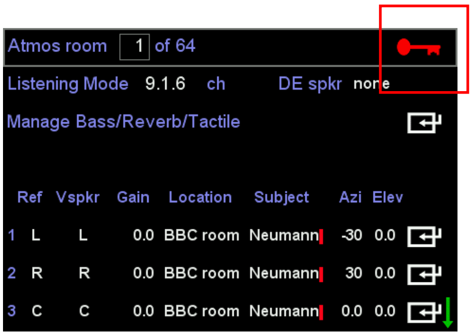


Listening Rooms menu

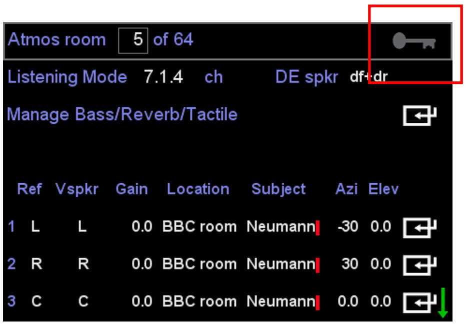
10.1 Select the room type: Atmos, DTS:X or PCM

Move to: **Home Page** menu: **Listening Rooms** menu

The Listening Rooms are listed under three headings corresponding to Dolby Atmos, DTS:X, and PCM, allowing for differences in virtual speaker positions and speaker naming conventions depending on the different formats (Tables 1, 2 and 3 in Appendix A). This list will be increased as new formats emerge and become popular. Up to 32 different Atmos rooms, 32 DTS:X rooms and 32 PCM rooms can be configured and saved. Select the room type and use the ENTER key command to move to the selected room configuration menu.



A locked Atmos listening room configuration.



An unlocked Atmos listening room configuration.

10.2 Configure the selected listening room

10.2.1 Select a room number

Up to 32 different Atmos, DTS:X or PCM rooms can be configured and stored, prior to loading into a preset. Rooms 1 to 4 of all three room-types are used for factory default settings, and therefore it is recommended that these rooms are not changed, and that rooms from 5 to 32 only are customised.

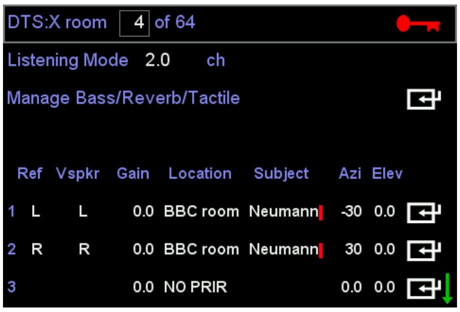
10.2.2 Unlock a room to change its configuration

If a room is locked (red key) the configuration parameters can be viewed but cannot be changed. Unlock a room using the ADJ+ or ADJ- keys in order to edit the parameters.

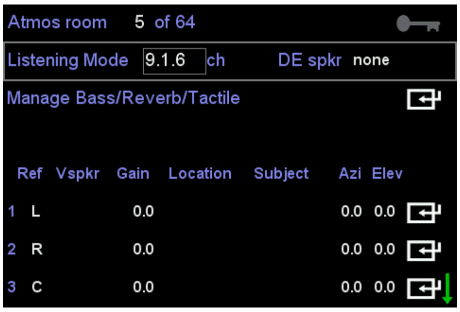
10.2.3 Set the Listening Mode

Use the ADJ+ and ADJ- keys to adjust the listening mode to the preferred format. Dolby Atmos formats have an additional option for Dolby Enabled speakers which should be set correctly for AV listening if used. The listening mode defines the number and arrangement of speakers for a particular listening room configuration. There are over 50 pre-configured listening modes for Dolby Atmos ranging from 2.0ch to 9.1.6ch, and for each mode the audio channels from 1 to 16 are populated with reference speaker names (Ref) that match the selected listening mode. For DTS:X listening rooms there are also over 50 pre-configured listening modes ranging from 2.0ch to 9.2.2h. In the Dolby Atmos and DTS:X listening modes the reference speaker names cannot be changed, and the channel number assigned to each speaker is also fixed – the channel numbers correspond to the 16-channel phono outputs on the rear panel of the A16. For PCM listening rooms, the user can edit the reference speaker names in the pre-configured listening modes and can also create custom layouts.

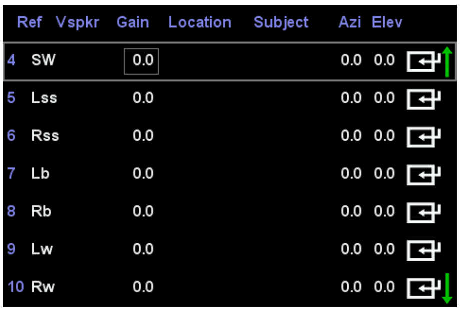
*NOTE: Tables 1, 2 and 3 in **Appendix A: Listening Rooms Loudspeaker Configurations** list all the configurations and speaker labels available for Dolby Atmos, DTS:X and PCM listening rooms.*



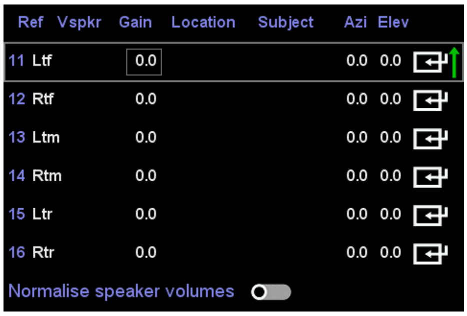
DTS:X listening mode 2.0ch



Reference loudspeaker names for Dolby Atmos 9.1.6ch (menu page 1)



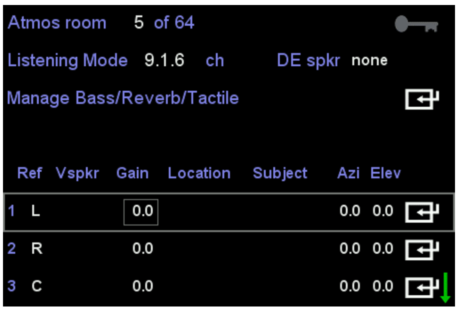
Reference loudspeaker names for Dolby Atmos 9.1.6ch (menu page 2)



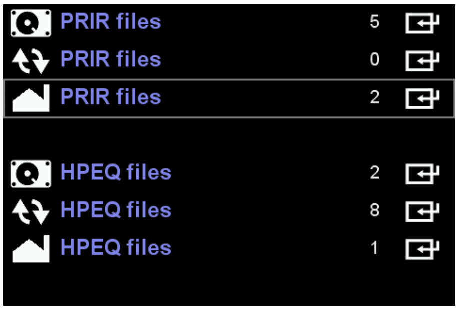
Reference loudspeaker names for Dolby Atmos 9.1.6ch (menu page 3)

10.3 Select virtual speakers for a Listening Mode from a PRIR file

Virtual speakers are the key component of SVS headphone rendering, and essentially recreate over headphones the experience of listening to reference loudspeakers. To select virtual speakers for a particular listening mode, move the selection box to any reference speaker and press the ENTER command key. This brings up the PRIR source selection menu.



To select virtual speakers to populate a Listening room, move the selection box to the first channel and use the ENTER command.



Selecting the PRIR file - choose the source.

PRIRs can be selected from permanent storage, the recycle buffer or from the factory PRIR files.

Select the desired PRIR file.

Any of the PRIR files in each source location can be chosen using the numerical selector at the top of the page. Information that describes each PRIR is also shown to identify and select a particular file. For example, the Location option will show an image of the room (if an image has been attached to the file).

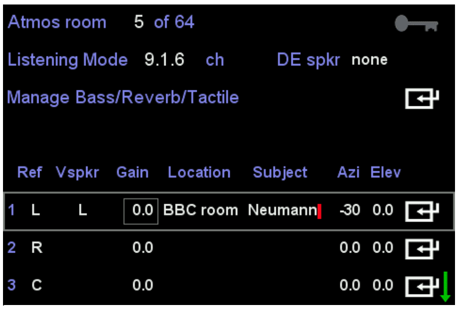
1. *Select the source of the PRIR file using the ENTER command key and move to the PRIR select menu.*
2. *Choose one PRIR file from the source location with the ADJ+ and ADJ- keys, using the information displayed.*
3. *Finally, select either ONE matching speaker from the PRIR, or select ALL matching speakers from the PRIR.*

10.3.1 Select one matching speaker

This menu option gives access to all the individual virtual speakers in the selected PRIR, allowing any single speaker to be selected, for example below, a L(eft) virtual speaker. Once selected the display reverts to the Listening Room configuration menu, and now shows the virtual speaker label, the virtual room name and the azimuth and elevation angles for that chosen virtual speaker. To select more individual virtual speakers, move the selection box to the correct channel in the Listening Room and repeat the procedure. Virtual speakers that are already attached to channels will be over-written by any new selection.

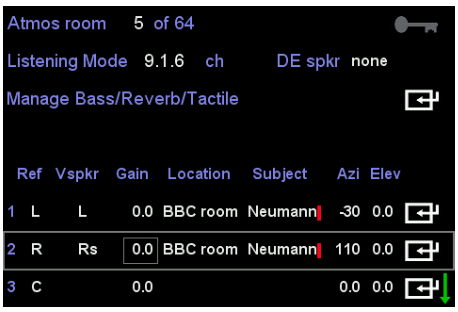
CAUTION: If there is a mismatch in names between the Reference and Virtual names, the A16 will not display the mis-matched speaker in the Speaker Map display if this Listening Room is used for a preset. Furthermore, the mis-matched speaker will not be rendered to SVS headphones, nor will it be output to the line outputs in the AV mode.

Selecting an individual l(eft) speaker from a PRIR file.

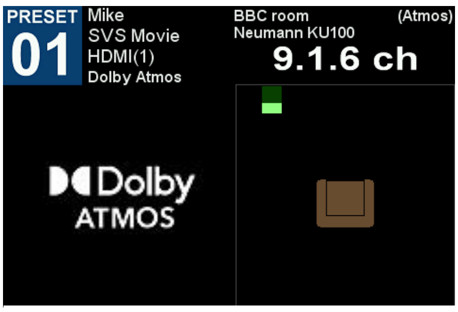


The selected L(eft) virtual speaker, from the PRIR, is inserted into the Listening Room, at the location of the selection box.

Below illustrates a speaker mismatch in channel 2 of a Listening Room, where the Right Surround virtual speaker (Rs) has been selected rather than the Right speaker reference name. This mismatch means that channel 2 is ignored if this listening room is used in a preset.



Mismatched speaker names in a Listening room - the virtual speaker from the PRIR (Rs) does not match the reference speaker name of the Listening Room (R).



Mis-matched speakers in the Listening Room are not displayed in the Speaker Map of the preset, and are not rendered to headphones. Only the matching L(eft) speaker will be heard.

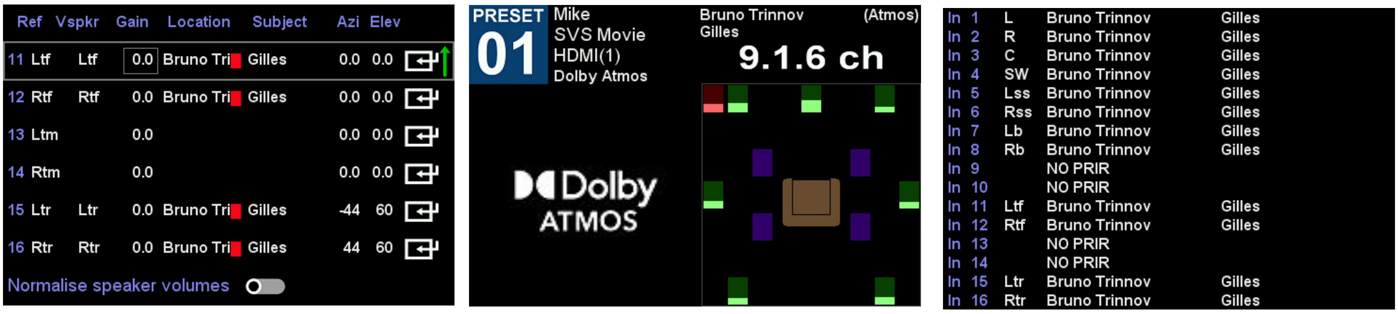
10.3.2 Select all matching speakers

This menu option automatically selects all the virtual speakers in the PRIR that match the reference speaker names in the Listening Room, and then returns the system to the Listening Room configuration menu. Information related to the virtual speakers is also displayed. Below illustrate matching virtual speakers in a 7.1.4ch listening room all taken from a single PRIR measurement. For this format (Atmos 7.1.4ch) some channels are not used and therefore have neither a reference name nor a virtual speaker.

NOTE: Using the **Select all matching speakers** option will over-write all matching virtual speaker names - even if these names have already been selected in a listening room. However, using both selection options allows a Listening Room to be created from virtual speakers selected from multiple PRIR files.

All matched speakers in a 7.1.4ch listening room

If any reference speakers cannot be matched the virtual speaker names will be displayed as blank entries (example below). These unmatched speakers will NOT be displayed in the Preset Speaker Map and will NOT be rendered to the SVS headphone output and will NOT be sent to the AV line outputs for listening through loudspeakers.



Unmatched Left Top Mid and Right Top Mid reference speakers. The PRIR selected did not contain virtual speakers with these names. Hence these speakers are not present in the speaker map nor are they heard in the headphone output.

10.3.3 Normalise speaker volumes

When enabled, this option equalises the loudness of all the virtual speakers in a configured listening room. This may be desirable if the listening room is constructed from virtual speakers from different PRIRs, since the PRIRs may have been measured in different rooms with differing reverberation characteristics. In this routine only the direct HRTF portion of the virtual speakers are used to calculate the normalisation factor for each speaker. When disabled this option retains any naturally occurring inter-speaker level differences between the virtual speakers in a listening room.

10.4 Set Bass Management / Tactile outputs / Stereo mixdown outputs

*Move to: **Home Page** menu: **Listening Rooms** menu: **Atmos or DTS:X or PCM rooms** menu: **Manage Bass/Reverb/Tactile** menu*

This menu sets listening room parameters related to bass management, reverberation control, the generation of tactile outputs for an external 'butt-kicker' device, and the generation of a non-virtualised stereo headphone mix from a selection of the audio input channels.

NOTE: Some of the bass management controls are designed for both SVS headphone listening and AV loudspeaker listening modes - these are labelled **hp/av**. Other controls are for SVS headphone only – these are labelled **hp**.

NOTE: Please refer to **Appendix I: Bass Management**, which illustrates the signal paths for the bass managed signals and their controls.

NOTE: There are different bass management options for Dolby Atmos / DTS:X listening rooms and PCM listening rooms. These differences are reflected in the differing menu options for these rooms.

10.4.1 Dolby Atmos and DTS:X listening rooms

For Dolby Atmos or DTS:X listening rooms there are two different stages of bass management. The first stage (labelled hp/av) is controlled by setting each loudspeaker size and the corner frequency of the global low-pass filter for all the speakers. The bass-managed signals can be output to real (AV loudspeaker mode) or virtual loudspeakers (SVS headphone mode). This bass management stage can also be by-passed, in which case only the LFE signal is sent to the sub-woofer speaker, real or virtual. For SVS headphone rendering there are two alternative bass management modes (labelled hp) that can be engaged – one mode uses a virtual sub-woofer loudspeaker, whilst the second mode, Direct Bass, by-passes the virtual sub-woofer. The Direct Bass mode was developed to circumvent some of the problems associated with measuring and acquiring good virtual sub-woofer speakers and relies on the observation that low-frequency acoustic signals have little or no perceived directionality. In Direct Bass mode the virtual sub-woofer speaker in the listening room is by-passed, and the bass-managed sub-woofer signal is instead fed directly into the left and right headphone outputs.

There are therefore four bass management options for Dolby Atmos and DTS:X listening rooms.

1. *hp/av bass management ON and hp Direct Bass ON. (Appendix I: Bass Management)*
2. *hp/av bass management ON and hp Direct Bass OFF. (Appendix I: Bass Management)*
3. *hp/av bass management OFF and hp Direct Bass ON. (Appendix I: Bass Management)*
4. *hp/av bass management OFF and hp Direct Bass OFF. (Appendix I: Bass Management)*

10.4.2 PCM listening rooms

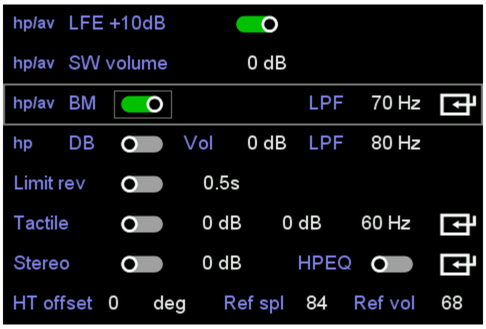
For AV loudspeaker listening using a PCM listening room there is no hp/av bass management stage, since the expectation is that bass management of the line output audio signals will occur after the A16. For SVS headphone rendering using a PCM listening room there are three options. Bass management can be enabled or disabled and, if enabled, the bass-managed low-frequency signal can be rendered through a virtual sub-woofer speaker or sent directly to the headphone outputs using the Direct Bass mode.

There are therefore three bass management options for PCM listening rooms.

1. *Bass management set to Direct Bass. (Appendix I: Bass Management)*
2. *Bass management set to Virtual sub-woofer. (Appendix I: Bass Management)*
3. *Bass management set OFF. (Appendix I: Bass Management)*

10.4.3 Bass Management for Dolby Atmos or DTS:X listening rooms

Please refer to **Appendix I: Bass Management**, for information on the function or each of these parameters.



Manage Bass/Reverb/Tactile menu for Dolby Atmos or DTS:X listening rooms.

10.4.3.1 hp/av LFE +10db

The LFE channel of movie soundtracks is usually reduced by 10dB.Therefore the LFE channel should normally be boosted by 10dB before being sent to a real or virtualised sub-woofer speaker.

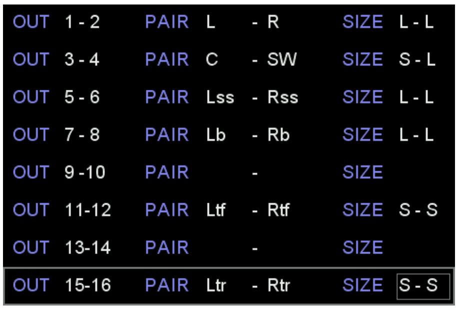
If set ON the LFE input is boosted by 10dB. If set OFF the LFE channel is not boosted.

10.4.3.2 hp/av SW volume

The gain of the bass-managed signal sent to the sub-woofer speaker can be set from -30dB to +10dB.

10.4.3.3 hp/av BM, LPF

Toggles ON or OFF the bass management (BM) control of audio signals (bitstream and PCM) and sets the corner-frequency of the low-pass filter for the bass-management routine (LPF=40Hz to 200Hz in steps of 10Hz). The ENTER command displays the Speaker Size configuration page, which is used to set the size of all the speakers in the listening room.



Setting the sizes of speaker pairs as part of the bass-management configuration of a listening rooom. This room is Dolby Atmos 7.1.4ch with a small centre speaker, and small top speakers.

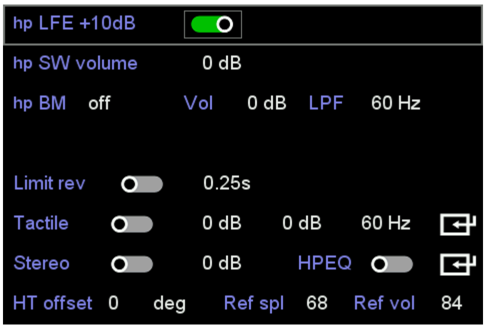
The sizes of each speaker, large or small, and the low-pass filter frequency, are used as parameters in the bass management routine to determine the routing of the low-frequency part of the audio signal passing through each speaker, both in the virtual SVS headphone mode and in the real AV loudspeaker mode. There are several different bass-management routines which have been designed for particular modes of operations, and these are all described more fully in **Appendix I: Bass Management**.

10.4.4 Bass Management for PCM listening rooms

For SVS headphone rendering of PCM audio signals the Realiser A16 provides three options for Bass Management.

1. *Bass management OFF.*
2. *Bass management using a virtual sub-woofer speaker.*
3. *Bass management using Direct Bass mode that by-passes the virtual sub-woofer speaker.*

For AV loudspeaker listening of PCM input signals the Realiser A16 does NOT include options for bass management. It is assumed that bass management will occur after the audio signals are output from the A16.



Manage Bass/Reverb/Tactile menu for PCM listening rooms.

10.4.4.1 hp LFE +10dB

If set ON, the LFE input signal is boosted by 10dB. On movie soundtracks the LFE is normally reduced by 10dB during production, and the signal is then boosted by 10dB in the movie theatres.

10.4.4.2 hp SW volume

Allows the gain of the sub-woofer virtual speaker to be amplified or reduced in the range +12dB to -30dB. A value of 0.0dB implies that the virtual sub-woofer volume is not altered.

10.4.4.3 hp BM

Bass Management (BM) can be set OFF or Virtual or Direct.

If set to Virtual, bass management is enabled, and a virtual sub-woofer speaker is used for the final bass-managed low-frequency signal. (Appendix I: Bass Management.)

If set to Direct, bass management is enabled, but the final bass-managed low-frequency signal is sent directly to the headphone outputs and by-passes the virtual sub-woofer speaker of the PCM listening room. (Appendix I: Bass Management). If set to OFF, bass management is disabled, and the LFE input signal is sent to the virtual sub-woofer speaker of the PCM listening room. (Appendix I: Bass Management.)

10.4.4.4 Vol

If bass management is enabled, Vol controls the output level of the low-frequency summation stage of the bass management routine and is set from +12dB to -30dB.

10.4.4.5 LPF

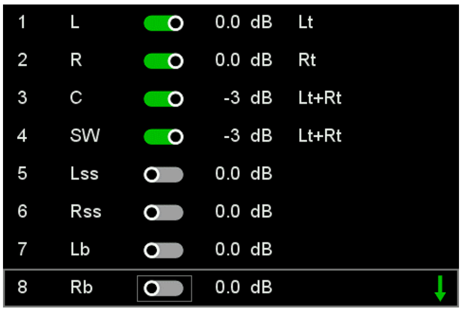
If bass management is enabled, LPF sets the corner frequency of the low-pass filter for the bass management routine, and is set to 60, 80 or 120Hz.

10.4.5 Limit Reverb

If set ON, this limits the reverberation time in the virtual listening room to the set value. If set OFF the reverberation time is set to the maximum value of 0.75ms.

10.4.6 Tactile (mixdown)

If set ON, the volume of each channel of the 2-ch tactile output signal can be reduced or amplified, and the corner frequency of the 4^{th} order, low-pass, IIR filter can be set (60/80/120Hz). To specify which channels will be used to create the tactile output use the ENTER command to move to the Tactile Mixdown menu.



Tactile output (menu page 1)

Tactile output (menu page 2)

In the above example, the low-pass filtered components of the L, R, C and SW channels of a 9.1.6ch signal are being mixed, at varying gains, to the 2-ch tactile outputs, Lt and Rt. Some of the low pass filtered signals are being sent to a single output channel (either Lt or Rt) and some are being sent to both channels equally (Lt+Rt). The remaining surround and height channels of the 9.1.6ch signal are not enabled and are therefore not contributing to the tactile output.

10.4.7 Stereo (mixdown)

This is explained in the Stereo Bypass chapter.

10.4.8 HT Offset

It is possible for the virtual loudspeaker layout to appear slightly rotated because of errors in the look angles taken up during a PRIR measurement, or because the head tracker mounted on the headphones is slightly twisted. **HT offset** is a head tracker azimuth offset entry with a range of +/- 10 degrees. If, for example, the soundstage exhibits a +5-degree rotation, then the HT offset would be set to -5 degrees to cancel out the apparent rotation. Note that the offset angle is independent of the HT function and is effective even if the HT has been disabled or disconnected.

10.4.9 Reference Levels (Ref SPL, Ref vol)

Ref vol and **Ref SPL** are values that set the reference level for a room and are explained in the Listening at Reference Level over Headphones chapter. **Ref vol** is the A16 volume setting that is required for the listening room to attain the **Ref SPL** level (assuming the headphone and the analog gain are the same when the reference level was first calibrated). Both values can be generated automatically by the Reference Level calibration routine. They can also be edited manually in this configuration menu. Ref vol and Ref SPL are also used as baseline volumes for the **Tri Ref** function found in the preset menu.

10.5 *General notes for Configuring a Listening Room*

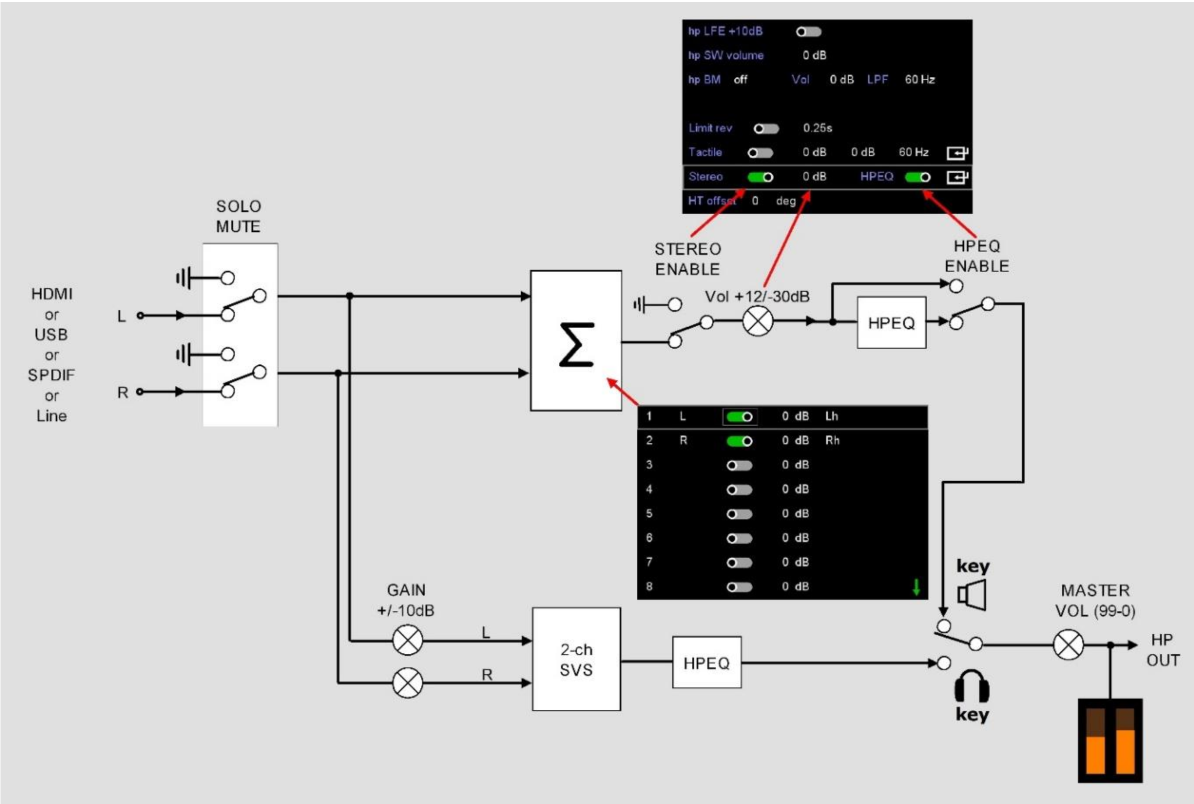
NOTE 1: *To prevent unintentional changes being made to the configuration of the Listening Room, it is recommended that the room be locked before being saved.*

NOTE 2: *To save the Listening Room configuration use the BACK key from page 1 of the Listening Room configuration menu. A progress bar will become visible in the top right-hand corner of the menu screen. This will return the display to the Listening Rooms menu.*

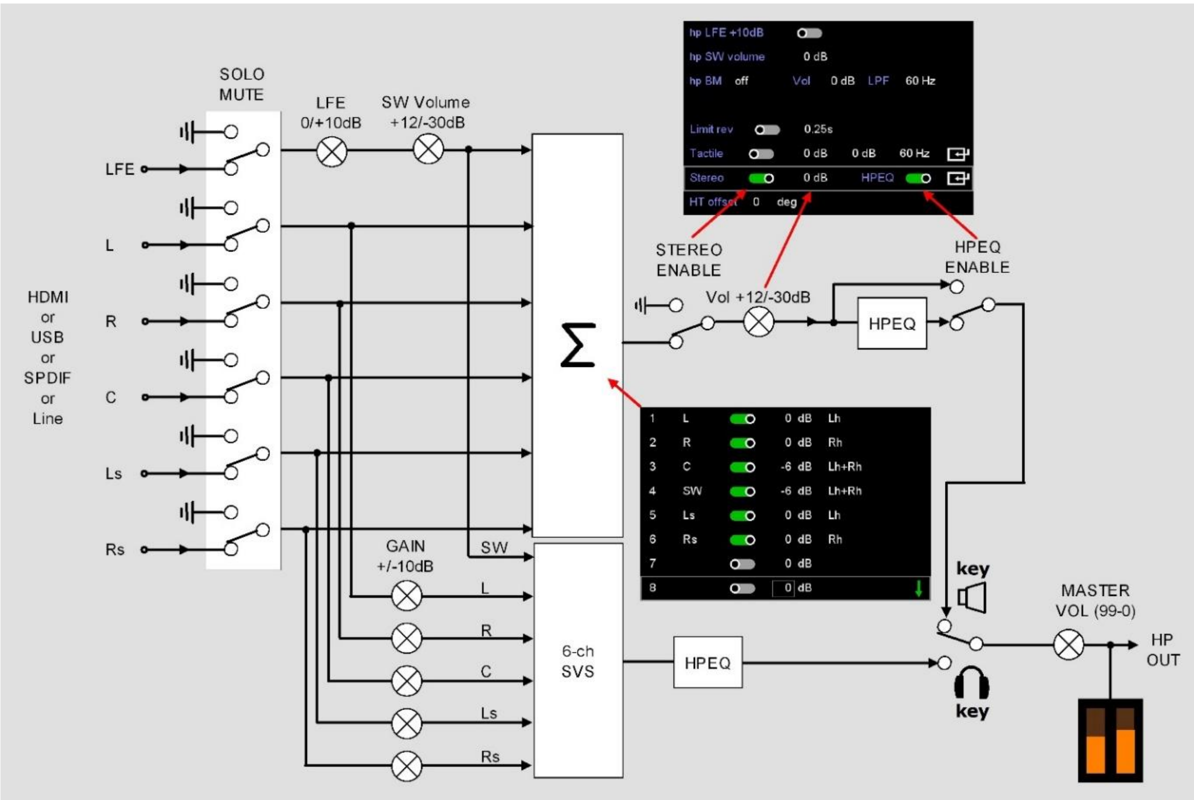
NOTE 3: *After the Listening Room has been saved it may be selected as one of the three listening rooms for a preset.*

11 Stereo Bypass

When AV mode is disabled in the active User A preset, the headphone key and speaker key switch can alternatively switch between SVS headphone mode and Stereo Mix Down mode, respectively. However, to switch to Stereo mode it must first be enabled in the listening room currently loaded and the corresponding mix down table configured appropriately. The stereo mix down path can be used to pass regular stereo content straight to the headphones, or it can be used to create a stereo mix down of multi-channel content, as shown in the schematic diagrams below.



Example of using the A16 to listen to regular stereo audio without virtualization but retaining the HPEQ



Example of using the A16 to listen to a 5.1ch stereo mix-down without virtualization but retaining the HPEQ

In the Stereo Mix Down mode, the user can choose to filter the stereo audio using the HPEQ specified in the active User A preset, prior to being output to the headphone.

11.1 *Steps* to listening to stereo audio using a 2ch Stereo Bypass method

1) In the current User A preset, disable the AV mode.

2) In the current User A preset, select the listening room you intend to use in stereo mix down mode (e.g., PCM room 4) and reload.

3) In the current User A preset, enter the PCM Audio management

4) Ensure both HDMI and SPDIF Upmixers are set to 'Direct'

PCM Source	Upmixer	Input format
HDMI	Direct	auto
USB	Direct	auto
Line	Direct	auto
SPDIF	Direct	2ch

5) Go to the home page, enter the Listening Room menu and select 'Loading only PCM'.

6) Enter the PCM room editor and select room 4.

7) Go down and enter the Configure menu

8) Enable Stereo mode

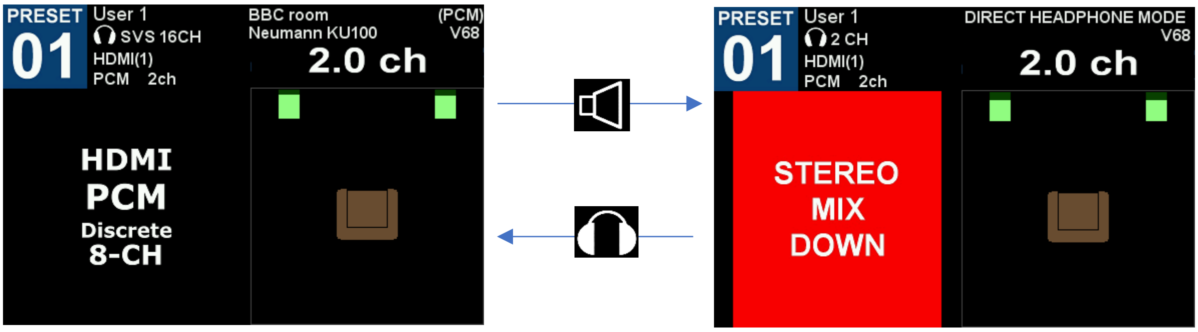
9) Enter the down mix editor and ensure L goes straight through to Lh and R to Rh.

1	L	on	0 dB	Lh
2	R	on	0 dB	Rh
3		off	0 dB	

10) Press the BACK key repeatedly to return to the home page. PCM room 4 should automatically reload as you leave the listening room editor. Otherwise reload the current User A preset manually.

11) Set your audio input source in the home page and start playback.

12) Initially the SVS rendered version of the stereo will be heard over the headphones. Press the speaker key on the remote to hear the raw stereo.

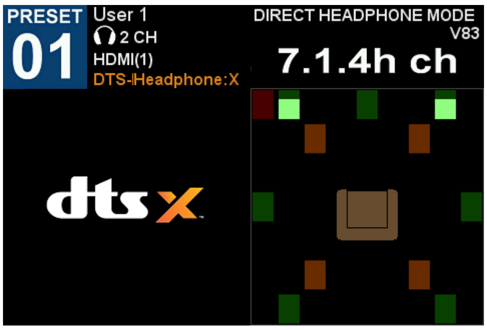


13) If it is desired to filter the stereo mix down audio using the preset HPEQ, then enable this feature in the PCM listening room editor.

14) If you need to adjust the volume of the stereo mix down audio without altering the SVS rendered volume, adjust the volume in the PCM room 4 Configure menu.

11.2 *Automatic Stereo Bypass for DTS headphone:X*

On reception of a DTS:X bitstream that flags binaural Headphone:X content the A16 will automatically engage the stereo bypass. Since Headphone:X content is already virtualised, the stereo bypass ensures that it is not virtualised for a second time using the SVS rendering.



12 Measuring a new PRIR in a sound room using the synchronous (ALL) method

There are two methods for measuring PRIRs – the Synchronous (ALL) method and the Asynchronous (ASYC) method. Choosing which method to use will often depend on how easily the A16 can be connected to loudspeaker amplifiers in the listening room that is being measured.

Method 1: Synchronous (ALL) – described in this chapter

In the synchronous (ALL) method the A16 generates the sine wave sweeps that are used to measure the individual loudspeakers. These sweeps are output from the 16-ch analogue output connectors and requires that the A16 be connected directly to the loudspeaker amplifiers.

This method is described as synchronous since the number of audio samples recorded through the binaural microphones exactly matches the number of audio samples generated within the A16 – the generator and recorder are sample-rate locked.

Method 2: Asynchronous (Async) – described in chapter 14

In the asynchronous method the A16 does not generate the sine wave sweeps, and therefore the A16 does not need to be connected to the loudspeaker amplifiers. Instead the sweep signals are pre-recorded and played back through the loudspeakers in the listening room.

There are two main types of pre-recorded sweep signals, bitstream and PCM. Bitstream test signals are used in consumer sound rooms where the loudspeakers are connected to an AV receiver capable of decoding Dolby Atmos bitstreams delivered via a DVD/BD player, or other source of bitstream audio. PCM test signals are used in professional sound rooms where the loudspeakers are connected to an audio editing workstation, or some other source of PCM audio signals.

This method is described as asynchronous since the number of audio samples recorded may not match the number of audio samples in the pre-recorded audio test signals – the generator and recorder are not sample-rate locked.

Choosing the Sync (ALL) or Async method

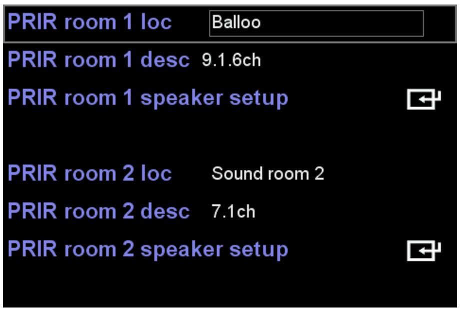
The choice of PRIR measurement method will usually depend on whether the A16 can be readily connected to the loudspeaker amplifiers in the sound room. Another important consideration is that the Sync (ALL) method allows the A16 to automatically bypass the SVS headphone rendering algorithm, using the tilt detector in the headphone-mounted head-tracker. This permits the user to easily A/B compare the measured virtual loudspeakers, rendered over the headphones, with the real loudspeakers in the listening room. With the Async method this switching (if required) must be done external to the A16.

12.1 Configure the PRIR Sound room

*Move to: **Home Page** menu: **Setting**s menu: **PRIR Sound Rooms** menu:*

PRIRs are measured using real loudspeakers in a real sound room, and the PRIR Sound Rooms menu allows configuration of the A16 for measuring PRIRs. The main part of the configuration process involves describing all of the loudspeakers being measured. This includes naming (or labelling) each speaker, setting its spatial position, and setting its size. A descriptive name of the sound room, and the overall format of the speaker arrangement in the room, should also be added.

Two separate rooms can be configured, and either one can be selected during the actual PRIR measurement process.



PRIR Sound Rooms menu.

12.1.1 Edit the PRIR room 1 location (loc)

The name (or location) of the room being measured can be changed or edited. This name will become part of the PRIR file and used to identify the PRIR.

12.1.2 Edit the PRIR room 1 description (desc)

The description of the speaker format in the room being measured can also be edited. The name of the format also becomes part of the PRIR file and used to identify the PRIR.

Tables 1, 2 and 3 in Appendix A list the format names (Modes) used informally for some standard arrangements of Dolby Atmos, DTS:X and PCM rooms, but other descriptions may be more accurate for non-standard arrangements of loudspeakers – for example Ambisonics.

12.1.3 Configure the PRIR room 1 speaker setup

Move to: ***Home Page*** *menu:* ***Settings*** *men**u: PRIR Sound Rooms*** *menu:* ***PRIR room 1 speaker setup*** *menu*

This option moves to a new menu that configures the analogue output channels of the A16 by assigning a speaker label, and other information, to each output channel number, for the purpose of making a PRIR measurement. The information is intended to accurately describe each of the loudspeakers being measured in the room, and this information becomes part of the PRIR file.

Ch	Spkr	Azi	Elev	Path	Gain	Size	UF	FF	hpf
1	L	-30	0.0	1.50	1.0	L			
2	R	30	0.0	1.50	1.0	L			
3	C	0.0	0.0	1.50	1.0	L			
4	SW	0.0	0.0	1.50	1.0	L			
5	Lw	-60	0.0	1.50	1.0	L			
6	Rw	60	0.0	1.50	1.0	L			
7	Lss	-90	0.0	1.50	1.0	L			

PRIR speaker setup menu: page 1: configuring each speaker in the room.

Ch	Spkr	Azi	Elev	Path	Gain	Size	UF	FF	hpf
1	L	-30	0.0	1.50	1.0	L			
2	R	30	0.0	1.50	1.0	L			
3	C	0.0	0.0	1.50	1.0	L			
4	Ls	-100	0.0	1.50	1.0	L			
5	Ls	-100	0.0	1.50	1.0	L			
6	Ltf	-44	60	1.50	1.0	S	Y	L	
7	Rtf	44	60	1.50	1.0	S	Y	R	

An example PRIR room configuration: 5.0.2ch format with two up-firing speakers simulating Ltf and Rtf speakers.

12.1.3.1 Ch

Channels 1 to 16 refer to the multichannel line outputs on the back panel of the Realiser A16.

12.1.3.2 Spkr

All the speaker labels that can be used are listed in Table 4 in Appendix B. Each channel that is being measured should be labelled with an appropriate speaker label.

> *Care should be taken when choosing speaker names (labels). The use of some speaker names in a PRIR may limit the ability of the virtual speakers to become associated with a particular format – in effect the virtual speaker in a PRIR may become locked out of a particular format due to its name. These restrictions only apply for bitstream and PCM audio that is input through HDMI, and are listed in Table 4 in Appendix B.*
>
> *For example, if a virtual speaker is labelled as Lh (Left height) it can be seen from Table 1 Appendix A, that this name is not used in any Dolby Atmos listening mode that the Realiser A16 is capable of rendering. Therefore, a virtual speaker labelled as Lh in any PRIR file cannot be matched to a speaker in any configuration of a Dolby Atmos Listening Room, and can only be matched in an appropriately configured DTS:X or PCM Listening Room.*
>
> *On the other hand, if a virtual speaker is labelled Ltf (Left top front) this name is used by Dolby Atmos, DTS:X and PCM, and therefore can be matched to appropriately configured Dolby Atmos, DTS:X or PCM Listening Rooms.*

12.1.3.3 Azi

The azimuth angle of the loudspeaker with respect to the listening position. Negative angles are used for left side loudspeakers, positive for right side speakers. This is an approximate angle used to describe the azimuth position of the speaker. It is not used in the SVS virtualisation algorithm.

12.1.3.4 Elev

Elev is the elevation angle of the loudspeaker, with respect to ear height, from the listening position. Positive angles are used for speakers above ear level, negative for below ear level. This is an approximate angle used to describe the elevation of the speaker. It is not used in the SVS virtualisation algorithm.

12.1.3.5 Path and Gain and Size

Path is the distance in metres to the loudspeaker from the listening position. This is not currently used and does not need to be changed.

Gain describes the relative output level of each loudspeaker during calibration. This is not currently used and should not be changed.

Size can be set to L(arge), for full bandwidth speakers, or S(mall) for speakers that cannot reproduce low frequencies.

12.1.3.6 UF, FF and hpf

UF refers to up-firing speakers designed to simulate 'height' or 'top' speakers by reflecting sound from the ceiling. The speaker's name should match the spatial direction of the simulated (or reflected) sound – for example a speaker labelled as 'Ltf' (Left top front) and described as UF means that this up-firing speaker simulates a speaker in the 'Ltf' position.

Each **UF** speaker is placed on top of an **FF** speaker (front-firing), and the name of this matching FF speaker can be added here to describe the physical position of the UF speaker.

Finally, the low-frequency limit of the up-firing speaker being measured is set in **lpf**.

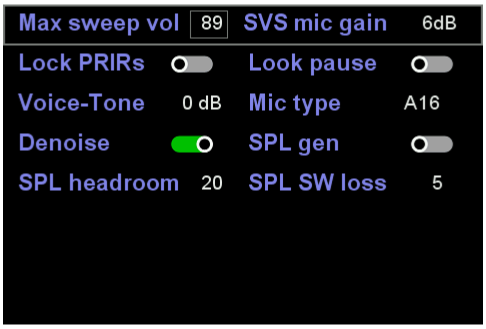
The example PRIR sound room above shows a room configured as 5.0.2ch. Channels 6 and 7 are up-firing speakers and are placed on top of the L(eft) and R(ight) front-firing speakers, respectively. The audio signals from channels 6 and 7 appear to come from the Left and Right top front positions, and so these names are the labels for the speakers.

The UF, FF and hpf descriptors are not currently used.

12.1.4 Configure the measurement settings

The measurement settings menu allows some parameters to be changed that impact on PRIR and HPEQ measurements.

*Move to: **Home Page** menu: **Settings** menu: **System** menu: **Measurement settings** menu*



Measurement settings menu.

12.1.4.1 Max sweep Vol

Sets the maximum output volume of the sine wave sweeps from the A16 during the PRIR measurement. The maximum output volume may need to be increased if the binaural microphone signal level is too low during the loudspeaker calibration routine.

12.1.4.2 SVS Mic gain

Sets the gain of the binaural microphone during the PRIR and HPEQ measurements. The mic gain may need to be increased to boost the binaural microphone signals if the recorded signals are too low in volume. This would normally become apparent during the loudspeaker calibration routine.

12.1.4.3 Lock PRIRs

Locked PRIRs can only be used by the A16 originally used during the measurement.

> *When set ON a measured PRIR is locked to the host A16.*
> *When set OFF a measured PRIR can be used by all A16 units.*

12.1.4.4 Look Pause

When Look pause is enabled the PRIR measurement will pause at the start of each look angle. This can give the subject more time to align their head with the current look angle orientation. Press the enter key to proceed.

12.1.4.5 Voice-Tone

This adjusts the volume of the voice prompts and the pilot tones by +/-12dB.

12.1.4.6 Mic Type

Switches the SVS microphone recording circuit to compensate for the differing frequency response characteristics of the A8 and A16 microphones.

12.1.4.7 Denoise

Denoise is a signal processing algorithm that reduces the effects of room background noise on the PRIR. Enable Denoise unless the sound room has a very low ambient noise level.

12.1.5 Connect the A16 to the loudspeakers in the sound room

1. *Switch off the power amplifiers of the loudspeakers in the sound room before making any connections.*
2. *Connect the 16-ch Line Outputs of the A16 to the loudspeakers according to the configuration of the PRIR Sound Room.*

3. Switch the power amplifiers of the loudspeakers back on.

12.1.6 Connect the SVS microphones to the A16

1. *Check that the Left and Right microphones are inserted in the L and R MIC sockets in the front panel of the A16.*
2. *To check that the microphones are working correctly run* ***Listen to Mics on HPB*** *in audio settings menu.*

12.1.7 Insert the SVS microphones in the ear canal

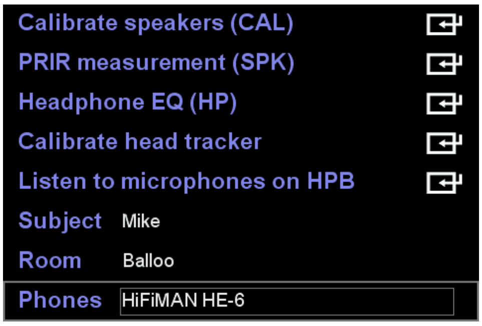
1. *Choose a foam insert size that allows the binaural microphones to be inserted such that the top of each microphone is flush with the entrance of the ear canal.*

12.2 Configure and run the loudspeaker calibration routine

The loudspeaker calibration routine is an option in the Apps menu.

Move to: ***Home Page*** *menu:* ***Apps*** *menu:*

12.2.1 Set the subject name, room name and headphone name



Apps menu.

12.2.1.1 Subject

Select a subject from the list of names in the Settings: Users menu using the ADJ+ and ADJ- keys. The subject name will become part of the PRIR measurement.

12.2.1.2 Room

Select a PRIR room from the two names set in Settings: PRIR Sound Rooms: PRIR room 1 loc and PRIR room 2 loc using the ADJ+ and ADJ- keys. The room name will become part of the PRIR file.

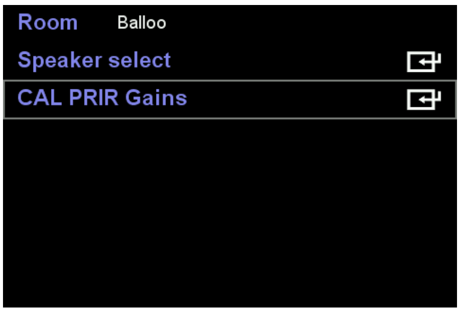
12.2.1.3 Phones

Select a headphone from the list of names set in the Settings: Headphones menu using the ADJ+ and ADJ- keys.

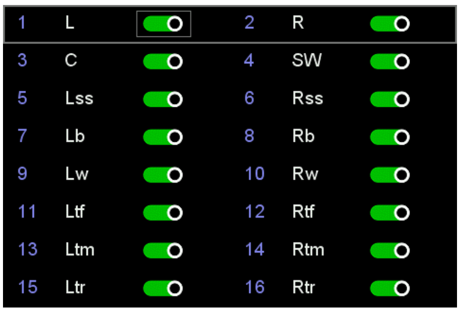
12.2.2 Set the loudspeakers to be calibrated

1. Move to the Calibrate speakers (CAL) menu.

Move to: ***Home Page*** *menu:* ***Apps*** *menu:* ***Calibrate speakers (CAL)*** *menu:*



Calibrate speakers (CAL) menu



Speaker select menu

2. Select the speakers to be calibrated by entering the Speaker select menu option.

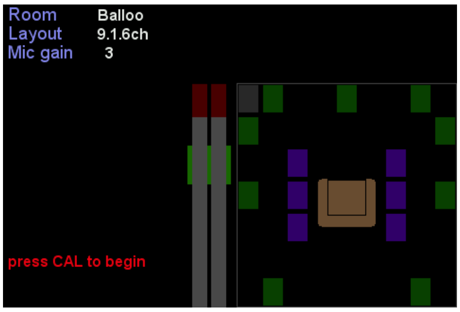
*Move to: **Home Page** menu: **Apps** menu: **Calibrate speakers (CAL)** menu: **Speaker select** menu*

The listed speakers will match the configuration of the selected PRIR Sound Room and all speakers will be switched ON by default. Switch OFF any speaker that does not require calibration, then return to the previous menu (BACK key).

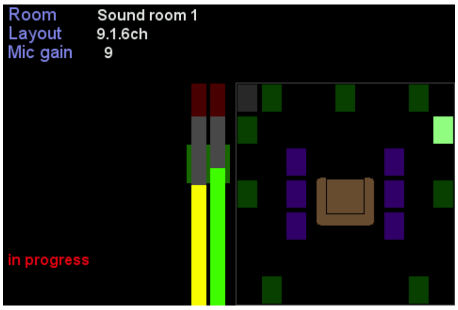
12.2.3 Run the Speaker Calibration routine

1. Move to the CAL PRIR Gains menu which displays the speaker map of all the speakers to be calibrated.

*Move to: **Home Page** menu: **Apps** menu: **Calibrate speakers (CAL)** menu: **CAL PRIR Gains** menu*



CAL PRIR Gains menu



Calibration routine showing the calibration levels for the right wide speaker. Note that the recorded right ear binaural signal is greater than the left ear signal.

2. Insert binaural microphones correctly.

3. Sit in the central listening location, looking straight ahead.

4. Press CAL on the remote control to begin the calibration routine.

One or more short sine wave sweeps will be heard from each speaker in turn. These sweeps are output at increasing volumes to try to record sufficiently high levels in the two SVS microphones - these levels are displayed in the graphic and change colour according to the level – yellow (low), green (good) and red (high - clipped).

The SVS microphones record two signals for each speaker being calibrated, one signal for the left ear and one for the right ear. In general, these two signals will be at quite different levels, the levels being dependent on the position of the loudspeaker with respect to the listener. For example, for loudspeakers which are at approximately +/- 90 degrees to the listener, i.e. left and right surround, the microphone in the ear facing the loudspeaker will record a higher signal than the microphone in the ear facing away from the loudspeaker.

The calibration routine tries to ensure that at least one of the microphones signals is recording a good (green) level by changing the level of gain for each loudspeaker being calibrated.

A sticky clip indicator will show if the calibration level has clipped during the calibration. This can be cancelled using the CANCEL key on the remote.

***Note:** The calibration sweep for the SW (subwoofer) speakers is done at the end of the routine, after all the full bandwidth speakers have been calibrated. Therefore, the user should wait until the routine has finished before moving BACK to the Apps menu.*

5. When the calibration routine is finished press BACK to save the gain levels for each loudspeaker.

If the calibration clips or the microphone levels are persistently too high, try one or both of the following options:

- *Reduce the global external gain of the loudspeaker amplifiers (using an external volume control).*

  and/or

- *Reduce the Mic gain (Home Page menu: Settings menu: System menu: Measurement settings menu: SVS Mic gain)*

  then run the calibration routine again.

If the microphone levels are persistently too low, try one or more of the following options:

- *Increase the Mic gain (Home Page menu: Settings menu: System menu: Measurement settings menu: SVS Mic gain)*

  and/or

- *Increase the maximum sweep level (Home Page menu: Settings menu: System menu: Measurement settings menu: Max sweep Vol:)*

and/or

- *Increase the global external gain of the loudspeaker amplifiers (using an external volume control)*

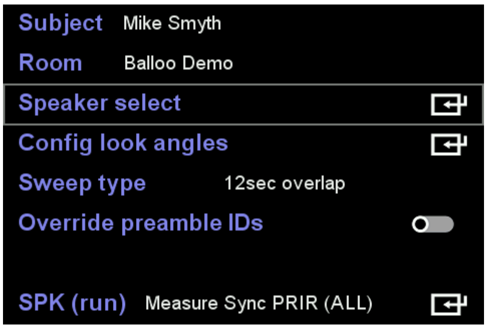
then run the calibration routine again.

If the SVS mic levels are sufficiently good (at least one green channel for each calibrated loudspeaker) then the speaker calibration for the PRIR measurement is complete. Use the back key to save the calibration settings and return to the Apps menu.

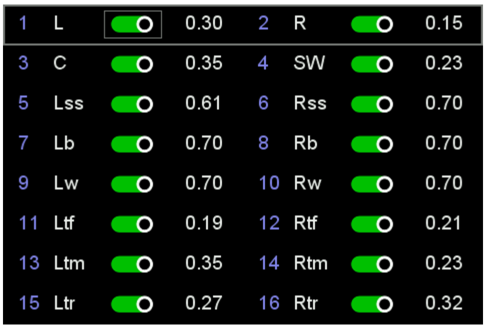
Note: Calibration gains are written to flash memory on completion of any CAL measurement. The same gains are recalled each time the PRIR routine is activated even if the A16 has since been powered down. Hence speaker calibrations do not necessarily need to be run each time a PRIR is to be measured if nothing significant has changed with the sound system since the last calibration.

12.3 Configure and run the PRIR Measurement routine

Move to: ***Home Page*** *menu:* ***Apps*** *menu:* ***PRIR measurement (SPK)*** *menu:*



PRIR measurement (SPK) menu



Speaker select menu

12.3.1 Subject and Room names

These names are set in the previous (Apps) menu.

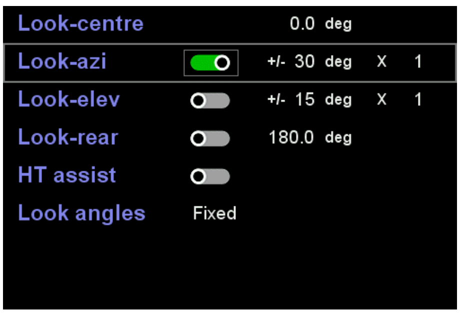
12.3.2 Select the loudspeakers to be measured

The Speaker select menu displays the speakers that have just been calibrated, and shows the gain level that has been applied to the calibration signal in order to maximise the recording SNR from the SVS microphones. These gain levels will be used during the PRIR measurement but can be increased or decreased in the Speaker select menu page using the ADJ+ and ADJ- keys if necessary. It is instructive to check and compare the gain levels of the speakers that have just been calibrated. Apart from the SW speaker, all the other speakers should have roughly the same gain if the loudspeakers in the room have been previously calibrated to have the same apparent volume at the listener sweet spot (in the example screen shot this is clearly not the case). Select which loudspeakers to measure using the ADJ+ or ADJ- keys. Use the BACK key to save this selection and return to the previous menu.

12.3.3 Configure the look angles for the PRIR measurements

A 'look angle' refers to the orientation of the head during the binaural measurements. For each head orientation, or look angle, all the selected loudspeakers will output a swept sine wave, in sequence, and these will be recorded by the binaural microphones. The number of look angles to use, and the angular span between them, is set by the user. In general, the more look angles used and the smaller the angular span between them, the greater will be the range and accuracy of head-tracking during playback. However, since each look angle requires more measurement time, the user can set the number of look angles that will be used, given that many applications do not require full 360-degree headtracking.

Move to: ***Home Page*** *menu:* ***Apps*** *menu:* ***PRIR measurement (SPK)*** *menu:* ***Config look angles*** *menu*



Config look angles menu

12.3.3.1 Look-azi

Look-azi is the azimuth, or rotational, angle of the head. Zero degrees is looking dead-centre. If Look-azi is enabled, the angular span of the look-angle and the number of look-angles must be set. The angular span can be set from +/-5 degrees to +/-60 degrees, in steps of 5 degrees, and the number of look-angles can be set from 1 to 11 but the maximum number depends on the size of the angular span. Some examples of these settings are:

- *If Look-azi is switched OFF, then only look-centre is used for the PRIR measurement (0.0 degrees). In this mode no interpolation is possible between different head orientations, and therefore SVS head-tracking during playback is not feasible.*
- *If look-azi is switched ON and the angular span is set to +/- 30 degrees and one look angle is selected for the PRIR measurement, then headtracking is possible for head movements of +/- 30 degrees during playback.*
- *If look-azi is switched ON, the angular span is set to +/- 30 degrees, five look angles are selected for the PRIR measurement, and look-rear is switched ON, then head-tracking is possible for head movements of 360 degrees during playback.*

The maximum number of look angles that can be set is 11, at an angular span of +/-15 deg. The factory default PRIR files (BBC, Surrey Rooms) have this configuration.

12.3.3.2 Look-elev

Look-elev is the elevation, or tilt, angle of the head.

NOTE: Elevation angles are not currently used.

12.3.3.3 Look-rear

Look-rear enables a measurement for a look-angle of 180 degrees. It complements the Look-centre look-angle of 0 degrees and allows for 360-degree headtracking. In general, it should only be enabled if 360-degree headtracking is required and a large number of look-angles has been selected (e.g. 3 angles @ +/- 45 degrees or 5 angles @ +/- 30 degrees etc)

12.3.3.4 HT assist

When **HT assist** is enabled, the **look angles** mode is limited to **fixed**. HT assist is a real time 2-axis pilot tone fed to all A16 line outputs that allows the subject to align their head orientation with those look angles programmed in the Look-Azi and Look-Elev menus. HT assist uses the A16 head-tracker mounted on a headband which is then fitted to the subject’s head to determine the actual head angles during the PRIR measurement.

Example Head band support on a dummy head during HT assist mode

Three different tone frequencies are output during HT assist depending on the pilot axis. The pilot tone for the azimuth axis outputs a mid-frequency tone to the speakers, the output level of which steadily reduces as the head tracker angle gets closer to the desired azimuth look angle. This reduction in volume begins once the head tracker is within +/-20 degrees of the desired look azimuth angle and is muted once it comes within 1 degree of the look angle. The pilot tone for the elevation axis outputs a high frequency if the head tracker elevation angle is higher than the desired look elevation angle, or a low frequency if the head tracker elevation is too low. As with the azimuth pilot tone, both high and low pilot tones decrease in volume as the head tracker gets closer to the desired look elevation angle.

Different combinations of the pilot assist tones, can be selected while in the main PRIR window using the left and right arrow keys and is indicated next to Pilot Axis (**horz**, **vert** and **both**).

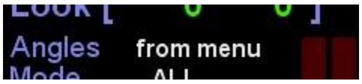
- *Selecting pilot axis **horz** causes the tone to track only azimuth head movements.*
- *Selecting pilot axis **vert** causes the tone to track only vertical head movements.*
- *Selecting pilot axis **both** causes the tone to initially track only azimuth head movements but once this angle is within +/-5 degrees of the desired azimuth angle, then the vertical tone will also be heard.*

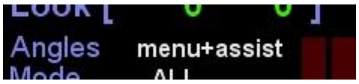
Pilot axis mode is selected using the left/right arrow key when running the PRIR measurement

For either axis, once the head tracker angle moves within 1 degree of the desired look angle, the pilot tone for that axis will mute. Hence, when the head orientation is within 1 degree of the desired look angle no pilot tones will be heard. The overall volume of the pilot assist tones can also be adjusted while in the main window using +/- HP-A volume control on the remote. When the pilot axis is set to **both or Azi,** the PRIR measurement will begin automatically once the subjects head moves within 2.5 degrees of the desired look angle and does not move outside that boundary for a period of 1 second. Irrespective of the mode, the PRIR measurement can also be manually started at any time by pressing the enter key. When the pilot axis is set to either **vert**, the PRIR measurement will not begin until the enter key is pressed.

12.3.3.5 Look angles

When Look angles are **Fixed** the head tracker is not used and the look angles inserted into the PRIR file are those defined by Look-azi, Look-elev and Look-rear fields in the Config Look Angles menu page - indicated in the main PRIR display by **Angles from menu.** If HT Assist is enabled at the same time, the look angles are still defined by the menu, but the head tracker runs in conjunction with the pilot tones to help the subject properly align their head - indicated in the main PRIR display by **Angles from menu + assist.**





When Look angles are **Free** the head tracker angle sampled just before the first sweep in each look angle is inserted into the measured PRIR file. To remind the user of this fact the main PRIR display flags **Angles from HT**.

12.3.4 Set the sweep type

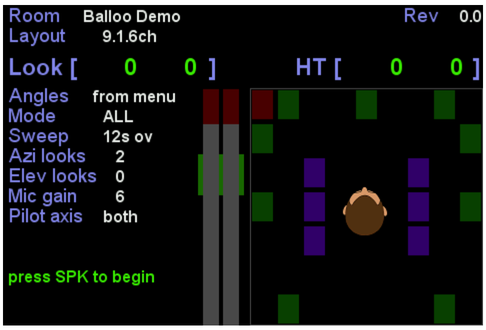
*Move to: **Home Page** menu: **Apps** menu: **PRIR measurement (SPK)** menu*

The sine sweep can be set to a length of 4 or 12 seconds, with or without an overlap. The overlap option decreases the measurement time significantly. The 12s overlapped sweep is recommended.

12.3.5 Load and run the PRIR measurement routine

*Move to: **Home Page** menu: **Apps** menu: **PRIR measurement (SPK)** menu: **SPK** menu*

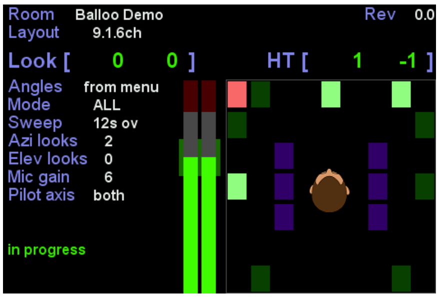
Entering the SPK menu displays the PRIR measurement speaker map.

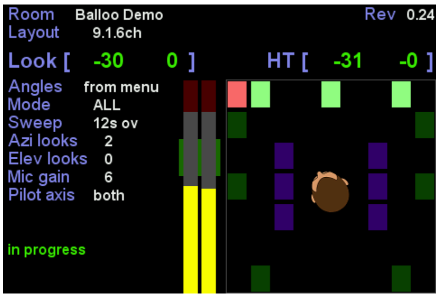


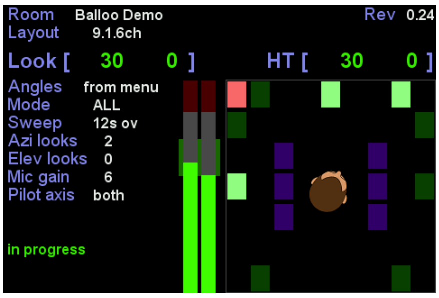
SPK menu showing the speaker map of the PRIR to be measured

NOTE: *Ensure that the SVS microphones are correctly inserted in each ear canal (left mic in the left ear) and that the subject is in the sweet spot of the room facing towards the centre speaker.*

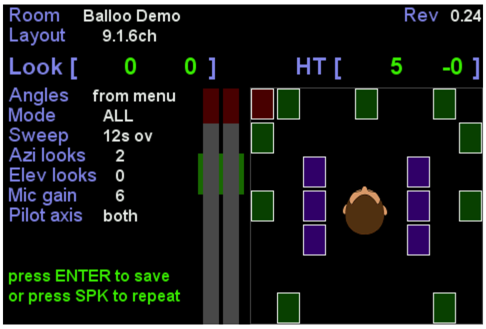
To begin the PRIR measurement procedure press the SPK key on the remote control and wait for the verbal instructions. During the measurement, the screen will display the speakers that are being measured, and the signal levels being recorded by the microphones. Moreover, the Look [0 0] angles will change to indicate the current look angle (or head orientation) the subject should adopt (the format is [azimuth, elevation]) and if a User A head tracker is connected, the HT [0 0] angles will follow the head tracker heading, irrespective of whether HT assist is enabled or Look angles is set to Free. In the 3-look example below the HT heading follows that of the desired 'Look' angle but the head tracker was unnecessary since Look angles were **Fixed**. If HT assist is enabled or Look angle is **Free**, it would be necessary to mount the head tracker onto the subject's head using the supplied headband in addition to fitting the SVS microphones.







Example PRIR measurement over 3 fixed look angles (0 degrees, -30 degree and +30 degrees)



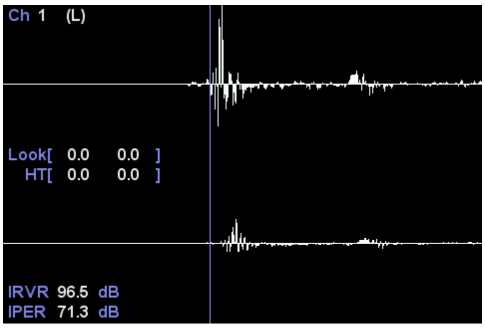
End of the PRIR measurement showing a white border around each measured

12.3.6 Assessing the Impulse Response Quality between looks

A PRIR viewer can be invoked to provide a means of visually assessing the quality of a PRIR measurement as it is in the process of being measured, or at the end of a measurement just before it is written back as a PRIR file.

Most PRIR measurements consist of a series of sweep measurements made at several discrete look angles. By allowing the measurement to pause between looks, it is possible to view the pre and post onset regions of the impulse response captured for each speaker. This allows the subject to abandon the measurement early in the process if the impulse data is deemed flawed and not have to wait until the entire measurement has finished. If either HT-Assist or Pause-between-looks are enabled, the PRIR measurement will be suspended at the end of each look-angle measurement.

Press the ENTER key to activate the viewer. The first screen will show the onset portion of the binaural PRIR for the speaker connected to ch1 output. The top waveform is the left ear recording and the bottom waveform is the right ear recording. Pressing the UP or DOWN arrow keys moves between channels. Pressing the LEFT or RIGHT arrow keys moves between looks. If you are viewing after only the first look angle measurement, then these keys will have no effect. To exit the PRIR viewer press ENTER again. A more complete explanation is shown on our YouTube channel.



PRIR viewer to assess IR quality

12.3.7 Saving the PRIR measurement

To save the PRIR measurement press the ENTER key

The current PRIR measurement is always saved to the first slot of the PRIR recycle buffer, and other older measurements in the recycle buffer are shuffled up one slot - the measurement stored in slot 16 is overwritten and lost. Therefore, PRIR measurements in the recycle buffer should be moved to the permanent internal storage or to an external SD card.

A progress bar is displayed while the measured PRIR file is being saved.

To exit from the PRIR measurement procedure press the BACK key.

13 Measuring personalised headphone HPEQ filters

13.1 Overview of HPEQ methods supported by the A16

Headphone EQ filters are used to try to flatten the frequency response of headphones when placed over a listener's ears. Because individual's have different ear shapes and ear canals, they need individualised headphone EQ filters. Up to four different EQ filters can be created within a single HPEQ file, and any one of these four filters can be selected within a preset configuration.

13.1.1 AutoEQ filter measurement using binaural microphones

For normal headphones, the standard procedure is to measure EQ automatically, using the SVS microphones inserted in the listener's ears. It is advantageous if this is done immediately after any PRIR measurement since the microphones would generally be in the same location in the ear canal. This procedure creates two filters; an autoEQ filter from the measured data, and a flatEQ filter which is generated as flat. AutoEQ requires no setup, other than fitting the micophones, and no user intervention and is therefore recommended for new users of the A16.

13.1.2 FlatEQ filter generation for IEM-type headphones

For in-ear type headphones (IEM-type), where it is not feasible to use binaural microphones to measure the EQ automatically, there is an option to generate an HPEQ file containing a flat EQ filter (flatEQ). This procedure also creates an autoEQ filter and a flatEQ filter, but both filters are generated as flat. In either case, once the HPEQ file has been generated, and the HPEQ file has been saved, the autoEQ or flatEQ filters may be selected for use within a preset.

13.1.3 Manual EQ modification of either the autoEQ or flatEQ filters

However, it is also possible to manually create two new filters, based on the autoEQ or flatEQ filters, using two different routines. The manual EQ stage can modify the filter created automatically (autoEQ) which, due to the position of the binaural measurement microphones, is not able to take account of any ear canal resonances. The manual EQ stage can also modify the flatEQ filter for IEM-type headphones.

13.1.3.1 Manual EQ using equal loudness pink noise – suitable for all headphones including IEM-type headphones

In the first manual routine, manLOUD, the loudness of a sub-band noise signal is adjusted to try to equalise the loudness between each band and thereby remove any peaks or notches. This routine does not require that the headphones be removed from the head whilst making a comparison and is therefore suitable for use with all types of headphones, including IEMs. Optionally a PRIR may be selected for the manLOUD method, in which case the HRTF part of the impulse response of the selected virtual speaker(s) are convolved with the filtered sub-band signals to externalise the signal. If the PRIR is not selected the filtered sub-band signals are heard without any virtualisation – in the centre of the head.

13.1.3.2 Manual EQ using an external reference loudspeaker – suitable for normal headphones

In the second manual routine, manSPKR, the loudness of sub-band noise signals heard through a virtual loudspeaker are compared to a real loudspeaker, and the virtual sub-band levels are adjusted to match those of the real loudspeaker reference. Since this technique requires that the headphones be removed from the head during each sub-band comparison between the virtual and real loudspeakers, it is not suited to IEM-type headphones. For normal headphones, the head-tracker facilitates the A/B switching between the real and virtual speakers. In the manSPKR method a PRIR must be selected, and the entire impulse response of the selected virtual speakers are convolved with the filtered sub-band signals.

NOTE: In the manual routines the base filter can be either the autoEQ or flatEQ filter, and new filters are created, either manLOUD or manSPKR. Once the manual EQ routines are completed the generated filters are added to the HPEQ file and may be selected as the HPEQ within a preset, thus facilitating the creation of filters for both headphones and IEMs.

13.2 Configure the A16 for an automatic HPEQ measurement of normal headphones.

13.2.1 Connect the SVS microphones to the A16

Insert the SVS microphones into the subject's ear canals – ensuring that the microphones in the left and right ears are connected to the L(eft) and R(ight) microphone inputs of the A16.

13.2.2 Set the headphone A output gain

Set the headphone gain of User A appropriately for the headphones to be measured (set to LOW if unsure)

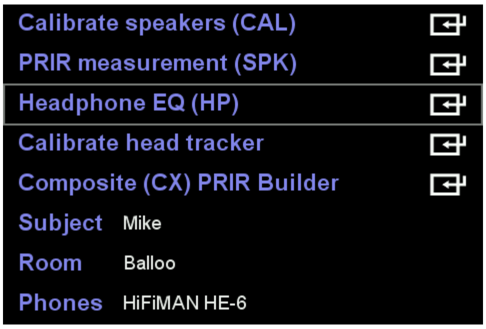
13.2.3 Connect headphones to the User A HP jack

The HPEQ is only measured at the Headphone A output. Once connected to the A output place the headphones correctly over the ears, taking care not to disturb the binaural microphones in the ear canals.

13.3 *Configure the HPEQ options*

13.3.1 Set the Subject name and Headphone name

*Move to: **Home Page** menu: **Apps** menu*



Apps menu

13.3.1.1 Select a Subject name

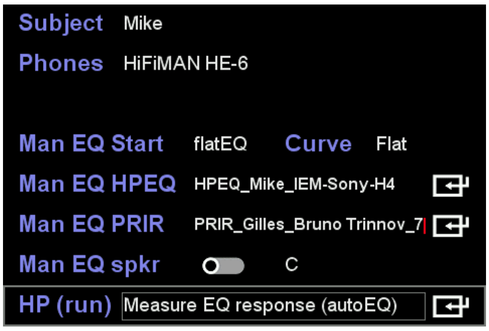
This name is selected from the list created in the Users Menu (Home Page menu: Settings menu: Users menu).

13.3.1.2 Select the Phones name

This name is selected from the list created in the Headphones Menu (Home Page menu: Settings menu: Headphones menu).

13.3.2 Set the HPEQ measurement options

*Move to: **Home Page** menu: **Apps** menu: **Headphone EQ (HP)** menu (Apps menu)*



Headphone EQ (HP) menu

13.3.2.1 Subject, Phones

These names are descriptive and become part of the HPEQ file name. They are set in the previous menu.

13.3.2.2 Man EQ Start, Curve, Man EQ HPEQ, Man EQ PRIR, Man EQ spkr

These options are not relevant and are ignored for the autoEQ measurement procedure.

13.3.2.3 HP (run)

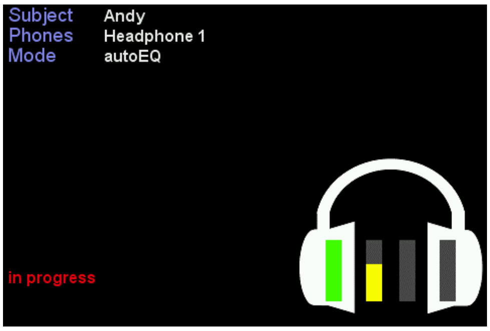
Set to **Measure EQ response (autoEQ)** and press the ENTER command. This loads and runs the autoEQ measurement routine.

NOTE: Ensure that the SVS microphones are inserted in the correct ears, and that the left headphone cup is on the left ear.

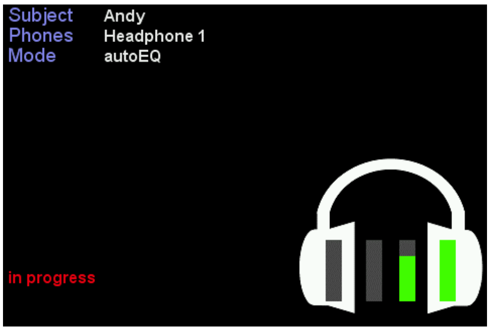
NOTE: Press the HP key to begin the automatic headphone EQ procedure. The entire measurement should be conduced in silence.

Automatic EQ measurement (autoEQ).

An initial calibration routine will send short sine sweeps, at increasing volumes, from the left and right drivers of the headphones to find an adequate microphone level. Immediately following the calibration, the measurement routine begins, and a long sine sweep is output first from the left headphone driver and then from the right headphone driver. The graphic icon shows the level of the sine sweep output from the headphone (outer meters) and the recorded level in the microphones (inner meters).



Automatic headphone EQ measurement showing the headphone signal and the recorded level in the left binaural microphone.



Automatic headphone EQ measurement showing the headphone signal and the recorded level in the right binaural microphone.

During the calibration and measurement routines observe the recorded levels of the microphones.

If the microphone level is too low (peaking at yellow):

1. *Change the GAIN switch of the HP output of User A to a higher setting.*

   and/or

2. *Increase the binaural microphone gain*

   (Home Page menu: Settings menu: System Menu: Measurement settings menu: SVS mic gain)

If the microphone level is too high (peaking at red):

1. *Change the GAIN switch of the HP output of User A to a lower setting.*

   and/or

2 *Decrease the binaural microphone gain*

   (Home Page menu: Settings menu: System Menu: Measurement settings menu: SVS mic gain)

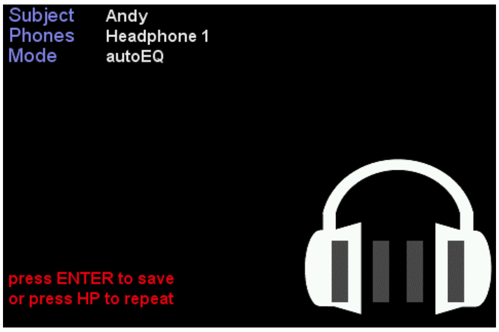
13.4 *Saving the HPEQ measurement*

At the end of the automatic headphone EQ procedure the user is prompted to either save the measured HPEQ fie, or to repeat the measurement. The HPEQ filter is saved to the first slot of the HPEQ recycle buffer, and older measurements are shuffled up one slot. The measurement in the 16th slot is overwritten and lost.

To save the HPEQ measurement press the ENTER key

Press the BACK key to exit from the automatic HPEQ measurement window and return to the HPEQ menu

It will be noticed that the HPEQ file contains the autoEQ filter created from the measured data, and also a flatEQ filter.

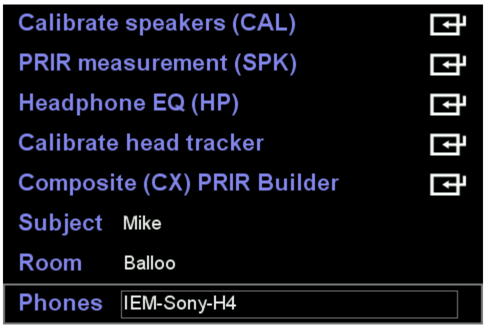


Saving the HPEQ measurement.

13.5 *Configure the A16 to generate a flat HPEQ filter.*

13.5.1 *Set the Subject name and Phones name*

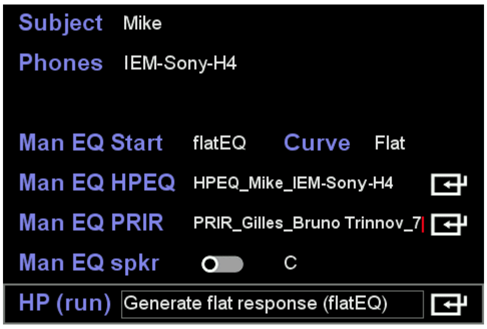
Move to: ***Home Page*** *menu:* ***Apps*** *menu*



Selecting the Subject and Headphones names.

13.5.2 *Set the HPEQ measurement options*

Move to: ***Home Page*** *menu:* ***Apps*** *menu:* ***Headphone EQ (HP)*** *menu*



Setting the Headphone EQ (HP) menu options.

13.5.2.1 Subject, Phones

These names are set in the previous menu.

13.5.2.2 Man EQ Start, Curve, Man EQ HPEQ, Man EQ PRIR and Man EQ Spkr

These options are not relevant and are ignored for the flatEQ measurement procedure.

13.5.2.3 HP (run)

Set to **Generate flat response (flatEQ)** and press the ENTER command to generate the filter and HPEQ file. No actual measurement is necessary.

NOTE: *The HPEQ file created using the Generate flat response (flatEQ) option includes an autoEQ filter and a flatEQ filter. Both filters are flat.*

13.6 *Manual HPEQ adjustment using an external loudspeaker as reference.*

13.6.1 *Set the HPEQ measurement options*

Move to: ***Home Page*** *menu:* ***Apps*** *menu:* ***Headphone EQ (HP)*** *menu*

Configuration of HPEQ parameters for manual adjustment of an autoEQ filter, by comparing a virtual centre speaker from a measured PRIR against a real external loudspeaker as reference. The selected sub-band pink noise signals have a flat response curve.

13.6.1.1 Subject, Phones

These names are set in the previous menu but are not relevant for the manual EQ stage.

13.6.1.2 Man EQ Start

Selects the base filter to be adjusted. The options are autoEQ or flatEQ.

13.6.1.3 Curve

The frequency response curve of the band-limited pink noise excitation signal. The options are flat, equal-loudness-80 or equal-loudness-20.

13.6.1.4 Man EQ HPEQ

Selects the HPEQ file that contains the filter to be manually adjusted. The ENTER command moves to the HPEQ file select menus allowing the desired HPEQ file to be selected. Note that the Man EQ Start option selects the actual filter from within this HPEQ file – select either the autoEQ or flatEQ filter.

13.6.1.5 Man EQ PRIR

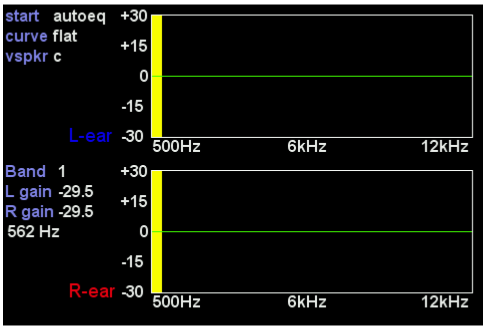
Selects the PRIR file from which the virtual speaker(s) are selected that will be used to compare to the real external speaker(s). The ENTER command moves to the PRIR file select menus allowing the desired PRIR to be selected. Normally this PRIR will have been measured using the same external reference speaker and room – in other words the comparison should be between an external speaker in a room and the measured PRIR of this speaker in the same location in the same room.

13.6.1.6 Man EQ Spkr

Selects the virtual speaker(s) that will be used during the A/B comparison with the external speaker(s). The options are centre, left + right, left + centre + right. The toggle switch option is ignored since this manual EQ mode always requires at least one virtual speaker.

13.6.1.7 HP (run)

*Set to **Compare to speaker (manSPKR)** mode and use the ENTER command to move to the multiband EQ page.*

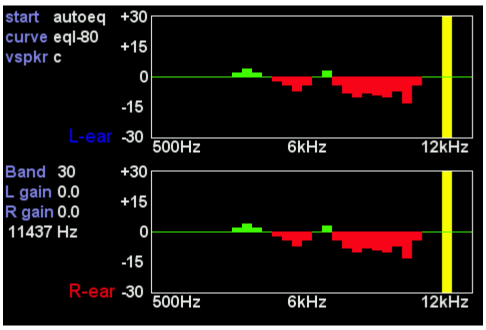


Multiband EQ page for manual EQ using an external loudspeaker.

This page displays two multiband EQ graphs, for the left and right headphone outputs, and some information relating to the measurement being conducted. There are 32 bands covering DC up to 12kHz. Band 1 is used as the reference signal and no adjustments can be made to this band, apart from the output volume which can be adjusted using the ADJ+ and ADJ- keys. This sets the reference volume for the whole test.

The selected band is yellow when heard through headphones (the virtual speaker(s)) and changes to grey when switched to the line outputs (the real speaker(s)). Switching is done automatically with the head-tracker – as the headphones are removed and tilted forward, the head-top detects the tilt angle and switches the sub-band noise signal to the line-outputs connected to the real loudspeaker(s). When the headphones are placed on the head again, the signal is switched back to the virtual speaker(s).

1. *Use the left and right arrow keys to move up and down the frequency bands of the EQ graphs.*
2. *Use the ADJ+ and ADJ- keys to adjust the level of both the L-ear and R-ear headphone signals in each band.*
3. Use the Vol A rocker key to adjust the level of the individual L-ear signal, and Vol-B for the individual R-ear signal.
4. Iterate over all the sub-bands after making any changes – changes in one band will affect bands on either side.
5. Save the changes using the HP key.



Manual EQ changes to an autoEQ filter using an external speaker as reference to create a manSPKR filter.

NOTE: The CANCEL key will remove ALL changes and create a flat filter.

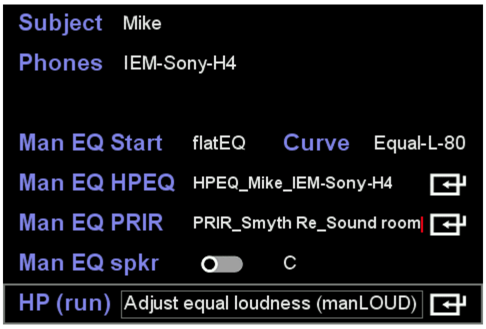
NOTE: SAVE the filter back to the HPEQ file using the HP key.

Once the manSPKR filter has been saved it will be visible as an optional filter in the HPEQ file and can be selected as the HPEQ filter within a preset.

13.7 *Manual HPEQ adjustment using an equal loudness curve.*

13.7.1 Set the HPEQ measurement options

*Move to: **Home Page** menu: **Apps** menu: **Headphone EQ (HP**) menu*



Configuration of the HPEQ menu options for manual EQ using the equal loudness technique. The flatEQ filter of the selected HPEQ file will be adjusted using an eqal-loudness-80 curve for the sub-band noise signals, and the headphone signals will NOT be virtualised with a PRIR.

13.7.1.1 Subject, Phones

These names are set in the previous menu but are not relevant to the manual EQ stage.

13.7.1.2 Man EQ Start

Selects the base filter within the HPEQ file to be adjusted. The options are autoEQ or flatEQ.

13.7.1.3 Curve

Selects the frequency response curve for the sub-band noise excitation signals. The options are flat, equal-loudness-20, equal-loudness-80.

13.7.1.4 Man EQ HPEQ

Selects the HPEQ file that contains the filter to be manually adjusted. The ENTER command moves to the HPEQ file select menus, allowing the desired HPEQ file to be selected. Note that the Man EQ Start option selects the actual filter from within this HPEQ file – either the autoEQ or flatEQ filter.

13.7.1.5 Man EQ PRIR

Optionally selects the PRIR file from which the virtual speaker(s) are selected. The HRTF of the impulse responses of the virtual speakers are convolved with the filtered sub-band noise signals to externalise the headphone signal. The ENTER command moves to the PRIR file select menus, allowing the desired PRIR to be selected.

13.7.1.6 Man EQ Spkr

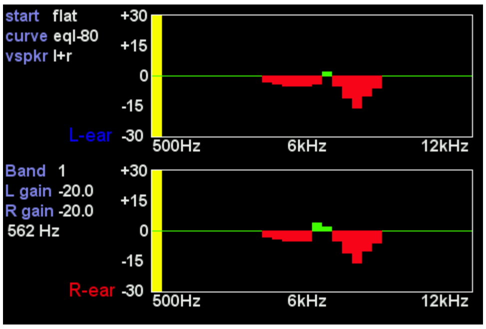
When toggled ON this enables the selection of a PRIR file and selects the named virtual speakers. The virtual speaker options are centre, left + right, left + centre + right. Only the HRTF part of the virtual speaker impulse responses is convolved. This gives some spatiality to the signal but removes any room response. If toggled OFF the sub-band noise signal is played directly to the left and right headphone outputs, and the PRIR file and virtual speaker(s) selection are both ignored.

13.7.1.7 HP (run)

*Set to **Adjust equal loudness (manLOUD)** and use the ENTER command to move to the multiband EQ stage.*

This page displays the same two-channel multiband EQ graph, for the left and right headphone outputs, and some information relating to the measurement being conducted. The selected band is yellow but there is no external speaker and therefore no switching of the signal is necessary.

1. Use the left and right arrow keys to move up and down the frequency bands of the EQ graphs.
2. Use the ADJ+ and ADJ- keys to adjust the level of both the L-ear and R-ear headphone signals in each band.
3. Use the Vol A rocker key to adjust the level of the individual L-ear signal, and Vol-B for the R-ear signal.
4. Iterate over all the sub-bands after making any changes – changes in one band will affect bands on either side.
5. Save the changes using the HP key.



Manual EQ changes to a flatEQ filter, using equal loudness sub-band noise signals.

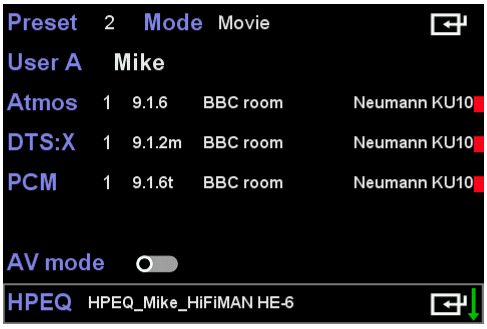
NOTE: The objective is to remove any peaks or troughs between the sub-bands, using Band 1 as the reference level.

NOTE: The CANCEL key will remove ALL changes and create a flat filter.

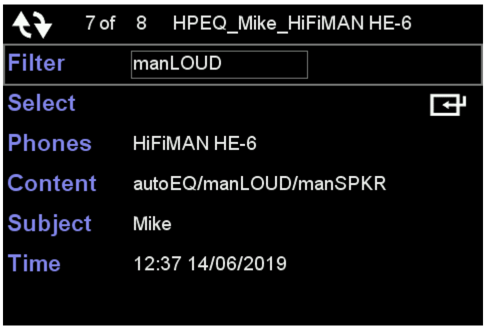
NOTE: SAVE the filter back to the HPEQ file using the HP key.

Once the manLOUD filter has been saved it will be visible as an optional filter in the HPEQ file and can be selected as the HPEQ filter within a preset.

File information showing the contents of an HPEQ file in the recycle buffer. Three filters are available – autoEQ, manLOUD and manSPKR.



Selecting an HPEQ file to configure Preset 2.



Selecting the manLOUD filter of the selected HPEQ file for Preset 2.

Confirmation of the selected HPEQ filter from the Speaker Map display informational page of the active preset.

14 Measuring a PRIR using the Async Method

As with the ALL method, the Asynchronous PRIR (ASYNC) method measures the room impulse responses using the SVS microphones placed in the subject's ears. However, unlike the ALL method, the sinewave sweeps for each look angle are sourced from an external player such as a DVD player or a DAW (DVD/DAW). This method can be convenient since the A16 does not need to connect into the playback chain of the sound room being measured. However, it does make the measurement process more laborious and drawn out, requiring signalling to be inserted into the audio tracks to inform the A16, among other things, which speaker is being measured and when to stop and start the deconvolution processing.

14.1 Theory of Operation

The Async PRIR measurement sequence is very similar to the ALL PRIR method. Asynchronous multi-channel tracks are generated for each look angle where the track comprises a series of sweeps, one for each loudspeaker in the sound room being measured. Each Async measurement comprise several such look tracks that measure all the loudspeakers in the sound room for each look angle. For example, Async PRIR measurements often comprise three look tracks, a look-centre track, a look-left-30-degrees track and a look-right-30-degrees track. The subject undertaking the PRIR measurement will look straight ahead while the look-centre track is played, then turn their head 30 degrees to the left before playing the look-left-30-degrees track and lastly turn their head 30 degrees to the right prior to running the look-right track. In each case the head angle inserted into the PRIR file will either be the look angle embedded in the preamble track or, if the subject is wearing the head tracker, optionally the head tracker angle. To measure a 5.1ch sound room at 3 look angles for example, a 6-track file containing a sweep in each channel, would be played 3 times, once for each look angle. Below is a typical 6-track PCM Wave file that illustrates the sequencing of the sweeps for each speaker in a 5.1ch room.

Example of sinewave sweeps arranged to measure each speaker in a 5.1ch sound room sequentially in time.

Before the A16 can conduct the measurement, it needs to know, a) the approximate playback volume, b) the type of test is being conducted, c) the optimal recording levels and d) the exact sampling frequency of the player (DVD/DAW). This information is acquired by the A16 by having the DVD/DAW play several short setup tracks prior to starting the main PRIR measurement.

14.1.1 Async Noise

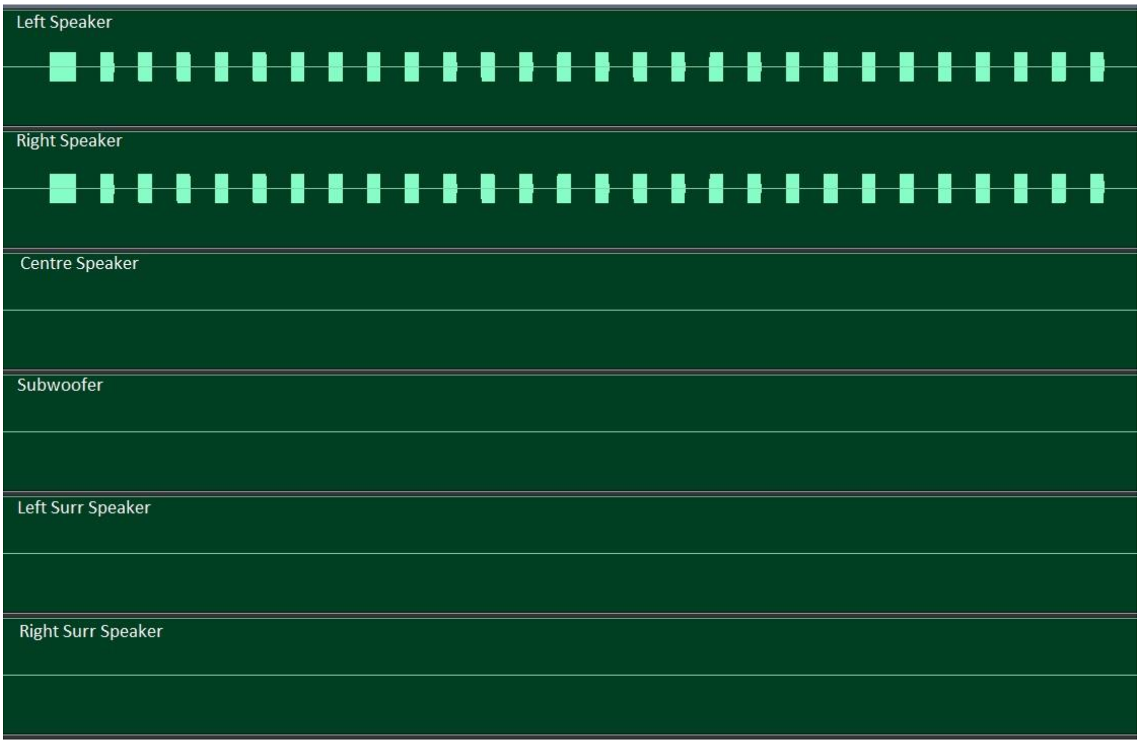
The first track that is played by the DVD/DAW is a 60 second white noise signal recorded at a specific signal level. The user adjusts the speaker playback volume level to achieve a certain level at the A16 SVS microphones. This is a first step at optimising the A16 record levels, finer optimisation is conducted using the Level Calibration track.

14.1.2 Async Preamble track

To speed up the configuration of the A16, an Asynchronous Preamble audio track signals to the realiser the format of the pending measurement using DTMF burst tones. Preamble signalling informs the A16 of:

1) the look angles
2) sweep type
3) speaker names
4) speaker format.

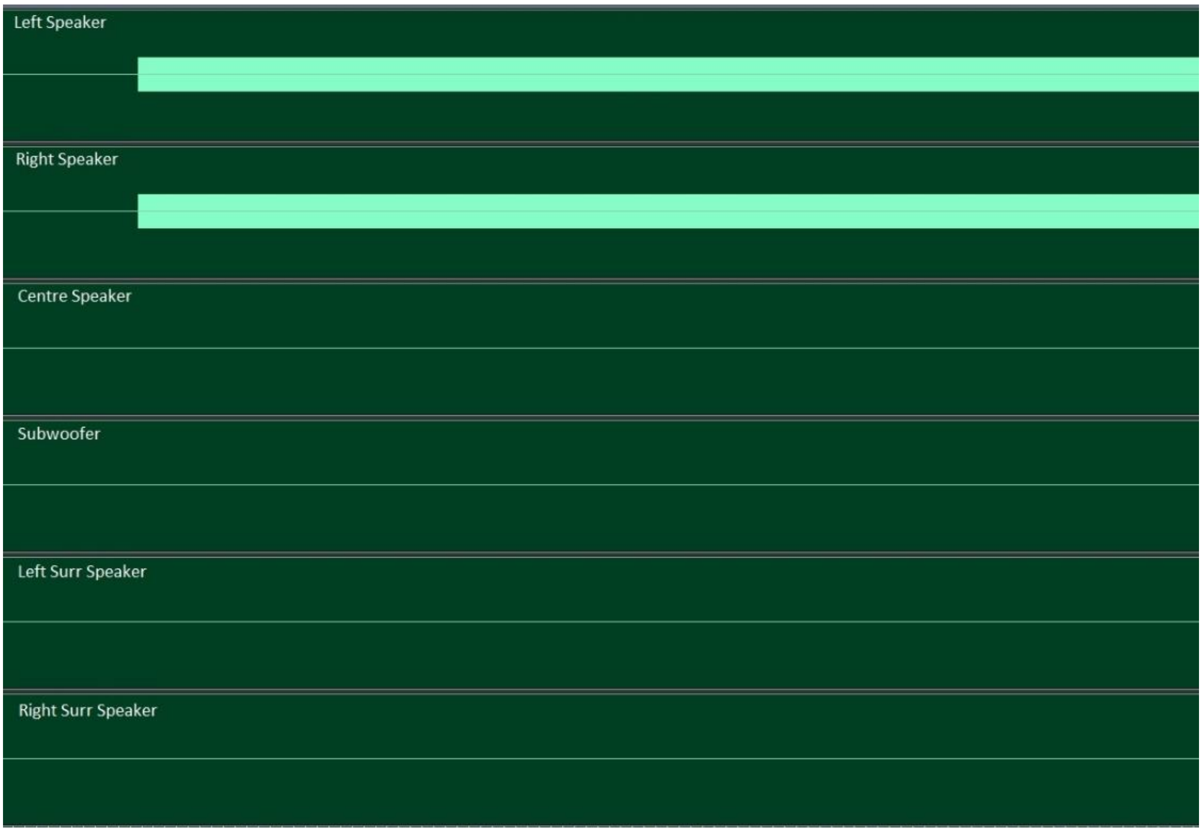
Information such as the subject's name and the location of the room are instead lifted from the selected PRIR sound room in the A16 Apps home page.



Example of preamble signalling in a 5.1ch measurement (only the first two channels carry the DTMF tones)

14.1.3 Async PLL track

The term 'Asynchronous' refers to the fact that the sinewave sweep sample clock (in the DVD player or DAW) is not locked to the sample clock running the deconvolution processing within the A16. Although both will be running at approximately 48kHz, often their actual frequency will be different by 30 to 40 parts per million (ppm). To reduce this difference a Phase Lock Loop (PLL) track is played by the DVD/DAW which comprises a fixed tone over a 60 second duration. As the tone is played, the A16 analyses the phase of the tone heard over the SVS microphones and automatically adjusts its own sample clock frequency to lock to that of the DVD/DAW player. Typically, this PLL process will reduce the difference between the playback and record systems to below 2ppm (less than 1 sample slip every 10 seconds @48kHz).



Example of PLL 60s tone in a 5.1ch measurement (only the first two channels carry the tone)

14.1.4 Async Level calibration

Level calibration is simply a track that outputs a short sinewave sweep to each speaker at the same volume level as the sinewave sweeps used for the main deconvolution. As the track plays, the A16 analyses the levels recorded by the microphones and adjust the microphone gain to maximise the recorded sweep SNR. This track is run after the noise track volume adjustment.

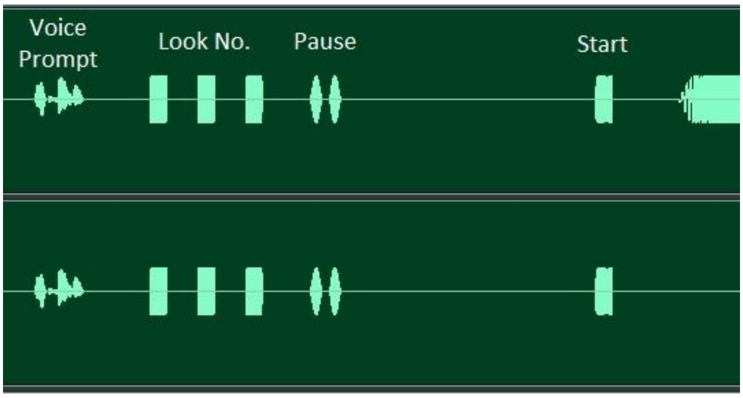
Example of a 5.1ch level calibration track

14.1.5 Async Sweeps for each look angle

Look tracks are generated for each look angle the subject intends to measure and must position their head appropriately before playing the tracks. Look angle measurements can be undertaken in any order and can be repeated as often as necessary with the exception that the right-most look track (or look down of the right-most look track) will complete the measurement.

Example of a 5.1ch look track

Prior to the actual sinewave sweeps, each look track embeds a Voice Prompt that reminds the subject which head angle is expected, some DTMF bursts (Look No.) that identifies the look angle, two short 2.2kHz tone bursts (Pause) used to signal an appropriate time to pause the playback and finally a DTMF burst (Start) that triggers the deconvolution process in the A16.



Look track preamble prior to start of sinewave sweeps

14.1.5.1 Using the Head Tracker to capture the subject Look angle

By default the A16 will use the look angles embedded in the preamble track for insertion into the measured PRIR file. A more accurate alternative is for the subject to wear the head tracker while carrying out the measurement, thereby inserting the actual head angles into the PRIR.

14.1.5.2 Using Head Track Assist

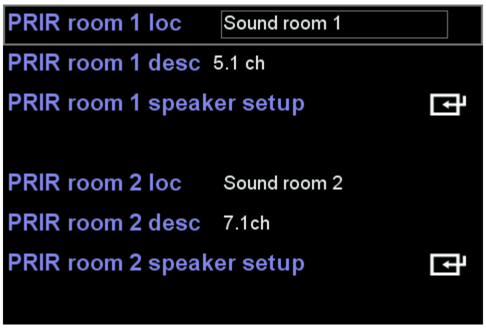
A pilot tone assist of the ALL method is also available in the ASYNC method. This can be used to improve the accuracy of the look angles taken up by the subject when using the head tracker by allowing the subject to know when their head orientation matches the look angles signalled in the preamble track.

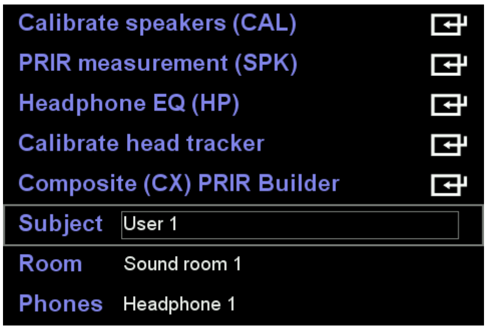
14.2 ASYNC method configuration

Before you begin an Async PRIR measurement, one of the PRIR sound rooms, under the settings menu, should be configured to match the real sound room being measured. This sound room, as well as the subject name, should then be selected in 'Room' and 'Subject' respectively in the Apps home page. On completion of a new Async PRIR measurement information from the selected user and sound room is copied into the PRIR file as follows.

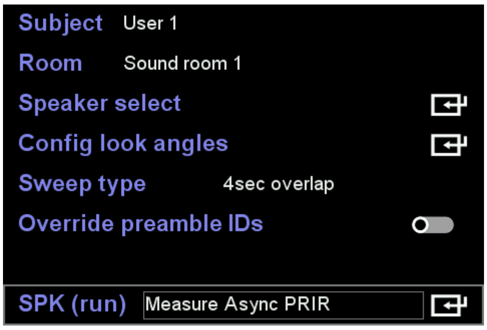
a) The subject name
b) The room description
c) The speaker setup
d) The speaker azimuth and elevation angles
e) The speaker path lengths

By default, the Async method will use the speaker IDs signalled in the preamble.



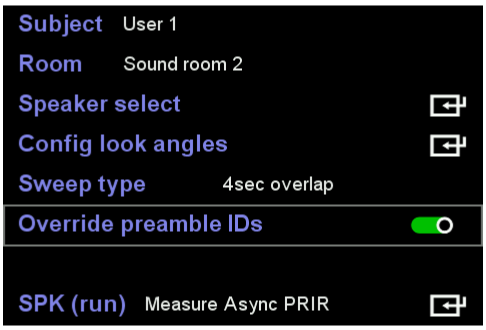


Next navigate to the **PRIR measurement (SPK)** and press enter. **Speaker select** and **Sweep type** menus are not used during Async PRIR measurements.

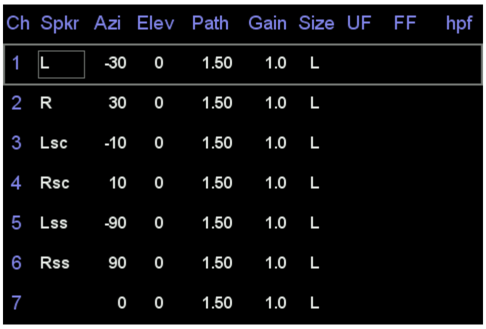


14.2.1 Override Preamble IDs

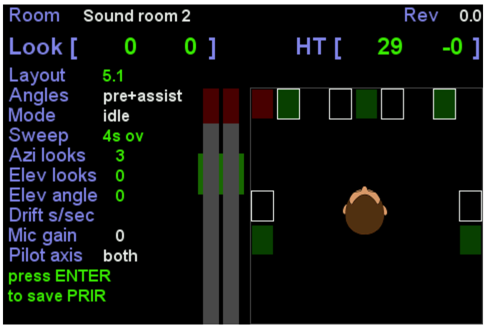
Normally asynchronous PRIR files inherit the virtual loudspeaker IDs (L, R, C etc) that are defined in the preamble track. However, there are times when it is desirable to insert into the PRIR file a different set of IDs, for example when the actual speaker layout does not conform to the preamble layout, or when one wishes to use generic multi-track Async sweeps to measure enhanced speaker positions. **Override preamble IDs** allows the user to override the loudspeaker IDs encoded in the preamble track.



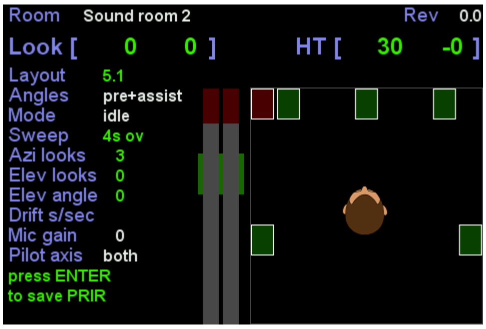
With **Override preamble IDs** enabled, loudspeaker IDs from the selected sound room are copied into the final Async PRIR. In the example below, Sound Room 1 describes a modified 5.1ch speaker layout where C and SW have been replaced by front screen speakers Lsc, Rsc and the surround speakers have been replaced by side surround speakers Lss and Rss.



Ch	Spkr	Azi	Elev	Path	Gain	Size	UF	FF	hpf
1	L	-30	0	1.50	1.0	L			
2	R	30	0	1.50	1.0	L			
3	Lsc	-10	0	1.50	1.0	L			
4	Rsc	10	0	1.50	1.0	L			
5	Lss	-90	0	1.50	1.0	L			
6	Rss	90	0	1.50	1.0	L			
7		0	0	1.50	1.0	L			



With **Override preamble IDs** disabled, loudspeaker IDs from the preamble track are used to populate the PRIR file. Below is a screen shot of a completed 5.1ch Async measurement made with the override disabled. The white boxes confirm the loudspeaker positions that will be transferred to the PRIR and correspond to the regular 5.1ch assignments L, R, C, SW, Ls and Rs.



By using the override feature, it is possible to measure any A16 speaker position by playing standard Async layouts such as 2ch, 5,1ch or 7.1ch from a BluRay player. For example, ch1 and ch2 outputs of an AV receiver/processor ordinarily drive the left front and right front speakers respectively. If these loudspeakers were temporarily moved to the Lsc and Rsc positions and the first two speaker IDs in the active sound room changed to Lsc and Rsc, then if the override is enabled whilst making a measurement using a regular 2ch (L, R) Async track, the PRIR will correctly assign the virtual speakers with IDs Lsc and Rsc. The same technique can be used to measure overhead or height speakers or any other speaker not defined in regular 2.0, 5.1, 7.1 layouts.

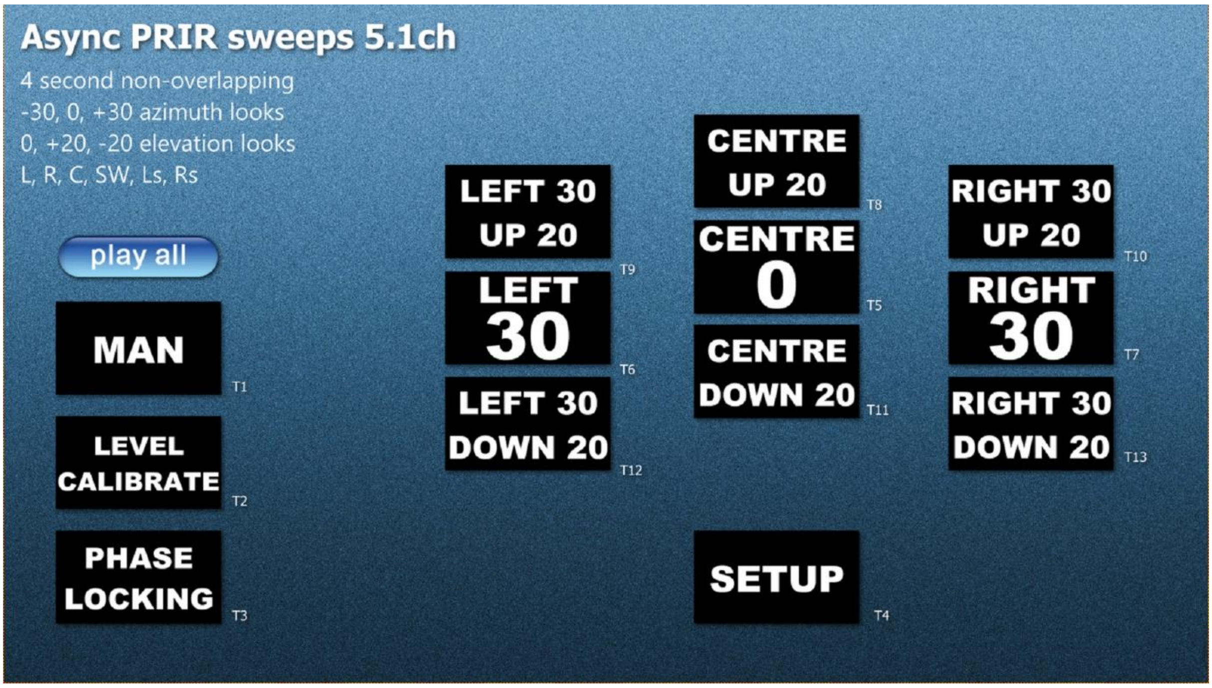
14.2.2 Config Look Angles

Navigate to **Config look angles** and press enter. **Look angles** is the only relevant configuration. When set to 'Fixed' the look angles inserted into the PRIR file are lifted from the Preamble Track.

When set to 'Free' the look angles inserted into the PRIR file are read from the User A head tracker at the beginning of each look track.

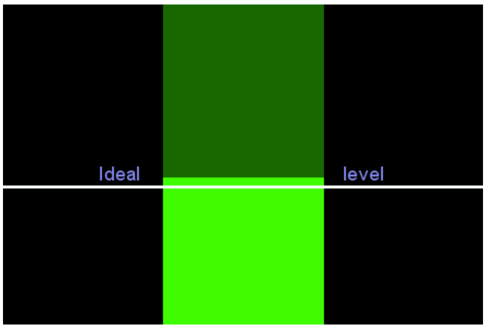
14.3 *Performing an ASYNC Measurement*

For this example Async Measurement, we will use the 5.1ch PCM AVCHD file shown below. This 5.1ch measurement suite contains all the tracks necessary to capture a 5.1ch speaker setup for three azimuth angles (centre, left 30 degrees and right 30 degrees) and two elevation angles (up 20 degrees and down 20 degrees)

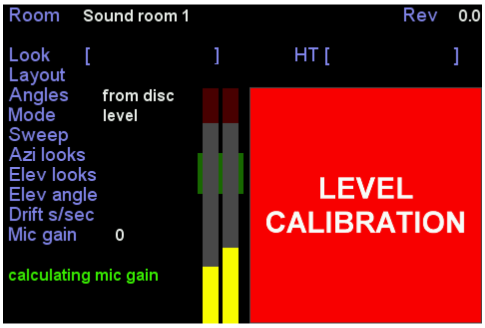


To start the Async PRIR application select **Measure Async PRIR** in **SPK (run)** and press enter.

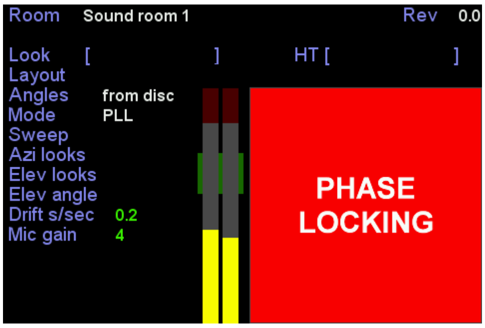
The first screen to appear displays a mono level bar that gauges the average loudness of the SVS microphones. Insert the in-ear microphones, sit in the sweet-spot, play the MAN noise track (T1) and adjust the volume until the levels average around the white line.



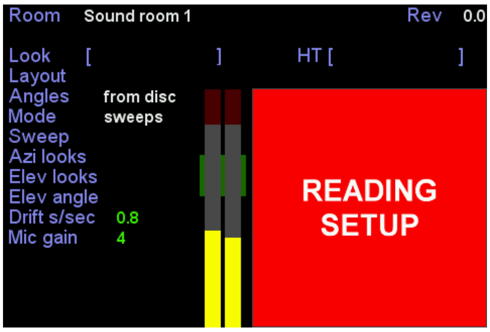
Press enter when you are happy with the level. Next play the Level Calibrate track (T2). The system will automatically detect when the track begins and the LEVEL CALIBRATION banner will appear on the A16 screen as shown below. This track consists of short sweeps played to each loudspeaker and the A16 finds the optimal microphone gain that maximizes the signal to noise ratio. This calibration should be undertaken in silence and with the subject looking towards the centre speaker.



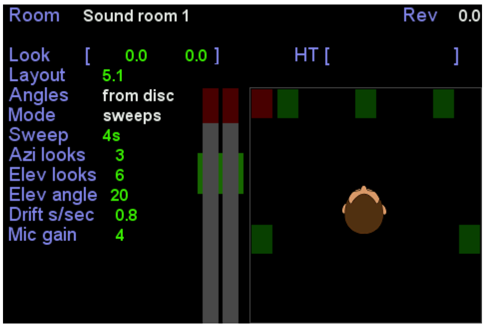
On completion of the level calibrate track, play the Phase Locking track (T3). Again, this is automatically detected by the A16. The purpose of this track is to allow the A16 to adjust its sample clock to match that of the players. This adjustment is required to maximize the quality of the resulting PRIR. In the example below the phase locking finishes with an average drift between the A16 and the video player of 0.2 samples per second, or 2 samples every 10 seconds. This calibration should be undertaken in silence and the subjects head should remain stationary throughout.



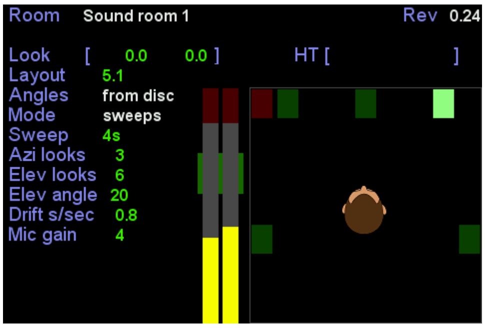
On completion of the phase locking, play the Setup (Preamble) track (T4). Again, this is automatically detected by the A16. The purpose of this track is to transfer the room layout, speaker IDs, look angles etc, embedded in the video track, to the A16.



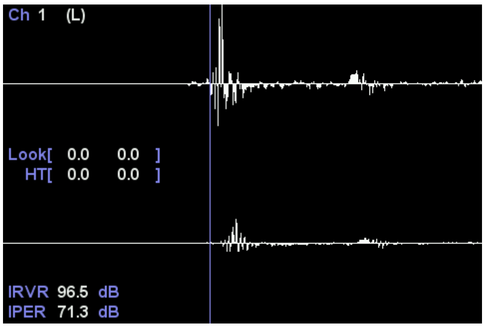
On completion of the setup track, the A16 display should now show the speaker setup that will be measured.



Once you are ready you can start the async PRIR measurement by playing the first look track Centre 0 (T5). The subject should position their head according to the voice instructions. For each look track a sweep is output to each loudspeaker in sequence.



As with the ALL method, it is possible to view the PRIR impulse response data at the end of each look track, or indeed on completion of the entire measurement, prior to saving back to the PRIR file. To activate the PRIR view window press enter. Use the up/down arrow keys to switch channels. Use the left/right arow keys to move between look angles. Press enter to exit.



In addition to the PRIR viewer, it is also possible to view a summary of the look angles that have been measured and those that remain to be measured. This ‘look grid’ window is activated by pressing the down arrow key while in the main Async PRIR window. On first running the Async Preamble audio track the look grid is initially populated with a red dot for each look track present in the Async sweep files. As each look is measured, the red dots change to green. By consulting the look grid page, it is easy to recall which looks have been measured and which have not. Note, all future Async measurement files/tracks generated by Smyth Research will be standardised on this look grid.

Proceed to play all the remaining look tracks (T6,7,…T13) positioning your head according to the voice prompt. It is not necessary to play the tracks in order, except that the final track (T13 in this example) which, on completion, is programmed to terminate the measurement. Furthermore, any of the tracks can be repeated at will if it is felt that a problem occurred during the sweeps, except again for the final track.

14.3.1 Improving Async look angles using the Head Tracker

Using the head tracker in conjunction with wearing the SVS microphones requires the use of the supplied head band. Attach the head tracker to the head band, fit the head band to the subject’s head, running both the HT cable and the SVS microphones back to the A16.

14.3.1.1 Using the Preamble Look angles in conjunction with HT assist.

With **Look angles** set to **Fixed** the look angles inserted into the PRIR file are simply a copy of those signalled on the preamble track. As a result, it is necessary for the subject to align their head with these look angles prior to each look measurement. Unless the look angles are very simple, having the subject move their head unaided can lead to significant degradation in the accuracy of the look angles and the SVS head tracking experience in general. In Async method a head tracking assist (HT assist) is always available and is output on User A headphone output. To use the assist, it is necessary to connect a small speaker to the headphone jack, or to an amplifier/loudspeaker. HT assist simply outputs a vertical and a horizontal pilot tone that guides the subject to orientate their head to achieve alignment with the look angle received from the preamble track. The operation of the HT assist is basically the same as that described in the ALL method except that it is for guidance only – it does not stop or start the look track playback. To give the subject sufficient time to align themselves, each look track must be paused following the double 2.2kHz burst (Pause) but before the DTMF deconvolution trigger (Start). Once head alignment has been achieved, the look track is set back in motion. In this way, the subject has the opportunity to reduce the look angle error before the deconvolution processing begins.

With **Look angles** set to **Fixed**, the look angles inserted into the PRIR file are those defined in the preamble track. Pilot tones are also output to HPA and use the same preamble look angles as the reference. To remind the user of this fact the main display flags **Angles pre + assist**.

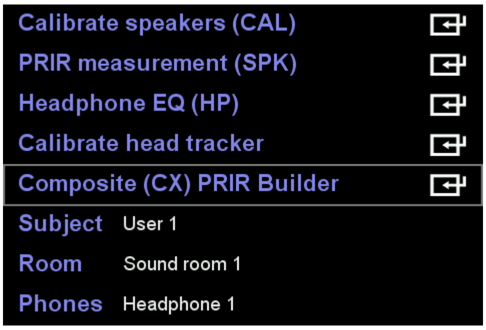
14.3.1.2 Replacing the Preamble Look angles with Head Tracker angles.

Switching **Look angles** to **Free** causes the look angles embedded in the preamble track to be replaced by the head tracker angle recorded at the start of each look track. Normally this will result in an accurate head tracking experience since there is no opportunity for the look angles to be in error. However, the downside is that the subject will not know with any certainty what the look angles actually are until the PRIR file has been generated. If this is an issue, one solution is to use HT assist in conjunction with the head tracker. Since HT assist pilot guides always reference the original preamble look angles, once alignment has been achieved the new look angles will be a very close copy. This serves two purposes. First it removes any tracking error and second it results in a PRIR that uses essentially the same look angles as the original preamble track. As described before, to give the subject sufficient time to align themselves, look tracks must be paused following the double 2.2kHz burst (Pause) but before the DFTM deconvolution trigger (Start). Once head alignment has been achieved, the look track is set back in motion. In this way, the subject has the opportunity to align their head with the preamble angle before deconvolution commences.

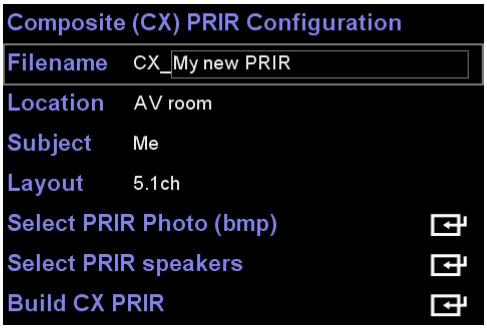
With **Look angles** set to **Free**, the look angles inserted into the PRIR file are those read from the head tracker at the start of each look track. Pilot tones are also output to HPA and use the original preamble look angles as the reference. To remind the user of this fact the main display flags **Angles HT + assist**.

15 Building a Composite PRIR file

The composite PRIR builder application is a tool to better manage PRIR files in general. The composite builder application can extract up to 35 virtual speakers from pre-existing PRIRs located in the A16's internal directory and create a new PRIR file comprising these speakers. PRIR speakers that have been created using different look-angle strategies can be loaded to the same composite PRIR without issue. The application also permits the original speaker labels, azimuth, and elevation angles to be altered, the virtual speaker impulse responses to be denoised and/or channel swapped, and for a photograph to be embedded into the PRIR file.



Proceed to the PRIR builder through the APPs menu page. On entering the Composite (CX) PRIR Configuration page, the PRIR filename, location, subject, and layout can be filled out using text entry. These text fields will be associated with the new PRIR. They are optional, but adding pertinent information make other functions, such as building sound rooms, more convenient as they remind the user as to the history of the speaker data. The new PRIR filename is always preceded with text 'CX_' to identify composite PRIRs.



15.1 PRIR Photo selection

A bit mapped (.BMP) photograph can be selected at this stage. The photo selector is programmed to look only in the 'realiser' directory of an external SD card. Furthermore, only 24-bit (colour) Bitmap pictures that are formatted 480 pixels (W) by 320 pixels (H) are accepted. Reformatting a photograph to Bitmap is widely supported in photo editing software. For example, the rudimentary Windows 'Paint' application can open any format (for example JPEG), resize the picture to 480x320 pixels and save back as 24-bit Bitmap. Similar applications are available on Mac PCs. Online JPEG to Bitmap convertors are also available on the internet.

Once you have installed an SD card with a 'realiser' directory containing 480x320 Bitmap photos, navigate to 'Select PRIR Photo' and press ENTER. The display will show the first photo found in the directory. If present, additional photos can be viewed by pressing either of the ADJ +/- keys. To select a photograph, press the enter key. To abort the photo viewing process, press the BACK key.

15.1.1 PRIR Speaker selection

To select which PRIR speakers you want to include in the new composite PRIR, navigate to 'Select PRIR speaker' and press ENTER. A blank speaker selection for the first seven speakers is initially displayed. Up to a total of 35 speakers can be included in a composite PRIR using additional pages accessed from below.

Each channel of the new PRIR is numbered on the left-side of the display. The virtual speaker ID (Vspkr), azimuth angle (Azi) and elevation angle (Elev) is displayed for each. 'Denoise' and 'Swap LR' operations can also be enabled or disabled for any channel.

Move the cursor to the desired channel and press ENTER. This will bring up the internal A16 PRIR directory. From the first line you can select the PRIR of interest - from the second line press ENTER to view the range of speakers available for this PRIR.

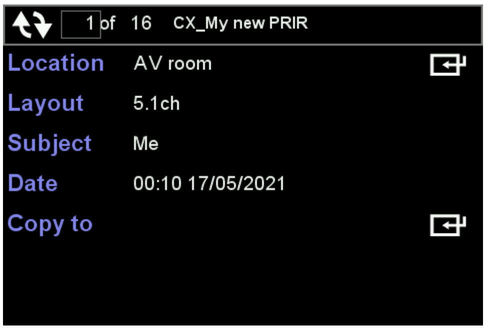
Navigate to the desired speaker and press ENTER. The menu will then jump back to the composite PRIR speaker list. Select the next PRIR channel and repeat the speaker selection. Continue until all the desired channels have been populated.

In the example above a new 5.1ch composite PRIR has been created from a collection of pre-existing PRIRs stored in the A16 internal PRIR directory. Note that any composite PRIR channel can hold any speaker. It is not necessary for the first channel to hold a 'L' speaker – any available speaker can occupy that channel. Indeed, composite PRIRs can even include speakers that share the same Vspkr ID. Neither is it necessary to populate channels contiguously – gaps in the populated channel list are permitted.

If no changes to the speaker ID, azimuth, or elevation angles are required then proceed to build the new PRIR by pressing BACK, navigating to 'Build CX PRIR' and pressing ENTER.

The routine will build the new PRIR channel by channel, add any selected Bitmap photograph and finally save the new composite PRIR to the A16 recycle PRIR directory.

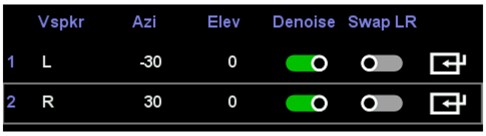
Navigate to ‘FILES’ in the home page and select ‘recycle PRIR files’ to view the new PRIR. If the composite PRIR has just been built, this will be the first file in the directory.



Composite PRIR files are the same as regular PRIR files in that they can be copied from the recycle directory to both the A16 internal PRIR directory and an external SD card realiser\PRIRs folder.

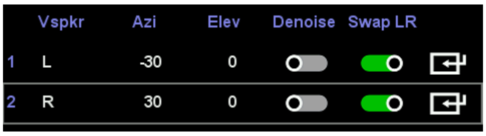
15.1.2 Denoising PRIR Speakers

Any PRIR speaker used to build a composite PRIR can first be denoised before being copied into the new file. A ‘Denoise’ enable/disable slider is provided for each channel allowing for selective denoising. As discussed previously, denoising is a process that reduces the reverberant effects of background noise present within the virtual speaker impulse response. Where a PRIR has been measured in studio conditions, denoising is rarely necessary.



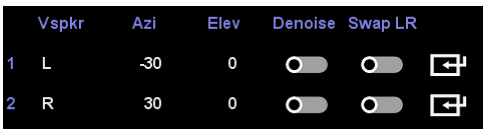
15.1.3 Left-Right Channel Swap

‘Swap LR’ slider controls are also available on a channel-by-channel basis. This functionality is only necessary where a PRIR has been measured with the microphones in the wrong ears, ie the left microphone in the right ear and visa-versa.



15.1.4 Altering Speaker IDs and/or Azimuth-Elevation Angles

As each composite channel is populated with speakers from pre-existing PRIRs, the speaker ID (Vspkr) and angular positions (Azi, Elev) are copied from those PRIRs. However, prior to initiating a build, the user is free to alter these values. Altering the Vspkr IDs has many uses. For example, where speakers were connected to the wrong amplifier channel during a PRIR measurement or the wrong A16 output was connected to the wrong loudspeaker, the PRIR Vspkr IDs needs changed to properly reflect the real position of the virtual speaker. This is particularly important when building listening rooms for Atmos/dtsX soundtracks since decoded audio is routed to virtual speakers according to their Vspkr ID. In this example a composite PRIR is loaded with PRIR speakers L and R. However, the Left and Right speakers are known to have been swapped during the PRIR measurement, meaning that the L vspkr is actually the right speaker and visa-versa.



This can be fixed by swapping the Vspkr IDs before building the new PRIR. Although not shown here, for the purposes of clarity the user may also wish to swap the azimuth angles to reflect the new positions.

	Vspkr	Azi	Elev	Denoise	Swap LR
1	R	-30	0		
2	L	30	0		

Another use is to give duplicated virtual speaker different IDs. In this example the same virtual speaker Ltf is loaded to two channels.

	Vspkr	Azi	Elev	Denoise	Swap LR
1	Ltf	-30	40		
2	Ltf	-30	40		

By altering the Vspkr ID of one of the channels to Lh, the subsequent composite PRIR is now compatible for both Atmos and dts-X listening rooms.

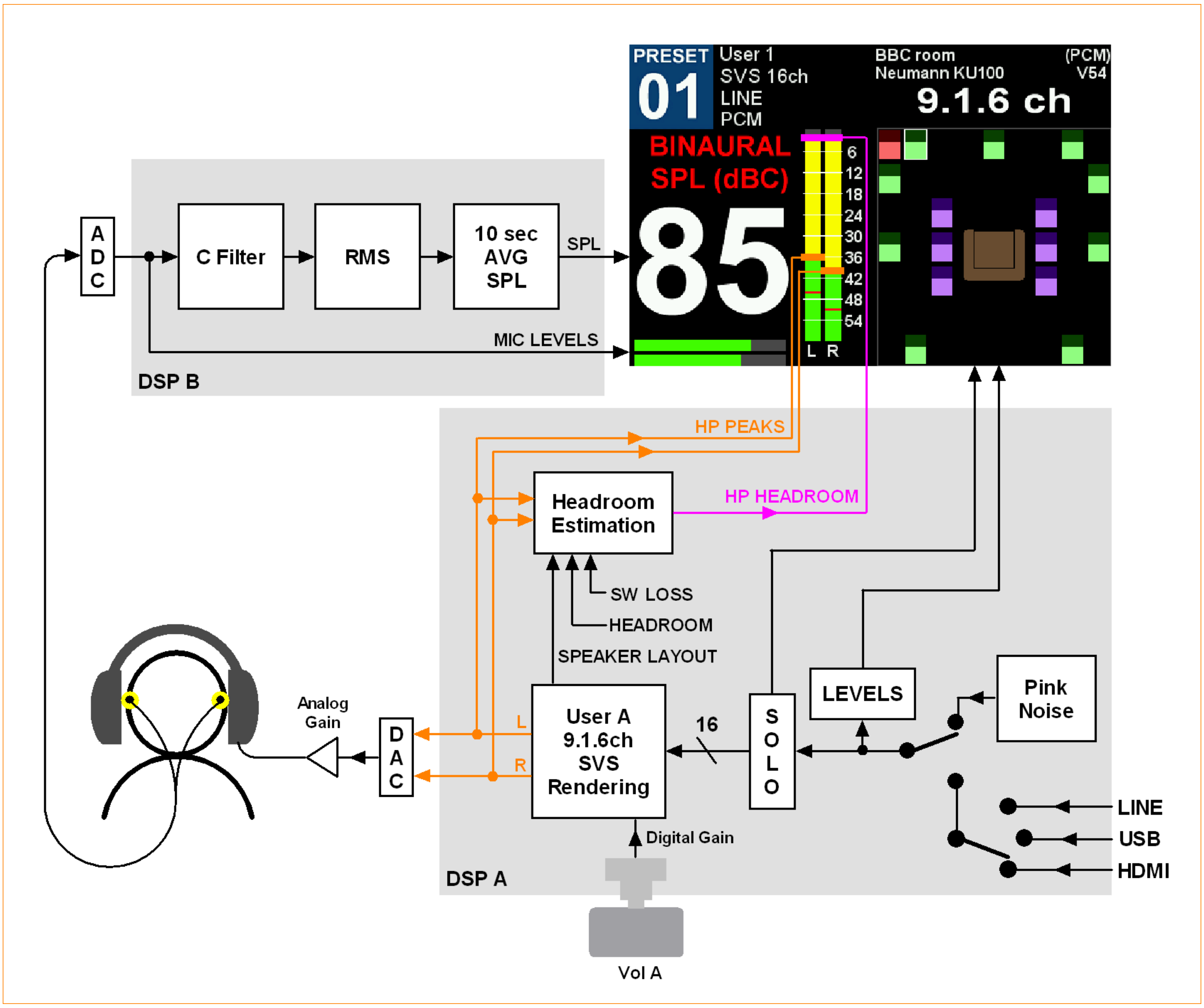
	Vspkr	Azi	Elev	Denoise	Swap LR
1	Lh	-30	40		
2	Ltf	-30	40		

15.1.5 Locked PRIRs

Where a composite PRIR includes speakers from a pre-existing PRIR that has been locked to the users A16, then the entire composite PRIR will be locked to their A16. To warn of this, PRIR speakers that are locked to the users A16 are flagged with a red 'L' against that channel. If the intention is to use the composite PRIR in a different A16, please ensure there are no locked PRIR speakers in the channel list prior to building the new PRIR.

	Vspkr		Azi	Elev	Denoise	Swap LR
1	L		-30	0		
2	R	L	30	0		
3	C		0	0		
4	Ls		-110	0		
5	Rs		110	0		
6			0	0		
7			0	0		

Movie soundtracks are commonly created in dubbing stages and mixing studios that are calibrated to a particular loudness or reference level. By playing back a movie soundtrack at the same reference level, the listener can replicate the sonic experience that would have been intended by the director during the production. Soundtracks that are destined for movie theatre playback are typically mixed at a reference level of 85dB SPL (measured at the sweet-spot while playing a -20dB pink noise signal through a single main speaker). Lower reference levels may be used for soundtracks targeted for home playback. For example, 79dB SPL and 76dB SPL reference levels are common for home movie and music playback. The A16 can calibrate the binaural sound pressure levels (SPL) as delivered by your headphones for the virtual listening room and to then set the listening levels referenced to this calibrated level. In this case the volume units are replaced by the reference level in dB SPL and the desired loudness experience is set by adjusting the playback reference level. For example, when you visit a cinema the playback reference level you will hear is typically between 79dB and 85dB SPL. By calibrating the A16 virtual listening room and headphone combination, this same playback level can be dialled in with ease allowing the listener to acoustically replicate what they experience at the movies with a high degree of confidence.

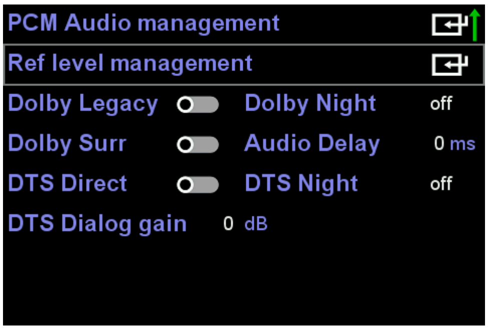


16.1 Theory of Operation

A reference level is the sound pressure level (SPL) of a full-band pink noise signal played out the left front or right front loudspeaker 20dB down from peak level, measured at the listening position. Typically, the SPL meter is set to filter the incoming microphone signal using a C filter and the calculated SPL value averaged using the slow setting. Hence to calibrate the playback volume to a reference level of 85dB, for example, one simply needs to play the pink noise signal from a DVD or Blu-ray player and adjust the speaker volume until the SPL meter reads 85dB SPL(C) at your listening position. From that point on, when listening to movie soundtracks, one just brings the volume back to the same position to listen at the 85dB reference level.

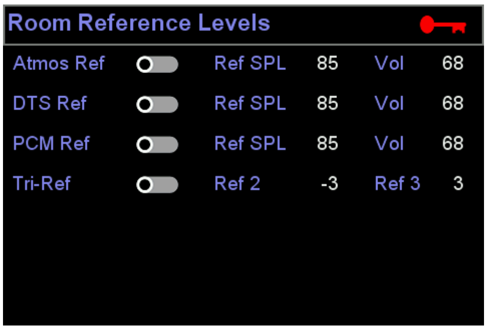
16.2 *Setting up the Reference Level using a simple comparison*

Setting the reference level in the A16 is essentially the same process. The only difference is that the listening room is a virtual experience in a headphone. Regular SPL meters cannot easily measure the volume levels in your ears. However, one way of overcoming this problem is to use a simple comparative procedure. In this case one sets up a real loudspeaker in a real room and sets its volume to the desired reference level using a pink noise test signal and the SPL meter positioned in the sweet spot. By sending the same pink noise signal into the A16, and selecting the left front loudspeaker in the headphone, a reference volume setting in the A16 is obtained when the pink noise intensity heard over the headphones is the same as what is heard directly from the speaker. Once you have this A16 volume value it can be entered into the Reference Level Management page of the Preset that holds the listening room used in the comparison.

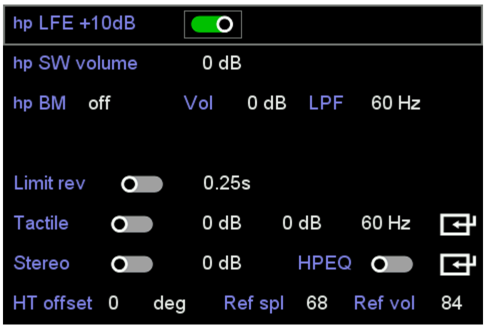


Reference Level management for Presets

For user A Presets the Ref Level Management entry point is found on the second page of the Preset. For user B Presets it is found in the main Preset page. On leaving the factory (or after a factory restore) the reference levels are set as shown below. The first three lines define the reference levels for the Atmos, DTS and PCM rooms, respectively. The fourth line defines the tri-ref function to be explained later.



Default Preset Reference Level setup

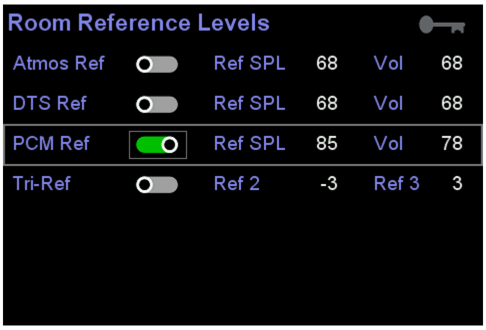


Reference Level data as they relate to the individual rooms

Each room reference level can be enabled or disabled. When enabled the A16 will use the Ref SPL volume value and the real time volume indicator will change from ‘V’ to ‘R’ and the volume screen will use REFERENCE as opposed to VOLUME as shown below. Note that these settings do not take effect until the Preset is reloaded.

Example Volume Screen when Ref SPL enabled

Assuming in the example of the sound room comparison, the reference level was measured at 85dB on the SPL meter and the volume of the left speaker of a virtual PCM room, loaded to the SVS renderer, matched the intensity of the real speaker when set to 78. Then the Ref SPL and Vol entries for the PCM Ref would be as follows.

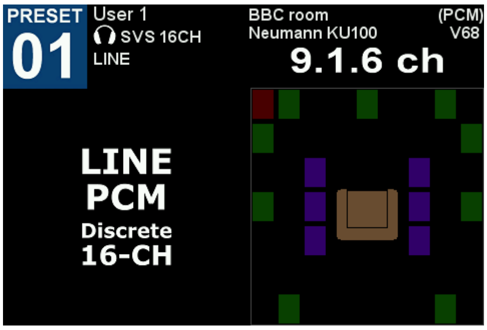


Using Reference Level SPL for PCM Room

By enabling the PCM Ref switch, the A16 will display the volume in reference SPL as opposed to volume units. However, the Preset must be reloaded for this to occur. Using a comparison method is easy to understand and relatively easy to undertake. However, unless the virtual PCM room is a measured copy (PRIR) of the real listening room you are using to make the comparison, the final reference level may only be accurate to within 2 to 3 dB since the spatial and tonal difference between them will make it difficult to achieve anything closer.

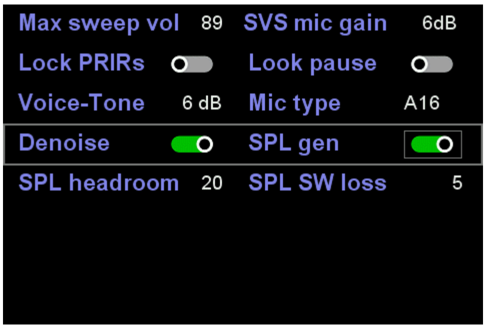
16.3 *Setting up the Reference Level using the Cal SPL Method*

Rather than subjectively comparing the loudness of a virtual speaker to a real speaker, the Cal SPL method instead makes use of the SVS microphones to measure the loudness level in the listeners ears while wearing headphones. Since all SVS microphones exhibit the same sensitivity to within +/-1dB, their voltage levels can be accurately mapped to actual SPL. A binaural SPL metering algorithm is run in real time within the A16 itself, with the microphones placed in the listeners ears. For Cal SPL, the user loads the virtual listening room to be calibrated (running on DSP A), feeds a pink noise signal into the virtual listening room, and then measures the pink noise level at the headphones by analysing the SVS microphone signals using the binaural SPL metering algorithm (running on DSP B). In the example below a 9.1.6ch PCM listening room will be subject to the reference level calibration procedure.



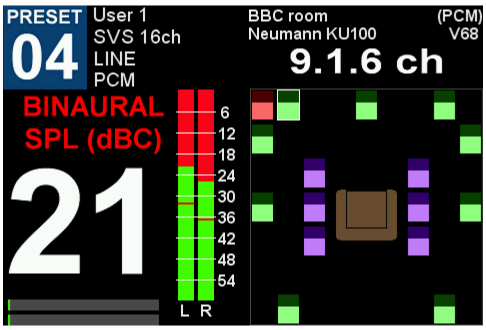
9.1.6ch PCM Listening Room

The user must then decide the pink noise source. Either an external signal is to be input via the LINE inputs (in this case) or the pink noise can be generated internally by enabling the pink noise SPL Gen in the MEASUREMENT settings page. In this example we will use the internal generator.



Enabling SPL Pink Noise

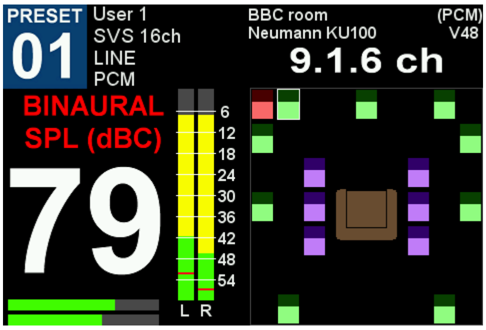
Finally, the user should press the PA key to display the speaker rendering page and then lower the A16 user A headphone volume to a safe level. Once the volume has been lowered the SPL calibration is started by pressing the CAL key.



Cal SPL Page

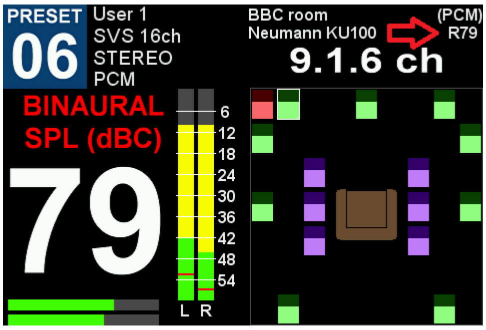
The SPL calibration routine always starts up with the front left virtual speaker in solo. Since in this example the pink noise is internally generated, the L-R headphone level meters immediately indicate the peak PCM signal level (green bars) heard through the headphones, and this level will rise and fall as the user A volume is adjusted in the normal way. Other speakers can be selected using the solo buttons on the remote control, but never more than one. Next the user inserts the SVS microphones into their ear canals (plugged into the A16 Mics jacks on the front panel) and then dons their headphones (plugged into user A HP jack), with the microphones still inserted. Since the headphone sensitivity is unknown it is recommended that the user A volume be first reduced by 30 units and then the user A switched gain (L-M-H) on the front panel set to H. By increasing the analogue gain in this way, the PCM headroom within the SVS rendering DSPs is maximised. If appropriate the HP balanced mode could also be deployed to further boost the analogue gain.

In the example below the user wishes to calibrate the reference level by finding the user A volume setting (using the remote control or using the volume knob) that brings the SVS microphone SPL levels to 79dB. In this example this is achieved with the volume set to 48 (V48 in the top right corner). The SPL measurement running in DSP B averages over a 10 second window so it is necessary to allow volume changes to take full effect before making further adjustments.



Reference level of 79dB SPL @V48

As with the comparative procedure described earlier, the SPL value of 79 and the volume value of 48 may be entered manually into the Ref Level Management entry page of the Preset used to load the listening room under test, or indeed any Preset that uses the same room. Note that the Preset must be reloaded for these to take effect. Alternatively, the values can be entered automatically by depressing the user B volume knob. In this case the PCM Ref switch is also enabled causing the A16 volume to switch to reference dB SPL. Whilst the PCM Ref switch remains active, each time the A16 loads that room from that Preset, the A16 will switch to reference volume units. Disabling the switch will cause the system to revert to regular volume units.

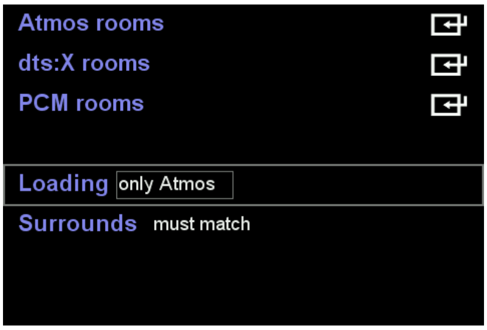


Depressing Vol B knob switches to Ref SPL

To exit the CAL SPL routine, press the CAL key again and the A16 will exit and then reload both user A and user B Presets. Note that Cal SPL only operates on listening rooms loaded by user A Preset. To calibrate a listening room intended for use on the B side, first create a user A Preset that uses the same listening room. Then run Cal SPL on that room. The room will now have updated reference level data. Reload the user B preset for the new data to take effect.

16.4 *Calibrating an Atmos or dts:X Room*

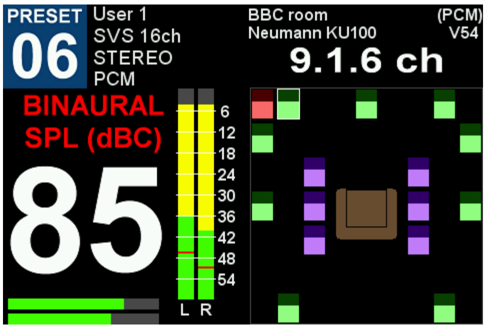
Since the SPL calibration routine operates on the listening room currently loaded in the user A Preset, then to calibrate an Atmos/dts:X listening room, one must ensure it is loaded prior to activating Cal SPL. The easiest way to force an Atmos/dts:X room load before activating Cal SPL is to disable the automatic Loading by selecting either 'only Atmos' or 'only dts:X'. On selecting either, the user A Preset will automatically load the selected room type.



Forced Listening Room load

16.5 *Estimating Reference Level Headroom*

For any reference level it is critical to know whether the combined signal level of the virtual sound stage at that volume level will have sufficient digital headroom as the input signals approach peak. To help with this calculation a headroom estimation algorithm is used to display yellow headroom bars that ride atop the green headphone peak level bars.

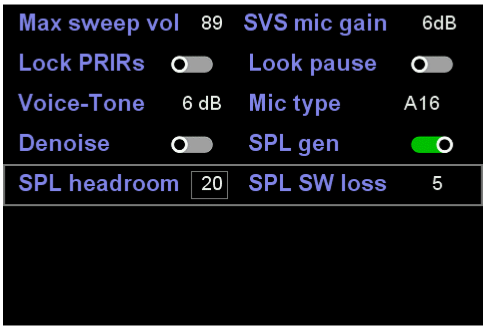


Headroom estimation at 85dB SPL

In the example above the Cal SPL routine is presently estimating the headphone headroom necessary to cope with peak incoherent signals on all inputs of a 9.1.6ch room set to a reference listening level of 85dB SPL. The peak headphone levels are shown in green. The L-ch is peaking around -36dBFS while the R-ch is peaking around -40dBFS. The yellow bars indicate the increase in level that would occur if all 16 channels were fed independent peak pink noise with all speakers set to 0dB gain, except the SW which is set at +10dB.

The estimation calculation uses the number of speakers, the SPL Headroom, and the SPL SW Loss. Both SPL Headroom and SPL SW Loss can be adjusted from the Measurement Setting page as shown below. SPL Headroom is simply the maximum peak input signal above reference level. Typically, this is +20dB. SPL SW Loss is the drop in measured SPL when switching from the main speakers to a subwoofer using a C filter (without the +10dB boost). Typically, a 80Hz subwoofer exhibits a drop of 5 to 6dB. In Cal SPL, a SW reference level of 85-5+10=90dB SPL is used in the estimator. Note that the headroom estimator does not consider Bass Management or Direct Bass processing but in most circumstances neither of these are likely to exceed a direct LFE-SW signal level.

In this example the routine estimates that the peak level could increase by a further 32dB giving at maximum binaural SPL of 117dB and leaving a margin of 4dB before the onset of clipping. In other words, we expect that typical dynamic excursion away from reference level, as might occur in action movies, should not clip the headphone PCM signal. When the estimation routine determines that insufficient digital headroom is available to guarantee clip free playback, the yellow bars are replaced by red bars to act as a warning. In such cases it would be necessary either to use more sensitive headphones or increase the analogue gain driving the headphones (L-M-H gain switch or external HP amplifier) or use a room with fewer speakers or reduce the reference listening level. Another option would be to accept the possibility of clipping and to consider enabling the automatic clip attenuation. In this way the A16 could take action to reduce the listening levels if clipping does eventually occur.



Digital Headroom parameters

Note also that the headroom calculation applies not only to the A16s internal headphone DACs but also to both SPDIF HP outputs and the USB stereo line out (in the case of user A). Note also that you must recalibrate the reference level if either the headphone type or the gain position on the A16 front panel slider switch are different to those used during the original calibration.

16.6 *Reference Levels and External Headphone Amplifiers*

For the reference level and headroom estimation to make sense, all volume adjustments must use the A16 volume controls (volume knob or remote control), both for the measurement itself and all subsequent use. An external headphone amplifier is simply external gain, and its volume setting forms part of the calibrated chain and therefore must remain fixed. Ideally the volume of the external amplifier would be left at maximum. This ensures the amplifier itself does not run out of steam due to lack of gain, but this may be excessive especially for low impedance headphones or those with a high sensitivity. It all depends on the parameters of the amp-headphone combination and would require some experimentation to find the best setting.

16.7 *Steps to calibrate a User A Listening Room reference level*

1) Set the Loading option in the Listening Room menu that selects the desired room type (Atmos/dts:X/PCM)

2) Plug in headphones and SVS microphones
3) Create a Preset for user A that assigns the listening room to be calibrated to the desired room type
4) In that Preset assign a HPEQ file for the headphone you intend to use
5) Ensure AV mode is disabled
6) Load the user A Preset just created
7) Ensure TEST mode is not active
8) Ensure the A16 is running in 16ch SVS mode
9) Navigate to SETTINGS->SYSTEM->MEASUREMENT SETTINGS and enable SPL Gen
10) In the same page set SPL headroom to 20 and SPL SW loss to 5
11) Return to the home page
12) Press PA key to display the speaker render page for user A
13) Press the CAL key and the SPL calibration screen will load after approximately 5 seconds
14) Initially reduce the headphone volume to a safe level
15) Insert SVS microphones and don headphones without disturbing the mics
16) Adjust the user A volume slowly until the desired binaural SPL is reached
17) Wait at least 10 seconds to let the meter settle between each adjustment
18) Observe the headphone headroom for any final SPL setting
19) If happy depress user B volume knob to log the reference SPL and volume numbers.
20) Press CAL key to exit

16.8 *Headphone SPL Headroom*

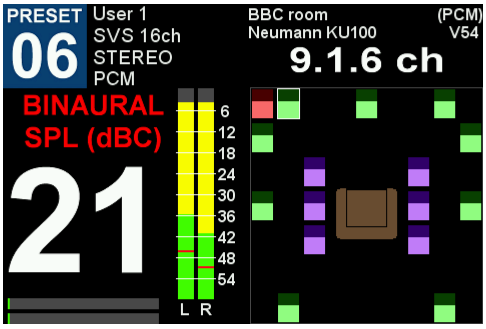
Although we now have some idea of the likely headphone voltage excursions the remaining issue is whether the headphone (or headphone plus external amplifier) in use has sufficient headroom to cope with such excursions and what levels of distortion are likely to accompany them if they do. This is subject to further study.

16.9 *Using only the Reference Level SPL measurement*

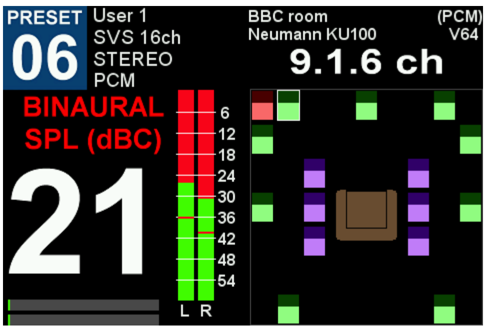
The binaural SPL measurement can, on its own, be useful where there is a need to estimate the SPL of a headphone not being driven by the A16 but from some other audio chain. For example, by playing -20dB pink noise through a headphone connected to a PC or a phone, the SPL levels are easily determined by activating the Cal SPL routine in conjunction with the SVS microphones. In this case the digital headphone headroom estimator should be ignored.

16.10 *Using only the Headroom Estimator*

Conversely, the headphone PCM signal headroom estimator can, on its own, be useful just to predict the available headroom for any A16 volume level. In this case neither SVS microphones nor the headphones are required, and the binaural SPL number can be ignored. The user simply loads the listening room using a user A Preset, enables the SPL pink noise function and activates Cal SPL. For any desired volume, if yellow bars are displayed above the green headphone peak level meters, then sufficient headroom exists. If red bars are present, then insufficient headroom exists and clipping is possible as the input levels approach peak amplitude. If the headphone clip attenuation is enabled, then the volume will automatically be lowered if clipping does in fact occur. Headroom estimation is also possible using the metering function described elsewhere in this document.



Vol=54, headroom sufficient



Vol=64, headroom insufficient

16.11 *Measuring Binaural SPLs for subwoofers*

While running the Cal SPL routine the binaural SPL for the subwoofer can be measured by pressing the remote LFE key. In theory, for a subwoofer (80Hz low pass) set to the same gain as the main speakers, the binaural SPL would be approximately 5dB below that for the main speakers. For example, if the desired SPL was 85dB SPL then the SW should measure around 80dB SPL (for a subwoofer with a 10dB gain, then the SPL would measure at 90dB). However, measuring SW SPL is somewhat inaccurate in practice, particularly using the apparatus the Cal SPL routine must use. Pink noise is louder in the low frequencies than in the high frequencies. This means that the SPL calculation is influenced more by the lows than the highs. As a result, any low frequency roll-off in the measurement chain will progressively degrade the SPL measurement accuracy as the bandwidth of the pink noise decreases. For example, a 20Hz roll-off will reduce the accuracy of a full-band pink noise (20Hz-20kHz) measurement by less than 1dB whereas the same roll-off when measuring a Subwoofer (20Hz-80Hz) will introduce an underestimation of almost 3dB. In our case the SVS microphones and the headphones are the source of the roll-off and hence there is a risk that the SW gain will be set too high because of this inherent underestimation. Our recommendation would be to calibrate the subwoofer using an SPL meter before capturing a PRIR. In that way the SW levels will already be at the correct level relative to the main speakers. If you must use the Cal SPL routine to set the SW speaker gain, we suggest the following procedure.

1) Select the left front speaker and adjust the A16 volume to attain the desired reference level.
2) Select the SW speaker and take note of the binaural SPL.
3) Exit Cal SPL and navigate to Listening Rooms and locate the SW for the room you are calibrating. Adjust the SW speaker gain (see below) such that the combined gain + binaural SPL will be approximately 8dB below reference level, for a 0dB SW chain, or +2dB above the reference level, for a +10dB SW chain. Let's assume the +10dB SW boost will be applied to the SW signal. Hence the SW speaker gain should be adjusted to produce a binaural SPL 8dB below reference. For example, if the reference level is 85dB and the measured SW SPL was 78dB then the SW gain should be set to -1dB so that the final SW SPL will be 77dB.
4) Exit the Listening Room menu (the room you were editing should be reloaded automatically)
5) Re-enter the Cal SPL routine. Confirm the SW SPL is correct relative to a main speaker reference SPL.

	Ref	Vspkr	Gain	Location	Subject	Azi	Elev
4	SW	SW	-1	BBC room	Neumann	-45	0
5	Lss	Lss	0	BBC room	Neumann	-90	0
6	Rss	Rss	0	BBC room	Neumann	90	0
7	Lb	Lb	0	BBC room	Neumann	-135	0
8	Rb	Rb	0	BBC room	Neumann	135	0
9	Lw	Lw	0	BBC room	Neumann	-45	0
10	Rw	Rw	0	BBC room	Neumann	45	0

Adjusting the SW speaker gain

Please note that the SW speakers supplied in the factory PRIR files are already level matched to the main factory speakers. If the LFE signal is to be boosted by +10dB then no gain changes are required. Apply a 10dB gain to the SW speaker if the LFE will be fed directly to the SW without the boost.

16.12 *Reference Levels for Sound Rooms with more than 16 speakers*

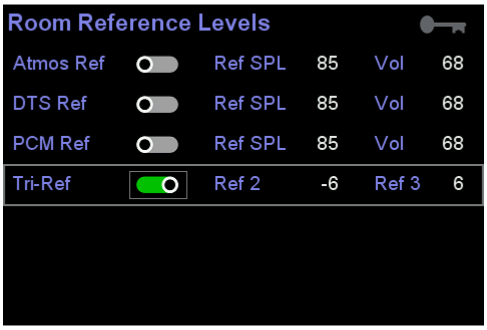
Because the SPL metering algorithm runs on the B-side DSP it is presently only possible to conduct 16-ch reference level measurements. However, if the speakers that occupy channels 17 through to 24 are from the same listening room as those in channels 1-16, it is possible to simply calibrate the first 16 channels and copy the calibration data to the 24ch room in the Listening Room Configuration menu. The only error that arises is that the headroom will be underestimated by approximately 1.5dB for a 24ch room compared to the 16ch room.

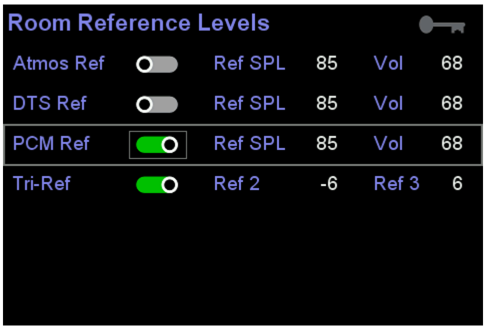
16.13 *Tri-Ref Volume*

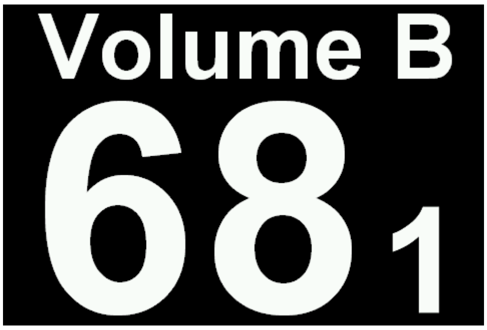
Tri-Ref volume is used in conjunction with the reference level data of the currently loaded room and allows the user to switch between the reference volume and two offset volumes (Ref 2 and Ref 3). When Tri-Ref is enabled, adjusting the volume for that user causes the A16 to cycle between the reference volume followed by Ref 2, followed by Ref 3, and then back to reference volume, and so on. The offset volumes can be adjusted up to +/-20dB from the reference.

In the example below, user B has enabled the Tri-Ref feature and the PCM room specified in user B preset is currently loaded. The Ref 2 and Ref 3 offsets are -6 and +6dB, respectively. When PCM Ref is disabled, Tri-Ref uses the Vol value as the baseline volume and Ref 2 and Ref 3 are offset from this value. As shown, on adjusting the user B volume, the system cycles between 68, 62 and 74. When PCM Ref is enabled, Tri-Ref uses the Ref SPL as the baseline volume and Ref 2 and Ref 3 are now offset from this value. As shown, the system now cycles between 85, 79 and 91.

Note: If reference level operation is not required the **Tri-Ref** function can be used as a simple 3-level volume control by disabling **Atmos/DTS/PCM Ref** and then adjusting the **Vol** value in the Listening Room Configuration menu to the desired baseline volume.


Room Reference Levels
Atmos Ref Ref SPL 85 Vol 68
DTS Ref Ref SPL 85 Vol 68
PCM Ref Ref SPL 85 Vol 68
Tri-Ref Ref 2 -6 Ref 3 6


Room Reference Levels
Atmos Ref Ref SPL 85 Vol 68
DTS Ref Ref SPL 85 Vol 68
PCM Ref Ref SPL 85 Vol 68
Tri-Ref Ref 2 -6 Ref 3 6


Volume B
68 1


Reference B
85 1


Volume B
62 2


Reference B
79 2


Volume B
74 3

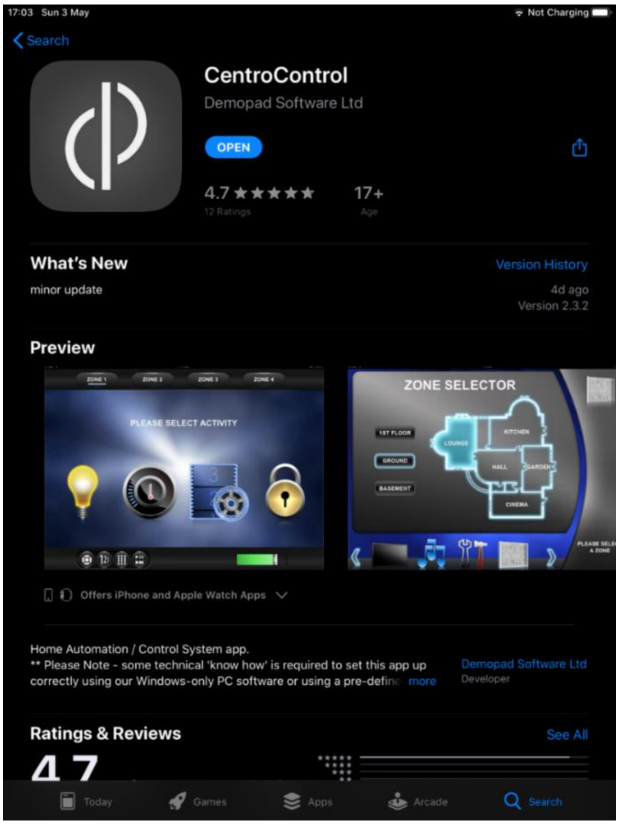

Reference B
91 3

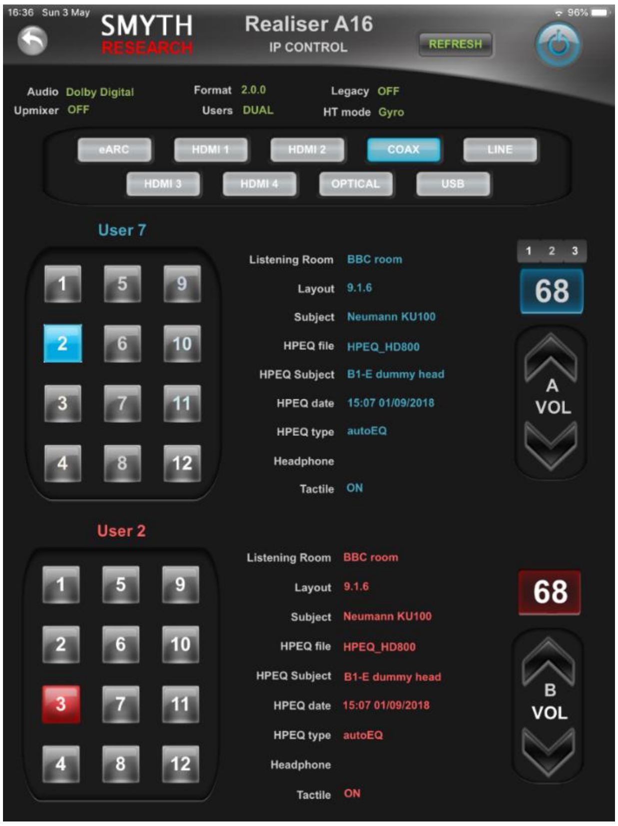
17 TCP Command Server

The A16 implements a simple TCP based command and response protocol that allows a remote device (client) to control and/or monitor certain operational aspects of the A16 in real time over a home network. Document 'Realiser A16 IP (TCP) Command Summary, 5 May 2020, S. Smyth' discusses our protocol and provides all the details necessary for anyone wishing to write their own TCP client application.

17.1 *DemoPad A16 Control App for IOS*

A free iOS A16 control app developed using third party app platform DemoPad, can be downloaded into your own copy of iOS Centro Control app (Apple Store) using the QR code included in our summary document. The DemoPad application was undertaken purely to test our server implementation and does not support all the functionality in the A16. Please refer to document 'Realiser A16 IP (TCP) Command Summary, 5 May 2020, S. Smyth' if you would like to try out our example application.





17.2 *Developing your own TCP client application*

Please refer to 'Realiser A16 IP (TCP) Command Summary, 5 May 2020, S. Smyth' for more information. All the current IP command codes are listed in this document. We suggest that anyone interested in developing their own IP command client application, test the commands that interest them using a TCP application such as Packet Sender. Packet Sender not only tests/analyses the packet payloads but also serves to illustrate the timing of the commands and responses.

17.3 *Developing your own DemoPad A16 Control App for IOS*

Anyone interested in developing their own Centro control app should proceed to DemoPad website (www.demopad.com) and download the DemoPad Designer tool (note this only runs on Windows). Anyone who wishes to modify our example app should email James (james@smyth-research.com) and we will send you a ZIP file of our project source files. Please note that use of our example Centro Control app and/or our project source files are entirely at your own risk.

18 Appendix A: Listening rooms loudspeaker configurations

18.1 Dolby Atmos Listening Rooms loudspeaker configurations (16ch)

		Line Output Channel Number															
#	**Mode**	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**	**14**	**15**	**16**
1	2.0	L	R														
2	2.1	L	R		SW												
3	2.2	L	R		SW					SW2							
4	2.0.2m	L	R											Ltm	Rtm		
5	2.1.2m	L	R		SW									Ltm	Rtm		
6	2.2.2m	L	R		SW					SW2							
7	3.0	L	R	C													
8	3.1	L	R	C	SW												
9	3.2	L	R	C	SW					SW2							
10	3.0.2m	L	R	C										Ltm	Rtm		
11	3.1.2m	L	R	C	SW									Ltm	Rtm		
12	3.2.2m	L	R	C	SW					SW2				Ltm	Rtm		
13	4.0	L	R			Ls	Rs										
14	4.1	L	R		SW	Ls	Rs										
15	4.2	L	R		SW	Ls	Rs			SW2							
16	4.0.2m	L	R			Ls	Rs							Ltm	Rtm		
17	4.1.2m	L	R		SW	Ls	Rs							Ltm	Rtm		
18	4.2.2m	L	R		SW	Ls	Rs			SW2				Ltm	Rtm		
19	5.0	L	R	C		Ls	Rs										
20	5.1	L	R	C	SW	Ls	Rs										
21	5.2	L	R	C	SW	Ls	Rs			SW2							
22	5.0.2m	L	R	C		Ls	Rs							Ltm	Rtm		
23	5.1.2m	L	R	C	SW	Ls	Rs							Ltm	Rtm		
24	5.2.2m	L	R	C	SW	Ls	Rs			SW2				Ltm	Rtm		
25	5.0.4	L	R	C		Ls	Rs					Ltf	Rtf			Ltr	Rtr
26	5.1.4	L	R	C	SW	Ls	Rs					Ltf	Rtf			Ltr	Rtr
27	5.2.4	L	R	C	SW	Ls	Rs			SW2		Ltf	Rtf			Ltr	Rtr
28	5.0.6	L	R	C		Ls	Rs					Ltf	Rtf	Ltm	Rtm	Ltr	Rtr
29	5.1.6	L	R	C	SW	Ls	Rs					Ltf	Rtf	Ltm	Rtm	Ltr	Rtr
30	5.2.6	L	R	C	SW	Ls	Rs			SW2		Ltf	Rtf	Ltm	Rtm	Ltr	Rtr
31	6.0	L	R			Lss	Rss	Lb	Rb								
32	6.1	L	R		SW	Lss	Rss	Lb	Rb								
33	6.2	L	R		SW	Lss	Rss	Lb	Rb	SW2							
34	6.0.2m	L	R			Lss	Rss	Lb	Rb					Ltm	Rtm		
35	6.1.2m	L	R		SW	Lss	Rss	Lb	Rb					Ltm	Rtm		
36	6.2.2m	L	R		SW	Lss	Rss	Lb	Rb	SW2				Ltm	Rtm		
37	7.0	L	R	C		Lss	Rss	Lb	Rb								
38	7.1	L	R	C	SW	Lss	Rss	Lb	Rb								
39	7.2	L	R	C	SW	Lss	Rss	Lb	Rb	SW2							
40	7.0.2m	L	R	C		Lss	Rss	Lb	Rb					Ltm	Rtm		
41	7.1.2m	L	R	C	SW	Lss	Rss	Lb	Rb					Ltm	Rtm		
42	7.2.4h	L	R	C	SW	Lss	Rss	Lb	Rb	SW2		Ltf	Rtf			Ltr	Rtr
43	7.2.2m	L	R	C	SW	Lss	Rss	Lb	Rb	SW2				Ltm	Rtm		

44	*7.0.4*	*L*	*R*	*C*		*Lss*	*Rss*	*Lb*	*Rb*			*Ltf*	*Rtf*			*Ltr*	*Rtr*
45	*7.1.4*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*	*Lb*	*Rb*			*Ltf*	*Rtf*				
46	*7.2.4*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*	*Lb*	*Rb*	*SW2*		*Ltf*	*Rtf*			*Ltr*	*Rtr*
47	*7.0.6*	*L*	*R*	*C*		*Lss*	*Rss*	*Lb*	*Rb*			*Ltf*	*Rtf*	*Ltm*	*Rtm*	*Ltr*	*Rtr*
48	*7.1.6*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*	*Lb*	*Rb*			*Ltf*	*Rtf*	*Ltm*	*Rtm*	*Ltr*	*Rtr*
49	*7.2.6*	*L*	*R*	*C*		*Lss*	*Rss*	*Lb*	*Rb*	*SW2*		*Ltf*	*Rtf*	*Ltm*	*Rtm*	*Ltr*	*Rtr*
50	*8.0*	*L*	*R*			*Lss*	*Rss*	*Lb*	*Rb*	*Lw*	*Rw*						
51	*9.0.2m*	*L*	*R*	*C*		*Lss*	*Rss*	*Lb*	*Rb*	*Lw*	*Rw*			*Ltm*	*Rtm*		
52	*9.1.2m*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*	*Lb*	*Rb*	*Lw*	*Rw*			*Ltm*	*Rtm*		
53	*9.2.2m*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*	*Lb*	*Rb*	*Lw*	*Rw*			*Ltm*	*Rtm*	*SW2*	
54	*9.0.4*	*L*	*R*	*C*		*Lss*	*Rss*	*Lb*	*Rb*	*Lw*	*Rw*	*Ltf*	*Rtf*			*Ltr*	*Rtr*
55	*9.1.4*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*	*Lb*	*Rb*	*Lw*	*Rw*	*Ltf*	*Rtf*			*Ltr*	*Rtr*
56	*9.2.4*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*	*Lb*	*Rb*	*Lw*	*Rw*	*Ltf*	*Rtf*	*SW2*		*Ltr*	*Rtr*
57	*9.0.6*	*L*	*R*	*C*		*Lss*	*Rss*	*Lb*	*Rb*	*Lw*	*Rw*	*Ltf*	*Rtf*	*Ltm*	*Rtm*	*Ltr*	*Rtr*
58	*9.1.6*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*	*Lb*	*Rb*	*Lw*	*Rw*	*Ltf*	*Rtf*	*Ltm*	*Rtm*	*Ltf*	*Rtr*

Table 1 Supported loudspeaker configurations for Dolby Atmos listening rooms in dual-user (16ch) mode.

18.2 *Dolby Atmos Listening Rooms loudspeaker configurations (24ch)*

#	Mode	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24
1	5.0.4h	L	R	C		Ls	Rs					Lh	Rh			Lhr	Rhr								
2	5.1.4h	L	R	C	SW	Ls	Rs					Lh	Rh			Lhr	Rhr								
3	7.0.4h	L	R	C		Lss	Rss	Lb	Rb			Lh	Rh			Lhr	Rhr								
4	7.1.4h	L	R	C	SW	Lss	Rss	Lb	Rb			Lh	Rh			Lhr	Rhr								
5	7.0.6h	L	R	C		Lss	Rss	Lb	Rb			Lh	Rh	Ltm	Rtm	Lhr	Rhr								
6	7.1.6h	L	R	C	SW	Lss	Rss	Lb	Rb			Lh	Rh	Ltm	Rtm	Lhr	Rhr								
7	8.0.4	L	R	C	Cr	Lss	Rss	Lb	Rb			Ltf	Rtf			Ltr	Rtr								
8	8.0.4h	L	R	C	Cr	Lss	Rss	Lb	Rb			Lh	Rh			Lhr	Rhr								
9	8.1.4	L	R	C	SW	Lss	Rss	Lb	Rb		Cr	Ltf	Rtf			Ltr	Rtr								
10	8.1.4h	L	R	C	SW	Lss	Rss	Lb	Rb		Cr	Lh	Rh			Lhr	Rhr								
11	8.0.6	L	R	C	Cr	Lss	Rss	Lb	Rb			Ltf	Rtf	Ltm	Rtm	Ltr	Rtr								
12	8.0.6h	L	R	C	Cr	Lss	Rss	Lb	Rb			Lh	Rh	Ltm	Rtm	Lhr	Rhr								
13	8.1.6	L	R	C	SW	Lss	Rss	Lb	Rb		Cr	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr								
14	8.1.6h	L	R	C	SW	Lss	Rss	Lb	Rb		Cr	Lh	Rh	Ltm	Rtm	Lhr	Rhr								
15	9.0.4h	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh			Lhr	Rhr								
16	9.1.4h	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh			Lhr	Rhr								
17	9.0.6h	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltm	Rtm	Lhr	Rhr								
18	9.1.6h	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltm	Rtm	Lhr	Rhr								
19	9.0.8	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltr	Rtr	Lhr	Rhr						
20	9.1.8	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltr	Rtr	Lhr	Rhr						
21	10.0.4	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Lth	Rth			Ltr	Rtr								
22	10.0.4h	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh			Lhr	Rhr								
23	10.1.4	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lth	Rth	Cr		Ltr	Rtr								
24	10.1.4h	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Cr		Lhr	Rhr								
25	10.0.6	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr								
26	10.0.6h	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltm	Rtm	Lhr	Rhr								
27	10.1.6	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Cr							
28	10.1.6h	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltm	Rtm	Lhr	Rhr	Cr							
29	10.0.8	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltr	Rtr	Lhr	Rhr						
30	10.1.8	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltr	Rtr	Lhr	Rhr	Cr					
31	11.0.4	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Lrs1	Rrs1	Ltr	Rtr								
32	11.0.4h	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Lrs1	Rrs1	Lhr	Rhr								
33	11.1.4	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Lrs1	Rrs1	Ltr	Rtr								
34	11.1.4h	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Lrs1	Rrs1	Lhr	Rhr								
35	11.0.6	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Lrs1	Rrs1						
36	11.0.6h	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltm	Rtm	Lhr	Rhr	Lrs1	Rrs1						
37	11.1.6	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Lrs1	Rrs1						
38	11.1.6h	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltm	Rtm	Lhr	Rhr	Lrs1	Rrs1						
39	11.0.8	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltr	Rtr	Lhr	Rhr	Lrs1	Rrs1				
40	11.1.8	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltr	Rtr	Lhr	Rhr	Lrs1	Rrs1				
41	11.0.10	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Lhr	Rhr	Lrs1	Rrs1		
42	11.1.10	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Lhr	Rhr	Lrs1	Rrs1		
43	12.0.4	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Lrs1	Rrs1	Ltr	Rtr								
44	12.0.4h	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Lrs1	Rrs1	Lhr	Rhr								
45	12.1.4	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Lrs1	Rrs1	Ltr	Rtr	Cr							
46	12.1.4h	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Lrs1	Rrs1	Lhr	Rhr	Cr							
47	12.0.6	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Lrs1	Rrs1						

48	12.0.6h	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltm	Rtm	Lhr	Rhr	Lrs1	Rrs1						
49	12.1.6	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Lrs1	Rrs1	Cr					
50	12.1.6h	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltm	Rtm	Lhr	Rhr	Lrs1	Rrs1	Cr					
51	12.0.8	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltr	Rtr	Lhr	Rhr	Lrs1	Rrs1				
52	12.1.8	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltr	Rtr	Lhr	Rhr	Lrs1	Rrs1	Cr			
53	12.0.10	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Lhr	Rhr	Lrs1	Rrs1		
54	12.1.10	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Lhr	Rhr	Lrs1	Rrs1	Cr	
55	13.0.4	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Lrs1	Rrs1	Ltr	Rtr	Lsc	Rsc						
56	13.0.4h	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Lrs1	Rrs1	Lhr	Rhr	Lsc	Rsc						
57	13.1.4	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Lrs1	Rrs1	Ltr	Rtr	Lsc	Rsc						
58	13.1.4h	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Lrs1	Rrs1	Lhr	Rhr	Lsc	Rsc						
59	13.0.6	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Lrs1	Rrs1	Lsc	Rsc				
60	13.0.6h	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltm	Rtm	Lhr	Rhr	Lrs1	Rrs1	Lsc	Rsc				
61	13.1.6	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Lrs1	Rrs1	Lsc	Rsc				
62	13.1.6h	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltm	Rtm	Lhr	Rhr	Lrs1	Rrs1	Lsc	Rsc				
63	13.0.8	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltr	Rtr	Lhr	Rhr	Lrs1	Rrs1	Lsc	Rsc		
64	13.1.8	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltr	Rtr	Lhr	Rhr	Lrs1	Rrs1	Lsc	Rsc		
65	13.0.10	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Lhr	Rhr	Lrs1	Rrs1	Lsc	Rsc
66	13.1.10	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Lhr	Rhr	Lrs1	Rrs1	Lsc	Rsc
67	14.0.4	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Lrs1	Rrs1	Ltr	Rtr	Lsc	Rsc						
68	14.0.4h	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Lrs1	Rrs1	Lhr	Rhr	Lsc	Rsc						
69	14.1.4	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Lrs1	Rrs1	Ltr	Rtr	Lsc	Rsc	Cr					
70	14.1.4h	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Lrs1	Rrs1	Lhr	Rhr	Lsc	Rsc	Cr					
71	14.0.6	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Lrs1	Rrs1	Lsc	Rsc				
72	14.0.6h	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltm	Rtm	Lhr	Rhr	Lrs1	Rrs1	Lsc	Rsc				
73	14.1.6	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Lrs1	Rrs1	Lsc	Rsc	CR			
74	14.1.6h	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltm	Rtm	Lhr	Rhr	Lrs1	Rrs1	Lsc	Rsc	Cr			
75	14.0.8	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltr	Rtr	Lhr	Rhr	Lrs1	Rrs1	Lsc	Rsc		
76	14.1.8	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltr	Rtr	Lhr	Rhr	Lrs1	Rrs1	Lsc	Rsc	Cr	
77	14.0.10	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Lhr	Rhr	Lrs1	Rrs1	Lsc	Rsc
78	15.0.4	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Lrs1	Rrs1	Ltr	Rtr	Lsc	Rsc	Ls1	Rs1				
79	15.0.4h	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Lrs1	Rrs1	Lhr	Rhr	Lsc	Rsc	Ls1	Rs1				
80	15.1.4	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Lrs1	Rrs1	Ltr	Rtr	Lsc	Rsc	Ls1	Rs1				
81	15.1.4h	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Lrs1	Rrs1	Lhr	Rhr	Lsc	Rsc	Ls1	Rs1				
82	15.0.6	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Lrs1	Rrs1	Lsc	Rsc	Ls1	Rs1		
83	15.0.6h	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltm	Rtm	Lhr	Rhr	Lrs1	Rrs1	Lsc	Rsc	Ls1	Rs1		
84	15.1.6	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Ltm	Rtm	Ltr	Rtr	Lrs1	Rrs1	Lsc	Rsc	Ls1	Rs1		
85	15.1.6h	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltm	Rtm	Lhr	Rhr	Lrs1	Rrs1	Lsc	Rsc	Ls1	Rs1		
86	15.0.8	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltr	Rtr	Lhr	Rhr	Lrs1	Rrs1	Lsc	Rsc	Ls1	Rs1
87	15.1.8	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Ltf	Rtf	Ltr	Rtr	Lhr	Rhr	Lrs1	Rrs1	Lsc	Rsc	Ls1	Rs1
88	16.0.4	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Lrs1	Rrs1	Ltr	Rtr	Lsc	Rsc	Ls1	Rs1				
89	16.0.4h	L	R	C	Cr	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Lrs1	Rrs1	Lhr	Rhr	Lsc	Rsc	Ls1	Rs1				
90	16.1.4	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf	Lrs1	Rrs1	Ltr	Rtr	Lsc	Rsc	Ls1	Rs1	Cr			
91	16.1.4h	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Lh	Rh	Lrs1	Rrs1	Lhr	Rhr	Lsc	Rsc	Ls1	Rs1	Cr			

Table 1a Additional loudspeaker configurations available for Dolby Atmos listening rooms in single-user (24ch) mode.

18.3 *DTS:X listening rooms loudspeaker configurations (12ch)*

		Line Output Channel Number															
#	**Mode**	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**	**14**	**15**	**16**
1	2.0	L	R														
2	2.1	L	R		SW												
3	2.2	L	R		SW					SW2							
4	3.0	L	R	C													
5	3.1	L	R	C	SW												
6	3.2	L	R	C	SW					SW2							
7	5.0	L	R	C		Ls	Rs										
8	5.1	L	R	C	SW	Ls	Rs										
9	5.2	L	R	C	SW					SW2							
10	5.0.2f	L	R	C		Ls	Rs					Ltf	Rtf				
11	5.1.2f	L	R	C	SW	Ls	Rs					Ltf	Rtf				
12	5.2.2f	L	R	C	SW	Ls	Rs			SW2		Ltf	Rtf				
13	5.0.2m	L	R	C		Ls	Rs							Ltm	Rtm		
14	5.1.2m	L	R	C	SW	Ls	Rs							Ltm	Rtm		
15	5.2.2m	L	R	C	SW	Ls	Rs			SW2				Ltm	Rtm		
16	5.0.2h	L	R	C		Ls	Rs					Lh	Rh				
17	5.1.2h	L	R	C	SW	Ls	Rs					Lh	Rh				
18	5.2.2h	L	R	C	SW	Ls	Rs			SW2		Lh	Rh				
19	5.0.4t	L	R	C		Ls	Rs					Ltf	Rtf			Ltr	Rtr
20	5.1.4t	L	R	C	SW	Ls	Rs					Ltf	Rtf			Ltr	Rtr
21	5.2.4t	L	R	C	SW	Ls	Rs			SW2		Ltf	Rtf			Ltr	Rtr
22	5.0.4h	L	R	C		Ls	Rs					Lh	Rh			Lhs	Rhs
23	5.1.4h	L	R	C	SW	Ls	Rs					Lh	Rh			Lhs	Rhs
24	5.2.4h	L	R	C	SW	Ls	Rs			SW2		Lh	Rh			Lhs	Rhs
25	7.0	L	R	C		Lss	Rss	Lb	Rb								
26	7.1	L	R	C	SW	Lss	Rss	Lb	Rb								
27	7.2	L	R	C	SW	Lss	Rss	Lb	Rb	SW2							
28	7.0.2f	L	R	C		Lss	Rss	Lb	Rb			Ltf	Rtf				
29	7.1.2f	L	R	C	SW	Lss	Rss	Lb	Rb			Ltf	Rtf				
30	7.2.2f	L	R	C	SW	Lss	Rss	Lb	Rb	SW2		Ltf	Rtf				
31	7.0.2m	L	R			Lss	Rss	Lb	Rb					Ltm	Rtm		
32	7.1.2m	L	R		SW	Lss	Rss	Lb	Rb					Ltm	Rtm		
33	7.2.2m	L	R		SW	Lss	Rss	Lb	Rb	SW2				Ltm	Rtm		
34	7.0.2h	L	R			Lss	Rss	Lb	Rb			Lh	Rh				
35	7.1.2h	L	R		SW	Lss	Rss	Lb	Rb			Lh	Rh				
36	7.1.2h	L	R		SW	Lss	Rss	Lb	Rb	SW2		Lh	Rh				
37	7.0.4t	L	R	C		Lss	Rss	Lb	Rb			Ltf	Rtf			Ltr	Rtr
38	7.1.4t	L	R	C	SW	Lss	Rss	Lb	Rb			Ltf	Rtf			Ltr	Rtr
39	7.2.4t	L	R	C	SW	Lss	Rss	Lb	Rb	SW2		Ltf	Rtf			Ltr	Rtr
40	7.0.4h	L	R	C		Lss	Rss	Lb	Rb			Lh	Rh			Lhs	Rhs
41	7.1.4h	L	R	C	SW	Lss	Rss	Lb	Rb			Lh	Rh			Lhs	Rhs
42	7.2.4h	L	R	C	SW	Lss	Rss	Lb	Rb	SW2		Lh	Rh			Lhs	Rhs
43	9.0.2f	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf				
44	9.1.2f	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf				
45	9.2.2f	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw	Ltf	Rtf			SW2	
46	9.0.2m	L	R	C		Lss	Rss	Lb	Rb	Lw	Rw			Ltm	Rtm		
47	9.1.2m	L	R	C	SW	Lss	Rss	Lb	Rb	Lw	Rw			Ltm	Rtm		

48	*9.2.2m*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*	*Lb*	*Rb*	*Lw*	*Rw*			*Ltm*	*Rtm*	*SW2*	
49	*9.0.2h*	*L*	*R*	*C*		*Lss*	*Rss*	*Lb*	*Rb*	*Lw*	*Rw*	*Lh*	*Rh*				
50	*9.1.2h*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*	*Lb*	*Rb*	*Lw*	*Rw*	*Lh*	*Rh*				
51	*9.2.2h*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*	*Lb*	*Rb*	*Lw*	*Rw*	*Lh*	*Rh*			*SW2*	

Table 2: Supported loudspeaker configurations for DTS:X listening rooms in dual-user mode (12ch).

18.4 *PCM listening rooms loudspeaker configurations (16ch)*

		Line Output Channel Number															
#	***Mode***	***1***	***2***	***3***	***4***	***5***	***6***	***7***	***8***	***9***	***10***	***11***	***12***	***13***	***14***	***15***	***16***
1	*2.0*	*L*	*R*														
2	*2.2.2t*	*L*	*R*		*SW*					*SW2*		*Ltf*	*Rtf*				
3	*2.2.2h*	*L*	*R*		*SW*					*SW2*		*Lh*	*Rh*				
4	*3.0*	*L*	*R*	*C*													
5	*3.2.2t*	*L*	*R*	*C*						*SW2*		*Ltf*	*Rtf*				
6	*3.2.2h*	*L*	*R*	*C*	*SW*					*SW2*		*Lh*	*Rh*				
7	*4.0*	*L*	*R*			*Ls*	*Rs*										
8	*4.2.2t*	*L*	*R*		*SW*	*Ls*	*Rs*			*SW2*		*Ltf*	*Rtf*				
9	*4.2.2h*	*L*	*R*		*SW*	*Ls*	*Rs*			*SW2*		*Lh*	*Rh*				
10	*4.2.4t*	*L*	*R*		*SW*					*SW2*		*Ltf*	*Rtf*			*Ltr*	*Rtr*
11	*4.2.4h*	*L*	*R*			*Ls*	*Rs*			*SW2*		*Lh*	*Rh*			*Lhs*	*Rhs*
12	*5.0*	*L*	*R*	*C*		*Ls*	*Rs*										
13	*5.1*	*L*	*R*	*C*	*SW*	*Ls*	*Rs*										
14	*5.2.2t*	*L*	*R*	*C*	*SW*	*Ls*	*Rs*			*SW2*		*Ltf*	*Rtf*				
15	*5.2.2h*	*L*	*R*	*C*	*SW*	*Ls*	*Rs*			*SW2*		*Lh*	*Rh*				
16	*5.2.4t*	*L*	*R*	*C*	*SW*	*Ls*	*Rs*			*SW2*		*Ltf*	*Rtf*			*Ltr*	*Rtr*
17	*5.2.4h*	*L*	*R*	*C*		*Ls*	*Rs*			*SW2*		*Lh*	*Rh*			*Lhs*	*Rhs*
18	*5.2.6t*	*L*	*R*	*C*	*SW*	*Ls*	*Rs*			*SW2*		*Ltf*	*Rtf*	*Ltm*	*Rtm*	*Ltr*	*Rtr*
19	*6.0*	*L*	*R*			*Lss*	*Rss*	*Lb*	*Rb*								
20	*7.0*	*L*	*R*	*C*		*Lss*	*Rss*	*Lb*	*Rb*								
21	*7.1*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*	*Lb*	*Rb*								
22	*7.2.2t*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*			*SW2*		*Ltf*	*Rtf*				
23	*7.2.2h*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*			*SW2*		*Lh*	*Rh*				
24	*7.2.4t*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*			*SW2*		*Ltf*	*Rtf*			*Ltr*	*Rtr*
25	*7.2.4h*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*			*SW2*		*Lh*	*Rh*			*Lhs*	*Rhs*
26	*7.2.6t*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*	*Lb*	*Rb*	*SW2*		*Ltf*	*Rtf*	*Ltm*	*Rtm*	*Ltr*	*Rtr*
27	*9.0*	*L*	*R*	*C*		*Lss*	*Rss*	*Lb*	*Rb*	*Lw*	*Rw*						
28	*9.2.2t*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*	*Lb*	*Rb*	*Lw*	*Rw*	*Ltf*	*Rtf*	*SW2*			
29	*9.2.4t*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*	*Lb*	*Rb*	*Lw*	*Rw*	*Ltf*	*Rtf*	*SW2*		*Ltr*	*Rtr*
30	*9.1.6t*	*L*	*R*	*C*	*SW*	*Lss*	*Rss*	*Lb*	*Rb*	*Lw*	*Rw*	*Ltf*	*Rtf*	*Ltm*	*Rtm*	*Ltr*	*Rtr*
31	*Custom#1*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*
32	*Custom#2*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*
33	*Custom#3*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*
34	*Custom#4*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*	*Any*

Table 3: PCM listening rooms loudspeaker configurations in dual-user mode (16ch)

19 Appendix B: Loudspeaker names and labels

19.1 Loudspeaker names and labels with default azimuth and elevation angles

#	Label	Name	Azimuth	Elevation	Atmos	DTS:X	PCM	PCM Custom
1	L	Left	-30	0	Y	Y	Y	Y
2	R	Right	30	0	Y	Y	Y	Y
3	C	Centre	0	0	Y	Y	Y	Y
4	SW	Subwoofer	-44	0	Y	Y	Y	Y
5	Ls	Left surround	-100	0	Y	Y	Y	Y
6	Rs	Right surround	100	0	Y	Y	Y	Y
7	Lb	Left back	-120	0	Y	Y	Y	Y
8	Rb	Right back	120	0	Y	Y	Y	Y
9	Lss	Left side surround	-90	0	Y	Y	Y	Y
10	Rss	Right side surround	90	0	Y	Y	Y	Y
11	Cr	Centre rear	180	0				Y
12	SW2	Subwoofer 2	44	0	Y	Y	Y	Y
13	Lw	Left wide	-60	0	Y	Y	Y	Y
14	Rw	Right wide	60	0	Y	Y	Y	Y
15	Lbs	Left back surround	-164	0				Y
16	Rbs	Right back surround	164	0				Y
17	Lc	Left centre	-20	0				Y
18	Rc	Right centre	20	0				Y
19	Lg	Left ground	-30	-20				Y
20	Rg	Right ground	30	-20				Y
21	Cg	Centre ground	0	-20				Y
22	Ch	Centre height	0	40				Y
23	Chr	Centre height rear	180	40				Y
24	T	Top	0	90				Y
25	Lh	Left height	-30	40		Y	Y	Y
26	Rh	Right height	30	40		Y	Y	Y
27	Lhs	Left height side	-90	40		Y	Y	Y
28	Rhs	Right height side	90	40		Y	Y	Y
29	Lhr	Left height rear	-120	40				Y
30	Rhr	Right height rear	120	40				Y
31	Ltf	Left top front	-44	60	Y	Y	Y	Y
32	Rtf	Right top front	44	60	Y	Y	Y	Y
33	Ltm	Left top mid	-90	60	Y	Y	Y	Y
34	Rtm	Right top mid	90	60	Y	Y	Y	Y
35	Ltr	Left top rear	-134	60	Y	Y	Y	Y
36	Rtr	Right top rear	134	60	Y	Y	Y	Y
37	Lsc	Left side centre	-10	0				Y
38	Rsc	Right side centre	10	0				Y
39	Ls1	Left surround 1	-74	0				Y
40	Rs1	Right surround 1	74	0				Y
41	Lrs1	Left rear surround 1	-110	0				Y
42	Rrs1	Right rear surround 1	110	0				Y
43	Lrs2	Left rear surround 2	-150	0				Y
44	Rrs2	Right rear surround 2	150	0				Y

45	*Lhw*	*Left height wide*	*-60*	*40*				*Y*
46	*Rhw*	*Right height wide*	*60*	*40*				*Y*
47	*Lhs1*	*Left height side 1*	*-110*	*40*				*Y*
48	*Rhs1*	*Right height side 2*	*-110*	*40*				*Y*
49	*Lbg*	*Left back ground*	*-120*	*-20*				*Y*
50	*Rbg*	*Right back ground*	*120*	*-20*				*Y*

Table 4: Loudspeaker names, labels and ID numbers

Notes

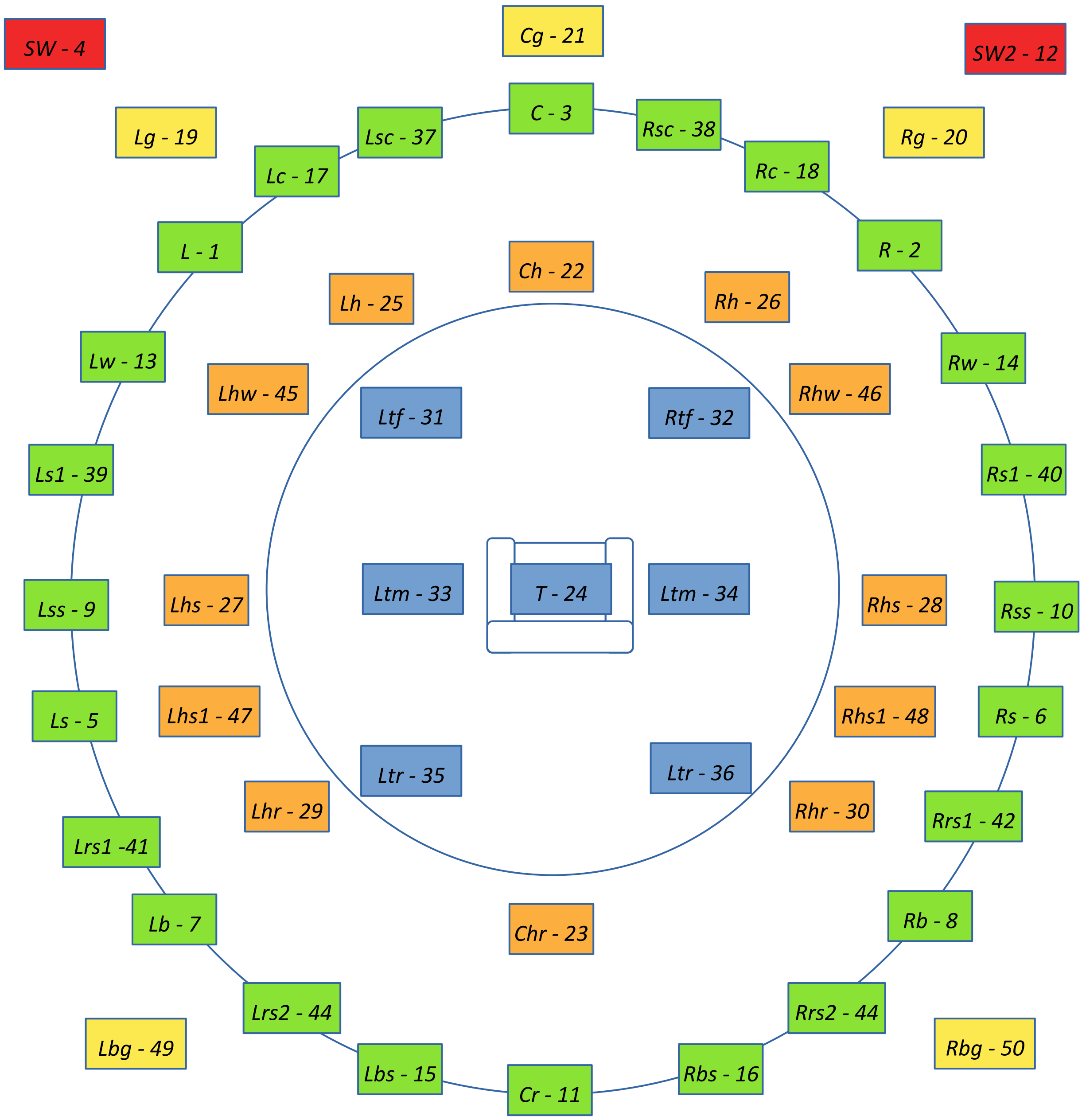
1. *The azimuth and elevation angles are default values – the angles can be changed to match a physical loudspeaker layout during a PRIR measurement.*
2. *The Dolby Atmos loudspeaker labels are limited to those in Table 1.*
3. *The DTS:X loudspeaker labels are limited to those in Table 2.*
4. *The PCM loudspeaker labels are limited to those in Table 3 (modes 1 to 30).*
5. *The PCM Custom modes can use all the loudspeaker labels (Table 3 (modes 31 to 34)).*

19.2 *Graphical representation of loudspeakers in the Speaker Map display of the A16*

SW 4	L 1	Lc 17	Lsc 37	C 3	Rsc 38	Rc 18	R 2	SW2 12
Lw 13	Lhw 45	Lh 25		Ch 22		Rh 26	Rhw 46	Rw 14
Ls1 39		Ltf 31	Lg 19	Cg 21	Rg 20	Rtf 32		Rs1 40
Lss 9	Lhs 27	Ltm 33		T 24		Rtm 34	Rhs 28	Rss 10
Ls 5	Lhs1 47	Ltr 35				Rtr 36	Rhs1 48	Rs 6
Lrs1 41	Lhr 29	Lbg 49		Chr 23		Rbg 50	Rhr 30	Rrs1 42
	Lb 7	Lrs2 43	Lbs 15	Cr 11	Rbs 16	Rrs2 44	Rb 8	

- *Ear level*
- *Height*
- *Overhead*
- *Ground*
- *Subwoofers*

19.3 *Graphical representation of speaker positions: loudspeaker names and ID numbers*



20 Appendix C: Calibrating the magnetic sensor in the head-top device

***Home Page** menu: **Apps** menu: **Calibrate head tracker** menu:*

The A16 head-tracker has three motion detecting sensors, inertial, magnetic, and optical. The inertial sensor operates continuously but it can also be stabilised, in a narrow central window, by either the magnetic or optical sensors.

If the magnetic sensor is chosen for stabilisation, it is recommended that the magnetic sensor be calibrated in the headphone listening location in the room, mounted correctly on the headphones, and ideally in the same 'head-space'.

The calibration routine aims to measure the strength and direction of the magnetic field in all directions around the head-tracker, and then uses this data to determine its orientation with respect to this external field.

The magnetic sensor may need to be re-calibrated if magnetisable objects are moved within the seating location.

Calibration procedure

Step 1. Mount the head-top on the headphones and connect the head-top as normal to the HT input A on the front panel of the A16.

Step 2. Move to the seating location where you expect to use your headphones.

Step 3. Start the calibration routine by selecting the Calibrate magnetics option - then push the ENTER key on the remote.

NOTE: The word 'calibrating' will become visible on the menu line and the head-top LED with turn ORANGE

Step 4. Holding the headphones in your hands, and ideally in the same headspace that the headphones would occupy when being used, tumble them around in complete 360-degree rotations in all orientations for a few minutes. The tumbling action is to allow the magnetic sensor to measure the field in ALL directions – i.e. not just in the horizontal rotational directions.

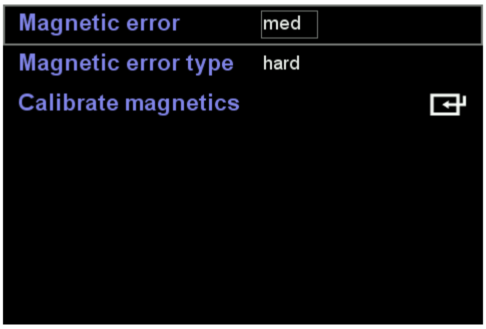
Step 5. Calibration is complete when the 'calibrating' word disappears in the display, and the LED on the head-top turns GREEN.

Step 6. Ensure that Stabilisation is set to magnetics in the HT Settings menu.

***Home page** menu: **Settings** menu: **System** menu: **HT Settings** menu: **Stabilisation** option*

Step 7. Finally, wearing the headphones, navigate to the Azimuth angles display of any Preset Speaker Map, look directly at the centre speaker, and push the button on top of the head-top to set zero degrees for the magnetic sensor.

With stabilisation set to magnetic, pushing the button on the head-top will re-set the zero degrees point of the magnetic sensor. Within the stabilisation window the magnetic sensor will reinforce the inertial sensor.



Calibrate head tracker menu

Magnetic error

Can be set to LOW or MED.

If set to MED the calibration routine will be less accurate but will take a shorter time to complete.

If set to LOW the calibration routine will be more accurate but will take a longer time to complete.

Magnetic error type

Can be set to HARD or HARD+SOFT

If set to HARD the calibration will be less accurate but will take a shorter time to complete.

If set to HARD+SOFT the calibration will be more accurate but will take a longer time to complete.

Calibrate magnetics

Use the ENTER key to run the magnetics calibration routine as outlined above.

NOTE: If calibration appears to be taking too long to complete, use the BACK key to terminate the calibration routine. Change the magnetic error or magnetic error type to reduce the calibration time and run the calibration routine again.

21 Appendix D: Setting up the head-tracker

The head-tracker consists of two parts:

1. A HEAD-TOP part that is mounted on top of a pair of headphones and is connected to one of the two HT ports on the front panel of the A16. The head-top has three tracking sensors, inertial magnetic and optical, and from these is calculated a head-tracking angle that is transmitted back to the A16.

2. A SET-TOP part that can be mounted on top of a monitor or speaker and is connected to the Set-Top port on the rear panel of the A16. The set-top has two functions: it transmits an IR pulse of light that may be used as a zero-degree reference angle; it also indicates the approximate headtracking angle using a grid of LEDs.

The Head-Top device that is mounted on the headphones

The Set-Top device that can be mounted in a central location such as a TV or video monitor

Theory of Operation

The SVS virtualisation algorithm uses head-tracking data to lock the position of virtual loudspeakers to an external location when the listener's head is rotated. If head-tracking data is not available the virtualisation still occurs, but the position of the virtual speakers will move as the listener's head rotates. The head-top has three positional sensors, inertial, magnetic, and optical. The inertial sensor is the primary means of determining head orientation but, to mitigate inertial drift, it can be optionally stabilised around the important central listening location (zero degrees), using either the magnetic or optical sensors. Within the stabilisation window the magnetic or optical sensors act to pull the inertial sensor to a 'corrected' heading. Outside the stabilisation window the inertial sensor operates by itself, but the heading angle leaks exponentially towards zero degrees. The user can switch between either method of stabilisation - advantages and disadvantages of each are given below.

Advantages of magnetic stabilisation
1. Does not need the set-top device to be connected.
2. Once calibrated the zero-degrees heading remains stable indefinitely.
3. Not affected by stray IR light sources.
4. Operates for single or dual users.

Disadvantages of magnetic stabilisation
1. May need to be re-calibrated if the seating position changes or magnetisable objects are moved in the room.
2. The zero-degrees orientation may need to be re-set for each listening session – using the push button on the head-top.

Advantages of optical stabilisation
1. Very stable and does not need to be calibrated. The zero-degrees orientation does not need to be re-set – the external set-top always defines the zero-degree orientation.
2. Not affected by external magnetised objects.

Disadvantages of optical stabilisation
1. Needs the set-top to be connected and in a centrally visible location at the zero-degree mark – often on top of the video monitor.
2. The set-top IR source may be visually distracting
3. The optical sensor in the head-top can be fooled by stray IR sources of light – both inside and outside the stabilisation window.
4. The reference IR pulse from the set-top may interfere with the remote control – operation with the remote control may feel 'laggy'.
5. Currently only operates for a single user.

Setting up and configuring the head-tracker.

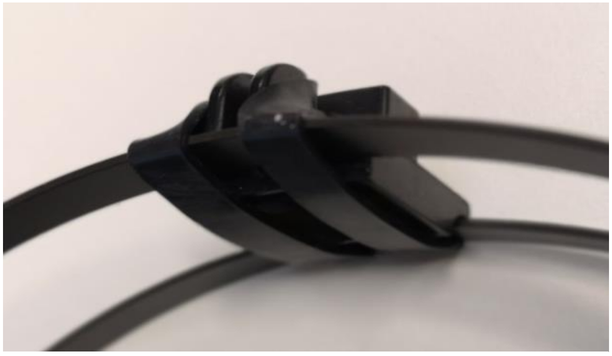
This consists of:

Step 1. Mounting the head-top to a pair of headphones and connecting this to the HT port on the front panel of the A16.
Step 2. Mounting the set-top in some central location (on top of the TV) and connecting this to the Set-top port on the rear panel of the A16.
Step 3. Configuring headtracking menu options.

Mounting the Head-top device

1. Connect the head-top mounting clip to the centre of the headband using one of the rubber bands.

Mounting the head-top clip to headphones using a rubber band



Mounting the head-top clip to the centre of the headphone band

2. Mount the head-top into the clip, with the black IR optical window facing forwards.

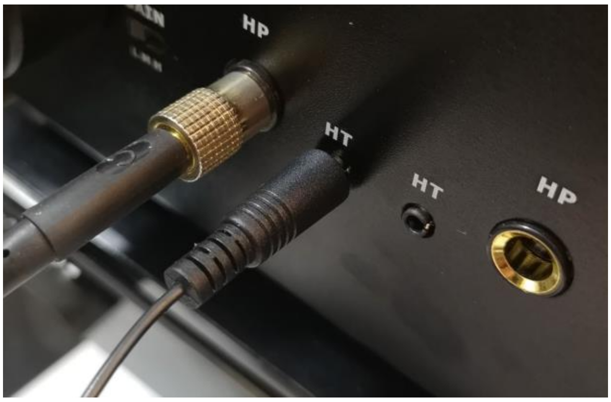
Mounting the head-top to the clip, front window facing forwards

Connecting the 90-degree connector of the HT cable to the head-top device

3. Connect some wire restraining clips to the headphone cord and connect the head-top cable along the headphone cord with these clips. These clips attempt to keep the head-top cable attached to the headphone cable.

4. Connect the 90-degree connector to the head-top and the straight connector to the appropriate HT port on the front panel of the A16.

Connecting the head-top cable to the headphone cable using clips

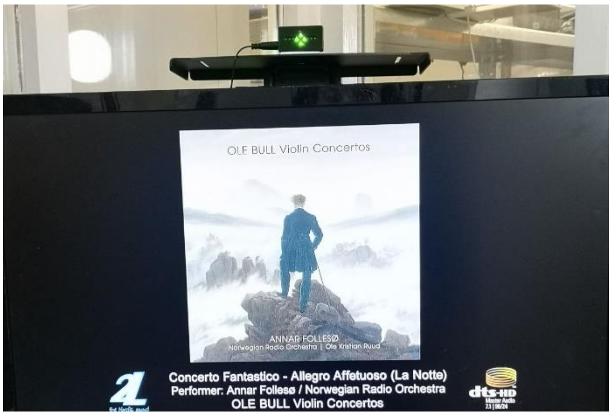


Connect the Head-top cable to appropriate HT port on the front panel of the A16

Immediately after connection, the LED on the head-top will momentarily turn RED while it checks the validity of the internal head-tracking program. A few seconds later it will flash 10 times in quick succession. Ten green flashes imply all is well. Ten red flashes imply the system has lost its calibration data. In this case the head tracker must be recalibrated as described in Appendix G. Following ten green flashes, the LED will turn red while it warms up. Warmup will typically take around 10-15 minutes. It is important that head tracker remain stationary during this time. The LED will turn steady green once the temperature has stabilised.

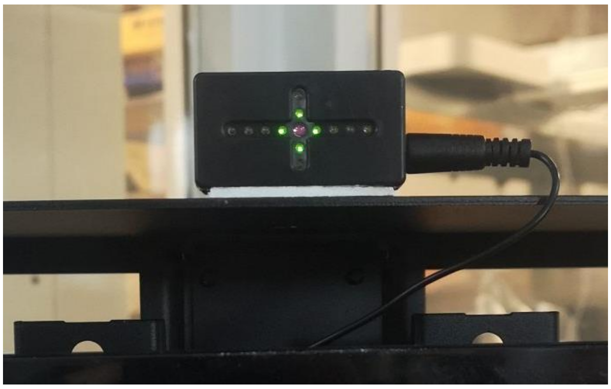
Setting up and configuring the head-tracker: the set-top device

1. Mount the set-top device in a central location using double-sided tape and connect it to the Set-Top port on the rear panel of the A16 using the set-top cable. The cable can be connected to either side of the set top. Immediately after connection to the A16 all the LEDs on the set-top unit will flash once to indicate they are working.



Mount the set-top in a central visible location

Connect the set-top cable to the Set-top port on the back panel of the A16



The four central LEDs ON indicates looking centre with the head level.

2. Navigate to the HT Settings menu

*Move to: **Home page** menu: **Settings** menu: **System** menu: **HT Settings** menu: **Stabilisation** option*

3. Set the stabilisation mode to **OPTICAL**.

4. Point the front face of the connected head-top device towards the set-top. The set-top should quickly indicate a zero-degree heading. At a head-top heading of zero degrees azimuth and zero degrees elevation the four central LEDs (around the IR LED) should all be ON. Movement of the headphones will now be reported by the LEDS on the set top – the outermost LEDs indicate a rotation of +/-30 degrees. The actual head-tracking angle for user A and B can be viewed while in the Speaker Map page using the HT button on the remote – the HT button toggles the display ON and OFF.

Pressing HT key while in the speaker map page

NOTE: Head tracker operation requires that both the Presets for User A and User B are loaded and active.

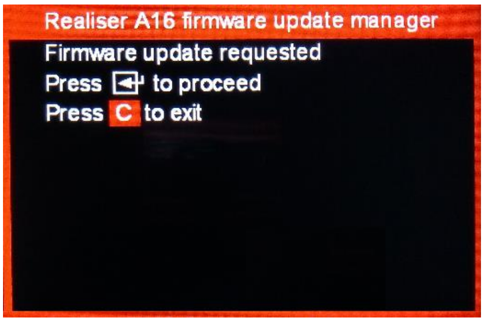
22 Appendix E: Updating the Realiser A16 firmware

Updating the firmware of your A16 is only necessary if your A16's current firmware is older than the latest downloadable version. The current revision of your firmware is found in 'Updates/About' accessed via the 'Settings' page as described below in step 6. If an update is required, please begin with step 1.

STEP 1. The new firmware for the Realiser A16 is uploaded through the micro-SD card slot on the front panel. First, obtain a micro-SD card (commonly 8 or 16 GB) and ensure it is formatted as FAT32. Second, create a 'realiser' folder in the root directory and copy the firmware file FIRMA001.SVS into the realiser folder. Insert this micro-SD card into the slot on the front of your A16.

STEP 2. Power up the A16 ensuring the power indicator LED is steady green. You can power it up using the remote control or by momentarily depressing either User A or User B volume knobs. Now turn off the A16 by pushing in and holding in the User A volume knob for at least 3 seconds. The LCD screen will switch off and the power indicator LED will turn red. Release the User A volume knob.

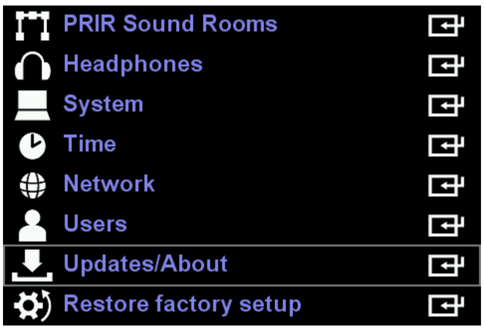
STEP 3. Push in and hold in the User B volume knob and, simultaneously, push in and release the User A volume knob. Then release the User B volume knob. The action of holding in B and depressing A activates the firmware update manager as shown below. The power indicator LED will also be blinking green.



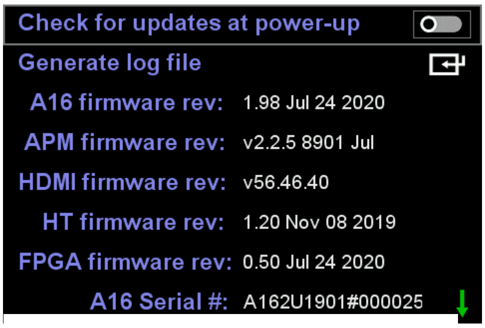
Firmware update manager menu.

STEP 4. Using the remote control, press the ENTER key twice to begin the firmware update. The A16 will enter a long period (30-35 minutes) of authenticating the software, installing the software, and rebooting. When the unit first reboots it will begin updating the firmware for the individual hardware modules (FPGA, APM Decoder etc). After the individual firmware modules have been reprogrammed the unit will reboot using the normal power-up sequence to the Speaker Map display for User A.

STEP 5. The firmware update is now complete. To confirm the firmware update was a success please check the revision numbers displayed in UPDATES/ABOUT accessed via the SETTINGS page. The firmware update PDF that is supplied with the firmware file will list the revisions numbers you should expect to see.

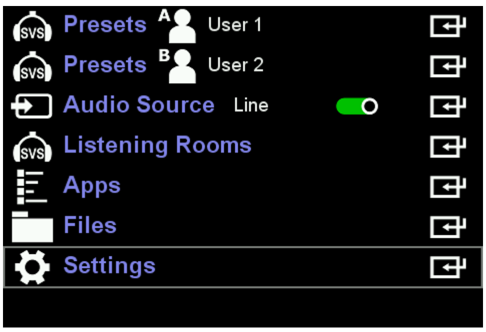


Firmware Updates/About.



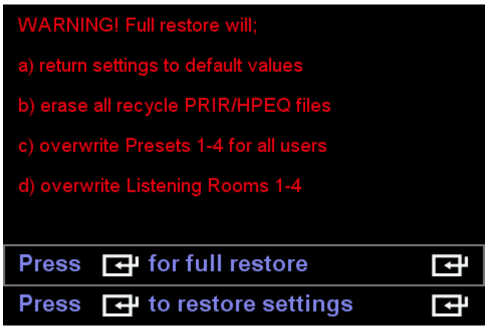
Firmware revision page

STEP 5. *Once you are happy the firmware is up to date, a full restore must be run to properly initialize new menu and Preset features First press BACK and then navigate to 'Settings' and press ENTER.*



Navigate to 'Restore factory setup' and press ENTER again.

Select 'full restore' and press ENTER again.



The full restore will take approximately 10 minutes to complete, thereafter the A16 will automatically return to the User A speaker map display.

23 Appendix F: Updating the A16 Head tracker Firmware

Updating the Head Tracker

The HT firmware updater programs a head tracker firmware file (FIRMHT01.SVS), held internal to the A16, into the head tracker via the HT cable. This internal firmware file is normally loaded to the A16 as part of a general A16 firmware update and as such is invisible to the user. However, on triggering the HT firmware update process the A16 will first check if such a file exists in the REALISER directory on the micro-SD card. If the file is found, then the internal firmware file is overwritten with this file, prior to starting the update process. If not, then the original internal file remains unaltered and is used to update the head tracker software.

Step 1) Plug HT into the User A side.

Step 2) Optionally insert the micro-SD card with FIRMHT01.SVS in the REALISER directory.

Step 3) Go to Settings>System>HT settings>Update HT firmware and press enter to start the update. This causes the internal firmware file to be flashed to the head tracker. The A16 display and the LED atop the HT report the progress of the update procedure in the following order.

	A16 display status	***HT LED status***
1	*Loading application*	*No change*
2	*Connecting to HT (may be brief)*	*Red+Green steady*
3	*Update in progress*	*Red flashing*
4	*Authenticating update*	*Red+Green steady*
5	*HT update validated*	*Temperature calibration procedure begins*

Step 4) Once the A16 status displays 'HT update validated', press the back key once to exit the update routine. The A16 will be unresponsive momentarily while the Room Presets for User A and User B are reloaded.

Step 5) Go to Settings>Updates/About and check that the reported HT firmware revision is as expected.

If the HT firmware update fails to follow the order shown above, for example it stalls at 'Connecting to HT', or alternates between 'Loading application' and 'Connecting to HT' then the communication with the HT has failed. First fully exit the Update HT firmware menu and unplug the head tracker. Then restart beginning step one.

Internal Head Tracker temperature warm-up following power-up

The head tracking measurements use an internal 3-axis gyroscope as the core 6DOF IMU engine. Gyroscopes must calibrate their output against temperature for accurate tracking. To maximise performance immediately after power-up the head tracker disables the gyroscope calculations, indicated by the RED led, until the internal temperature has reached steady state and exits this state once the head tracker has been deemed stationary for a minimum of 1 second thereafter, indicated by the GREEN led. This warmup typically takes 5-10 minutes from cold, and it is recommended that the head tracker remain stationary until the LED indicates GREEN. For non-stabilised and magnetically stabilised head tracking modes, no heading angle is output until the GREEN led is active. For optically stabilised head tracking a raw optical heading angle is output while the RED led is active.

Automatic Head Tracker temperature calibration

Apart from the initial temperature stabilisation on power-up, temperature calibration is also periodically undertaken every second the head tracker remains stationary, both during the warm-up phase (RED led) and beyond (GREEN led), as indicated by a blink on the led. Due to the high sensitivity of the sensors used in the tracker, such calibrations will only occur when the head tracker and/or headphone are completely stationary. Movement or vibration of any kind will prevent such calibrations.

During use it is possible for the internal temperature to deviate from that of the last calibration. If this exceeds 3 degrees, the LED will change from GREEN to GREEN+RED. To recalibrate the user simply needs to immobilise the head tracker for 1 second, after which the LED will return GREEN.

Typical operating steps

Step 1) Connect the head tracker to the A16 and turn the A16 on. The LED atop the head tracker will initially turn RED and will blink RED every second if stationary.

Step 2) Keeping the head tracker stationary, wait until the blinking led turns GREEN. This may take 5 to 10 minutes.

Step 3) Once blinking GREEN the head tracker is ready for use.

Step 4) If at any time the LED turns GREEN+RED, keep the headphone stationary for 1 second (led will return to GREEN) and continue using.

Running the Head Tracker without Stabilisation

With stabilisation disabled the head tracker heading angles are generated solely using the internal gyroscopes. The set-top is not required for this mode although it can still be connected to display user A head tracking. Stabilisation window settings are not used for this mode of operation. Despite proper temperature calibration, gyroscopes exhibit drift over time, that is, the calculated heading will slowly change even while the head tracker is kept stationary. Gyroscopes are also prone to drifting because of mechanical shock, vibration and lateral motion. To combat these sources of drift the A16 head tracker applies leakage to the calculations that forces the angle to slowly converge to 0-degree heading. This leakage is referred to as 'Drift compensation' in the A16 HT settings menu. Two leakage settings are possible, 'fast' and 'slow'. Fast causes the heading to converge to 0 degrees at a rate of 1 degree per second (60 degrees per minute), while 'slow' reduces this to 0.25 degrees per second (15 degrees per minute). This leak in the heading occurs regardless of the position of the head tracker, meaning that even if the listener removes the headphones and placed them on a table, the heading will continue to converge to zero.

The effect of leakage is to cause zero-degree azimuth to align with the average direction of the headphone. The listener can still turn their head left or right and experience a stable soundstage since the convergence is relatively slow compared to the rate of head turning. But leakage only makes sense if the user spends most of their listening time looking in one direction, for example watching a TV or computer screen, or simply sitting in a chair listening to music.

High leakage is recommended when the head tracker is likely to experience significant movement during use, say for example, when using the A16 for audio editing at a workstation. Low leakage is recommended for normal listening situations.

With stabilisation disabled it is necessary to manually reset the heading to zero at the beginning of a listening session. Typically, the listener will don the headphones, look in the direction of zero-degree azimuth (look at the centre speaker or TV) and then reset the head tracker. Resetting the head tracker angle can be achieved in two ways. First It can be zeroed using the momentary push button atop the head tracker. It is recommended that the headphone be in an upright position when this switch is depressed. Alternatively, it can be zeroed automatically by tilting the headphone down (greater than 45-degree incline) and then bringing the headphone back to within 10 degrees of upright. The second mode is convenient for demonstrations. However, if the demonstration AB mode is also in use, one needs to be careful to ensure that all AB comparisons are undertaken with the listener looking at the centre speaker (or TV) since the action of taking the headphones off and then on again now also sets the heading to zero.

Using the head tracker without stabilisation is not recommended for HT assisted personalisation measurements due to the convergence caused by the leakage. However, if this mode must be used then the leakage should be set to 'slow'. Alternatively switch the mode to 'optical' stabilisation without connecting the set-top (since no leakage is applied in optical mode and without the set-top the heading calculation reverts to the gyroscope).

Running the Head Tracker using Optical Stabilisation

This mode requires the use of the set-top which typically acts as the reference zero azimuth position (or centre speaker). The head tracking LED display on the front of the set-top shows the heading for user A. Drift compensation is not used in the optical stabilisation mode. The stabilisation window setting controls the optical azimuth angle, either side of the set-top location, over which the optical angle is used to correct drift in the gyroscope azimuth tracking. When operating outside of this window, gyroscope drift simply goes uncorrected. The 'wide' stabilisation window covers an azimuth of 60 degrees either side of the set-top position. The 'narrow' window covers +/-30 degrees. In both cases the optical azimuth angle is used to correct the gyroscope angle only when the head tracker is also within +/-30 degrees of the fully upright (vertical) position. Listeners should therefore ensure the headphone is not positioned on their head at an angle outside this range otherwise gyroscope azimuth drift will go uncorrected even if the head tracker is within the stabilisation window.

As with the 'none' stabilisation head tracker mode, in the 'optical' stabilisation mode the gyroscope heading can be reset to zero using the momentary switch atop the head tracker. However, if the head tracker heading is inside the azimuth stabilisation window, the gyroscope azimuth angle will automatically adapt to the optical heading within fractions of a second following the reset. Resetting the head tracker by tilting does not function in the optical mode.

Optical stabilisation also supports a fixed offset angle mode of operation. This is typically used when it is not possible (or desirable) to mount the set-top in a 0-degree azimuth location. To set the offset angle, place the set-top in the required offset position, point the head tracker 0-degree azimuth while sitting in the listening position, ensuring there is clear line-of-sight between the set-top and the head tracker, and press-and-hold the switch atop the head tracker for 3 seconds (one short blink followed by a long blink on the led). Offset angle values are stored in flash memory within the head tracker and restored automatically on future power-ups. To disable the offset angle mode of operation press-and-hold the switch atop the head tracker for 0.5 seconds (one short blink on the led). It should be stressed that when setting the offset angle, or operating in an offset location in general, the set-top offset angle should not exceed the stabilisation window range. For example, if a 'narrow' window is selected then the set-top should be located well within 30 degrees either side of 0-degree azimuth. For 'wide' it should be well within 60 degrees of 0-degree azimuth.

If the set-top is disconnected or the optical line-of-sight between the set-top and the head tracker interrupted, the head tracking continues to operate seamlessly using only the gyroscope. However, gyroscope drift is no longer corrected, and the head tracker effectively runs in the 'none' stabilisation mode except that no drift leakage is applied.

The optical stabilisation mode is ideal for use with HT assisted personalisation measurements. During such measurements, the look angles are deliberately arranged to alternate either side of 0-degree azimuth thereby causing the head tracker to realign as the head transitions the stabilisation window.

The optical azimuth tracking used in the A16 head tracker is susceptible to false tracking under certain conditions. A pulsing IR transmitter (850nm) in the set-top acts as an optical beacon to the head tracker allowing it to know zero-degree azimuth. If another more-powerful IR source is also in view of the head tracker optics, then this can overload the detection circuits and cause the tracking to lock to this erroneous IR source. Concentrated light sources such as halogen spots emit strongly in the 800-900nm IR band and if placed close to the head tracker can lead to false tracking. The solution is simply to keep such lights out of view of the head tacker. Sunlight is also problematic, and blinds should be kept drawn if optical tracking is in operation and sunlit objects could enter the view of the head tracker.

Running Optical Stabilisation during warm-up

In optical mode a raw optical heading (Optical A8 mode) is output during warm-up. This feature is provided simply to allow head tracked listening to proceed immediately following power-up. Since during warm-up the gyroscope heading is disabled, the native optical heading is only stable while there remains a clear line-of-sight between the set-top and head tracker and the head tracking angle does not exceed +/- 65 degrees. Operation outside this range or interruptions to the IR beam will result in the heading freezing at the last valid angle.

Running the Head Tracker using Magnetic Stabilisation

The set-top is not required for this mode although it can still be connected to display user A head tracking. Drift compensation is not used in the magnetic stabilisation mode. The stabilisation window setting controls the magnetic azimuth angle over which the magnetic angle is used to correct drift in the gyroscope azimuth tracking. When operating outside of this window, gyroscope drift simply goes uncorrected. The 'wide' stabilisation window covers a magnetic azimuth of 20 degrees either side of the zero-azimuth position (set using the momentary switch atop the head tracker). The 'narrow' window covers +/-10 degrees. In both cases the magnetic azimuth angle is used to correct the gyroscope angle only when the head tracker is also within +/-15 degrees of the fully upright (vertical) position. Listeners should therefore ensure the headphone is not positioned on their head at an angle outside this range otherwise gyroscope azimuth drift will go uncorrected even if the head tracker is within the stabilisation window.

Because the magnetic compass uses the earth's magnetic field to measure head tracker azimuth motion, it is necessary for the user to manually set the zero-degree azimuth angle using the switch atop the head tracker. Depressing this switch resets the gyroscope heading and establishes a reference angle for the magnetic compass. Unlike optical stabilisation, this magnetic reference angle must be re-entered using the momentary switch for each new sitting position the listener takes up. Resetting the head tracker by tilting does not function in the magnetic mode.

The magnetic stabilisation mode is not recommended for use with HT assisted personalisation measurements.

The magnetic azimuth tracking used in the A16 head trackers are susceptible to false tracking under certain conditions. Since the compass uses to earth's magnetic field to calculate the heading any distortion of this field will introduce tracking errors. Dynamic and magneto-planar headphones are the obvious source of field distortion due to the presence of magnetic material in their driver construction, as well as nearby unshielded loudspeakers. However, if the head tracker is calibrated atop such headphones and undertaken with the headphone in the listening position, many of these distortion sources can be compensated for in the calibration process. Nonetheless, magnetic interference can vary significantly even over short distances and so accurate tracking using the earth's magnetic field is always going to be problematic in the home environment.

24 Appendix G: Head Tracker Thermal Calibration

If the head tracker signals 10 red flashes on power up or has stopped working all together, corruption of its internal thermal calibration data is the likely cause. The head tracker will not function in this state, and it will be necessary to perform a thermal recalibration. The steps are as follow.

Step 1. Update the head tracker with the latest firmware. See update instructions in this manual.

Step 2. Unplug the head tracker and place it in a freezer for 30 minutes. This ensures the starting temperature is sufficiently low.

Step 3. Have the A16 powered up and situated in a room at normal room temperature (18-35 degrees Celsius) and have both User A and User B presets running.

Step 4. Remove the head tracker from the freezer and immediately connect the head tracker to the HT jack input of User A. Place the head tracker on a solid surface and ensure it does not move or suffer vibration during the warmup period.

Step 5. The head tracker will now slowly warm up. Early in the warmup phase the LED will turn red and begin blinking. This signals the start of the calibration, and this blinking will continue until finally the LED turns green, 10 to 15 minutes later.

Once it has reached its green blinking state, the head tracker is ready to use. On future powerup the head tracker should signal 10 green flashes to indicate the thermal data is valid.

25 Appendix H: Connections

Headphone connections

The headphones are connected by using the headphone jack on the front panel of the Realiser. There are two jacks (1/4-inch 3 Pole) corresponding to User A and User B.

Headphone jacks for User A and User B with a Headphone plugged into User A

Head-tracker and IR reference connection

Detailed pictures of how the head-tracker and IR reference set-top are connected to the Realiser are in chapter 3.

Connecting loudspeakers

To be added …

Speaker installation

To be added …

Speaker configuration

To be added …

Speaker and amplifier connection

To be added …

Connecting a TV

To be added …

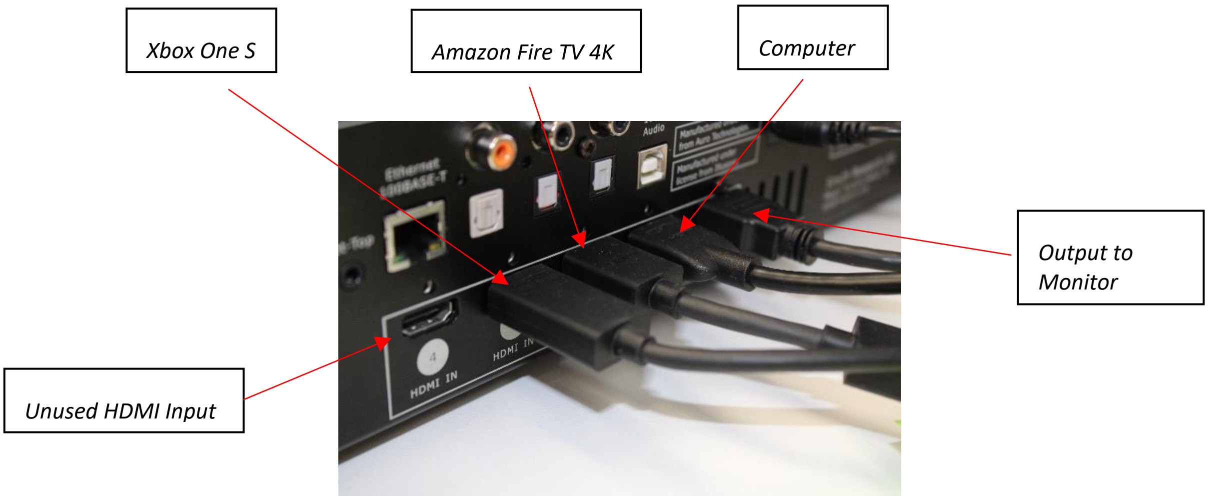
Connecting via HDMI

Several different devices can be connected by using one of the 4 HDMI input on the back panel of the Realiser. A list of Devices that are compatible with the Realiser A16 are listed below:

- *DVD Players with HDMI output*
- *Blu-Ray Players with HDMI output*
- *Apple TV*
- *Amazon Fire TV*
- *Xbox One, Xbox One S and Xbox One X*
- *PlayStation 4*

- *Other Smart TV sticks with HDMI output.*
- *HDMI Output on a computer.*

These devices can be then connected to a monitor via the HDMI output port on the rear panel. This allows the Realiser to connect 4 different devices via HDMI to a monitor or TV via a single HDMI port.



HDMI connections on the rear of the Realiser.

To select which HDMI input is output to the monitor, use the ADJ+ and ADJ- buttons on the Audio Source option on the home screen to cycle through the audio options. The Realiser will automatically switch to that HDMI input.

Connecting a computer to the USB port

To be added …

Connecting to the digital optical and digital coaxial ports

To be added …

Connecting to the analogue line-in ports

To be added …

26.1 Atmos / DTS:X AV bass management ON (HDMI, Coaxial) (HP DB disabled)

In this scenario the bass management of the decoded Atmos/DTS:X channels is enabled, and the bass managed low-frequency signal is passed to the virtual sub-woofer for SVS headphone rendering, or to a real sub-woofer for AV loudspeaker listening.

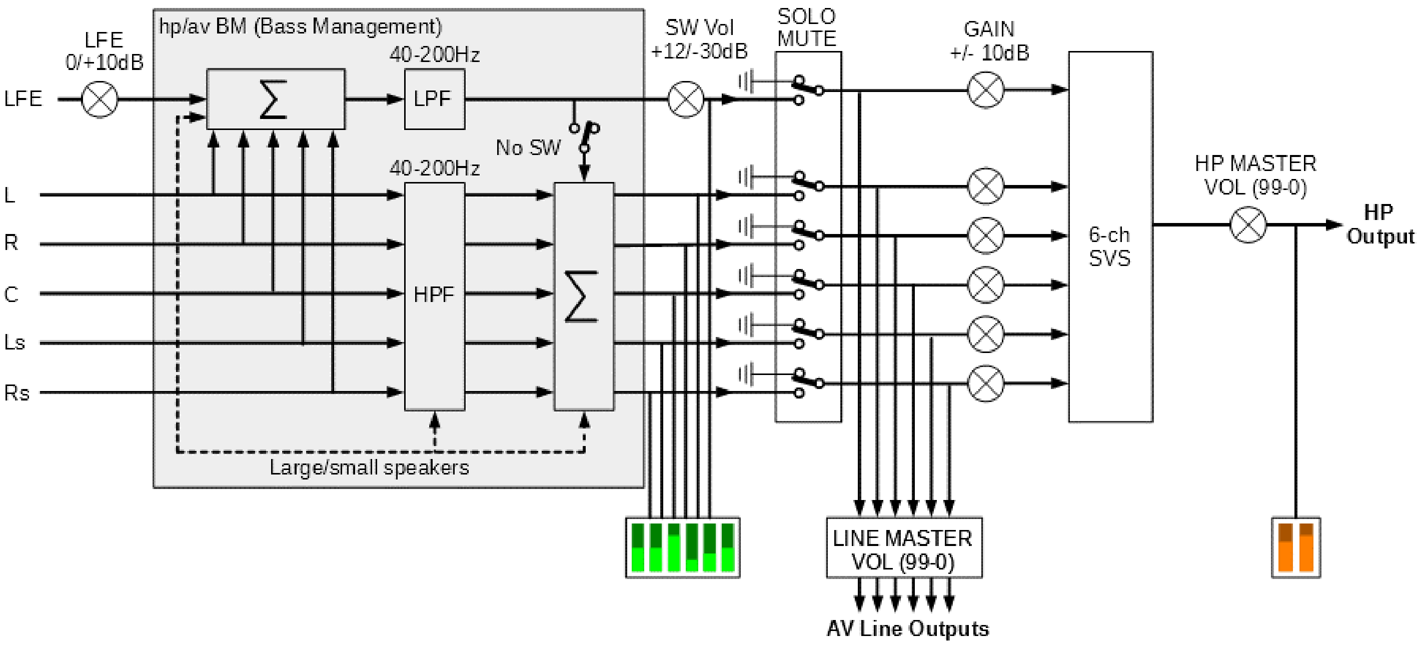
Parameters: 1: hp/av LFE +10dB (set ON or OFF)

2. hp/av SW volume (set from +12dB to -30dB)

3: hp/av BM (set ON) LPF (set from 40Hz to 200 Hz) Speaker Size Table (set to large or small)

4: hp DB (set OFF) Vol (NOT USED) LPF (NOT USED)

5: Gain (set from +10dB to -10dB for individual virtual speakers)



Bitstream bass management ON, SVS headphone Direct Bass OFF

26.2 Atmos / DTS:X AV bass management OFF (HDMI, Coaxial) (HP DB disabled)

In this scenario the bass management of the decoded Atmos/DTS:X channels is disabled, and the decoded LFE channel is passed to the virtual sub-woofer speaker for headphone rendering, or to a real sub-woofer for loudspeaker listening.

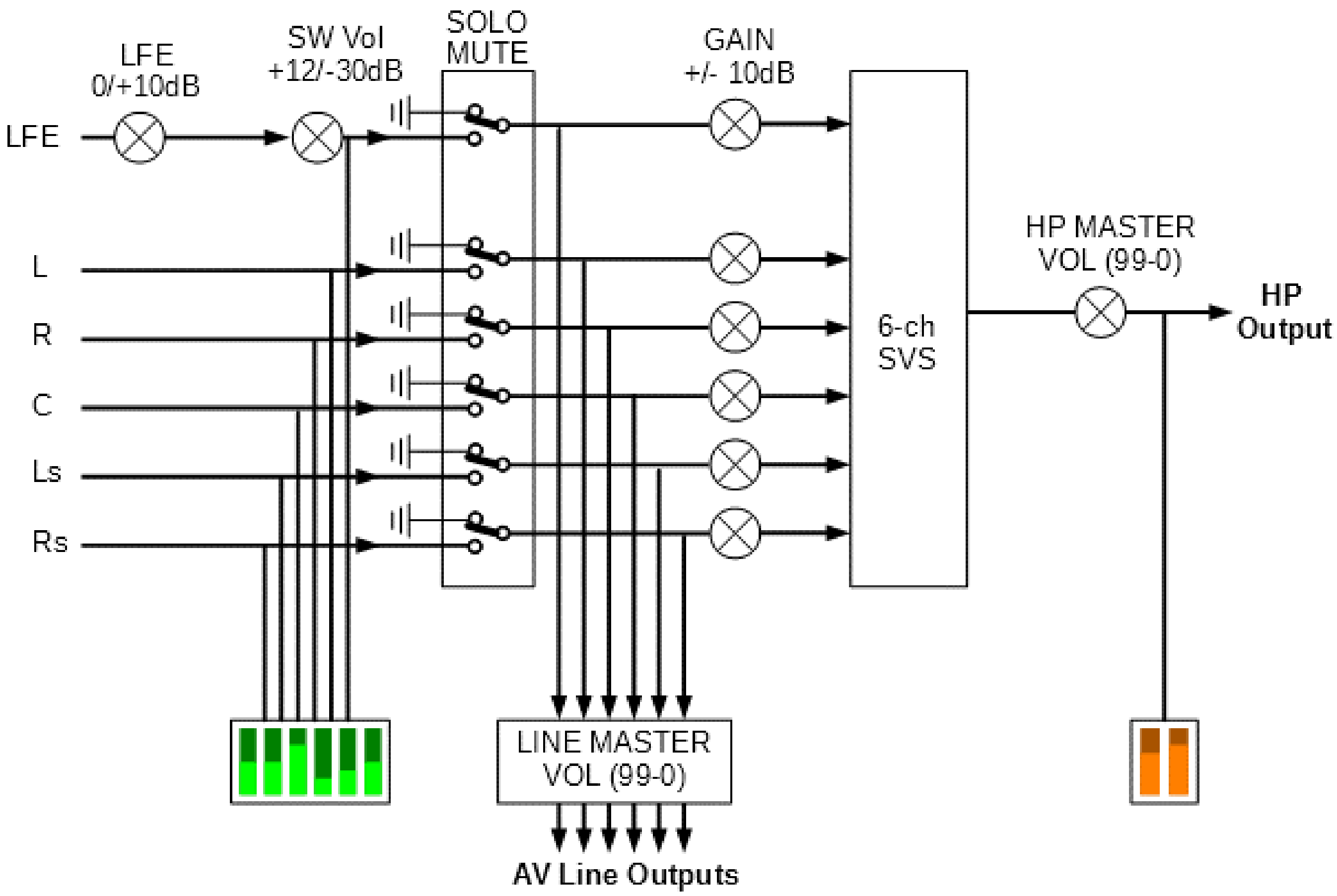
Parameters: 1: hp/av LFE +10dB (set ON or OFF)

2: hp/av SW volume (set from +12dB to -30dB)

3: hp/av BM (set OFF) LFE (not used) Speaker Size Table (not used)

4: hp DB (set OFF) Vol (not used) LPF (not used)

5: Gain (set from +10dB to -10dB for individual virtual speakers)



Bitstream bass management OFF, SVS headphone Direct Bass OFF

26.3 *Atmos / DTS:X AV bass management ON (HDMI, Coaxial) (HP DB enabled)*

In this scenario the bass management of the decoded Atmos/DTS:X channels is enabled, but the virtual sub-woofer speaker is NOT used for bass management for SVS headphone rendering. Instead the bass managed low frequency part of the signal, including the decoded LFE channel, is sent directly to the headphones. The Direct Bass mode is not relevant for AV loudspeaker listening.

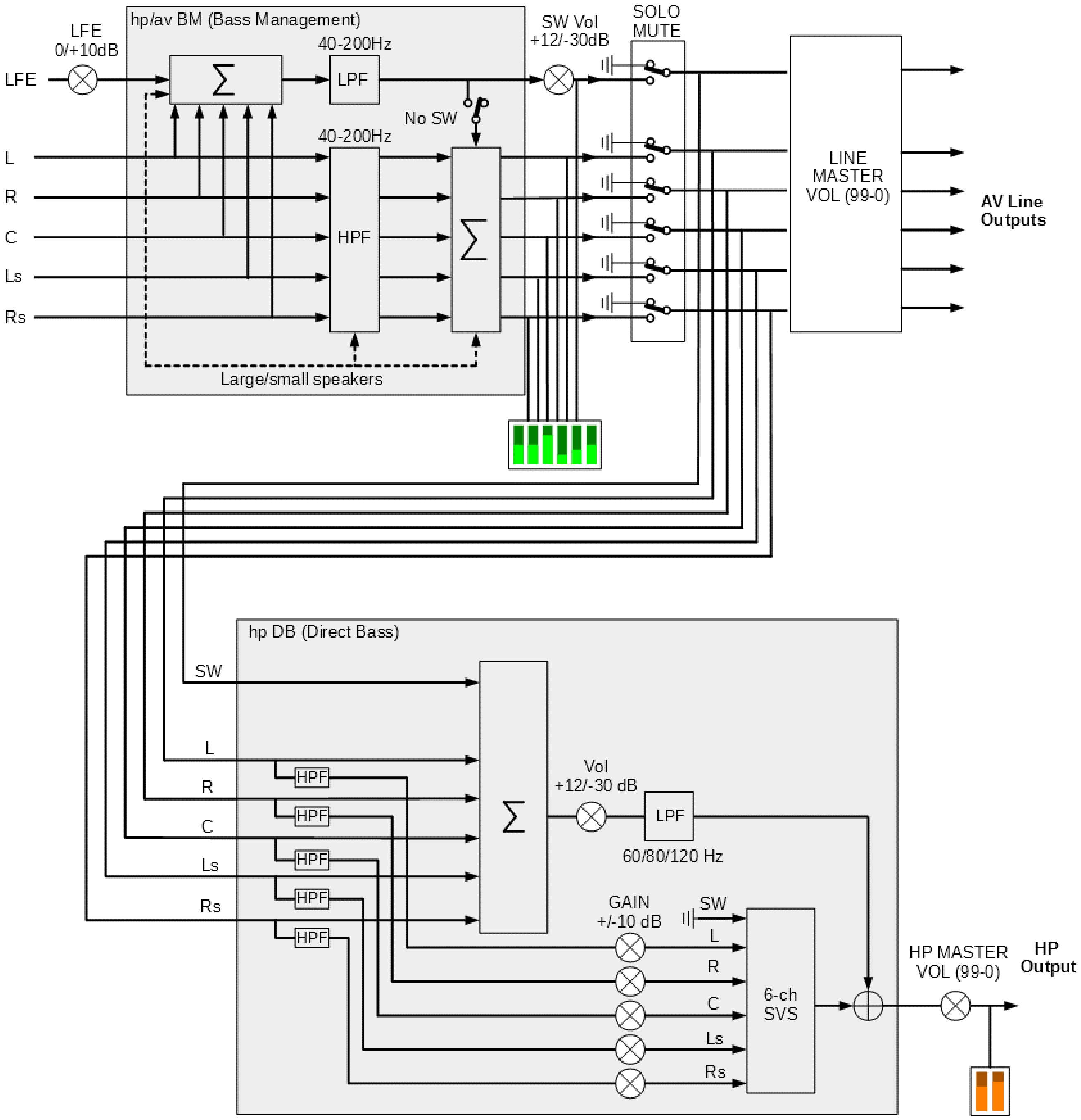
Parameters: 1: hp/av LFE +10dB (set ON or OFF)

2: hp/av SW volume (set from +12dB to -30dB)

3: hp/av BM (set ON) LPF (set from 40Hz to 200Hz) Speaker Size Table (set large or small)

4: hp DB (set ON) Vol (set from +12dB to -30dB) LPF (set to 60/80/120Hz)

5: Gain (set from +10dB to -10dB for individual virtual speakers)



Bitstream bass management ON, SVS headphone Direct Bass ON

26.4 Atmos / DTS:X AV bass management OFF (HDMI, Coaxial) (HP DB enabled)

In this scenario the bass management of the decoded Atmos/DTS:X channels is disabled, and the virtual sub-woofer speaker is NOT used for bass management. Instead the bass managed part of the signal, including the decoded LFE channel, is output directly to the headphones. The Direct Bass mode is not relevant for AV loudspeaker listening.

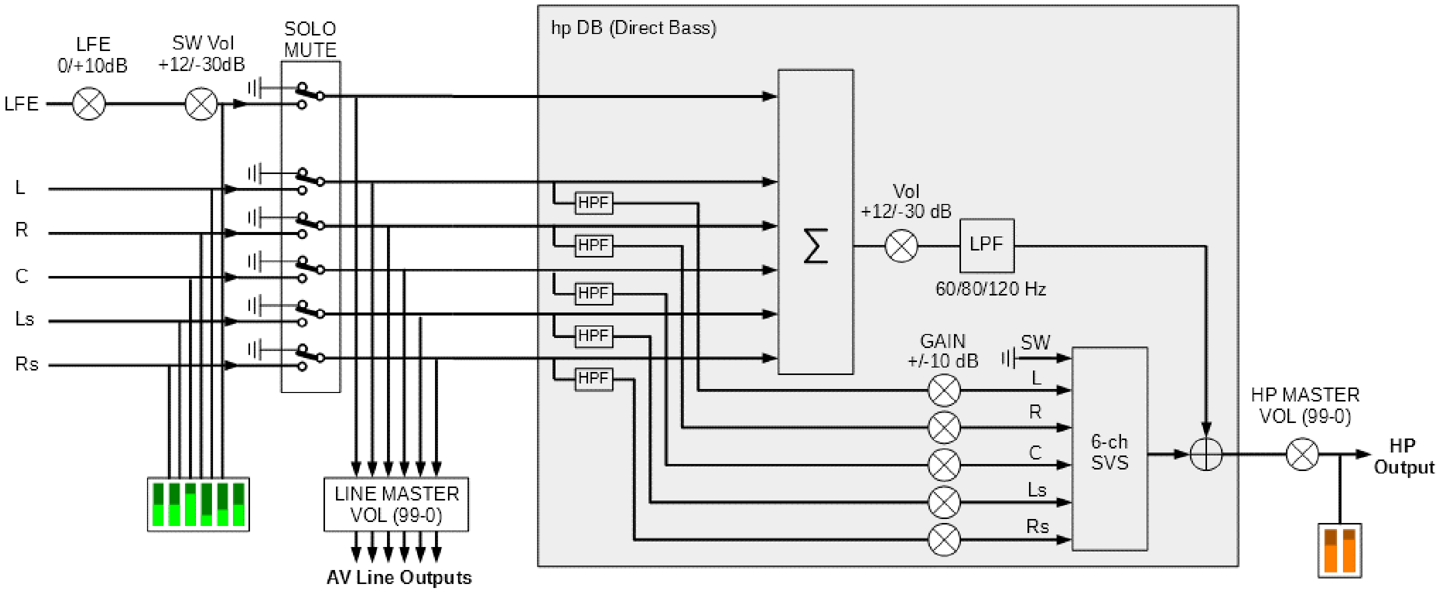
Parameters: 1: hp/av LFE +10dB (set ON or OFF)

2: hp/av SW volume (set from +12db to -30dB)

3: hp/av BM (set OFF) LPF (not used) Speaker Size Table (not used)

4: hp DB (set ON) Vol (set from +12dB to -30dB) LPF (set to 60/80/120Hz)

5: Gain (set from +10dB to -10dB for individual virtual speakers)



Bitstream bass management OFF, SVS headphone Direct Bass ON

26.5 PCM 'Direct' bass management (USB, Line)

In this scenario the virtual sub-woofer speaker is NOT used for bass management. The bass managed part of the signal is output directly to the headphones.

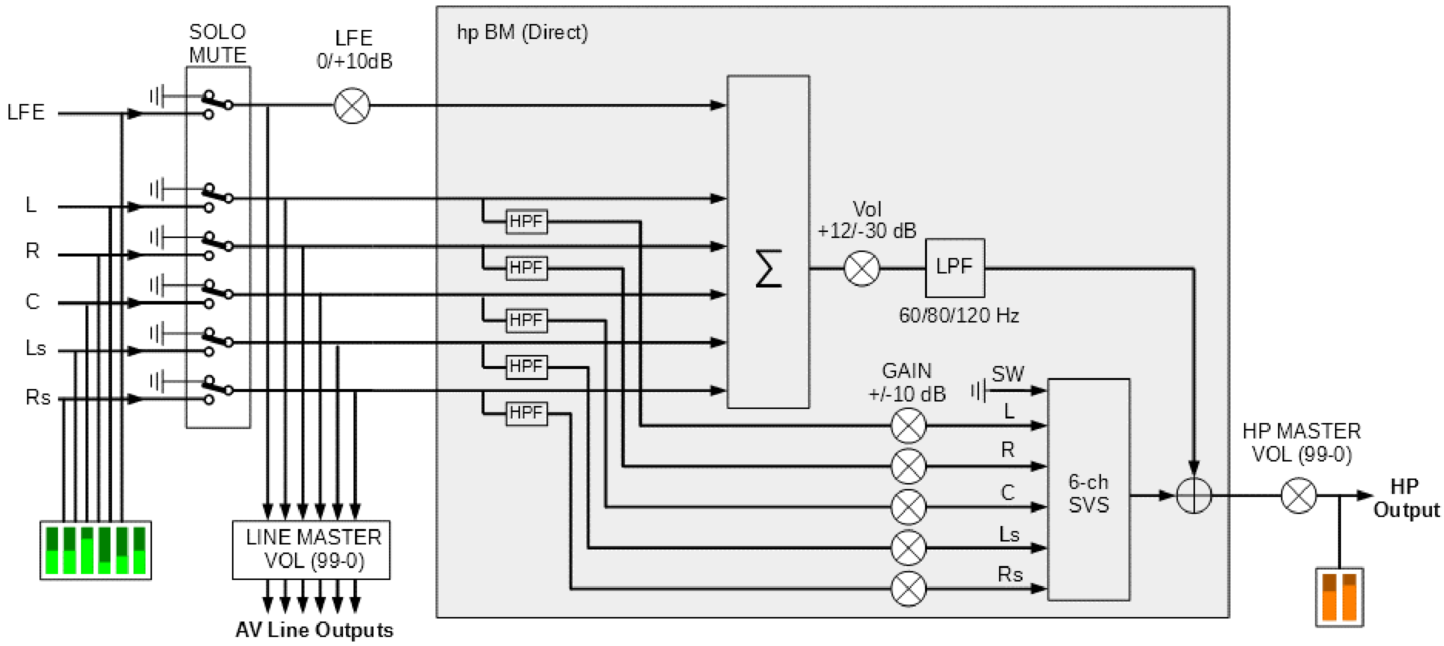
NOTE: For PCM listening rooms there is NO bass management for AV loudspeaker outputs. The AV line output signals are output immediately after the SOLO/MUTE function.

Parameters: 1: hp LFE +10dB (set ON or OFF)

2: hp SW volume (set from +12dB to -30dB)

3: hp BM (set to DIRECT) Vol (set from +12dB to -30dB) LPF (set 60/80/120Hz)

4: Gain (set from +10dB to -10dB for individual virtual speakers)



PCM bass management OFF, SVS headphone Direct Bass ON.

26.6 PCM 'Virtual' bass management (USB, Line)

In this scenario the bass managed part of the signal is passed through to the virtual sub-woofer speaker for headphone rendering. The AV line output signals are output immediately after the SOLO/MUTE function.

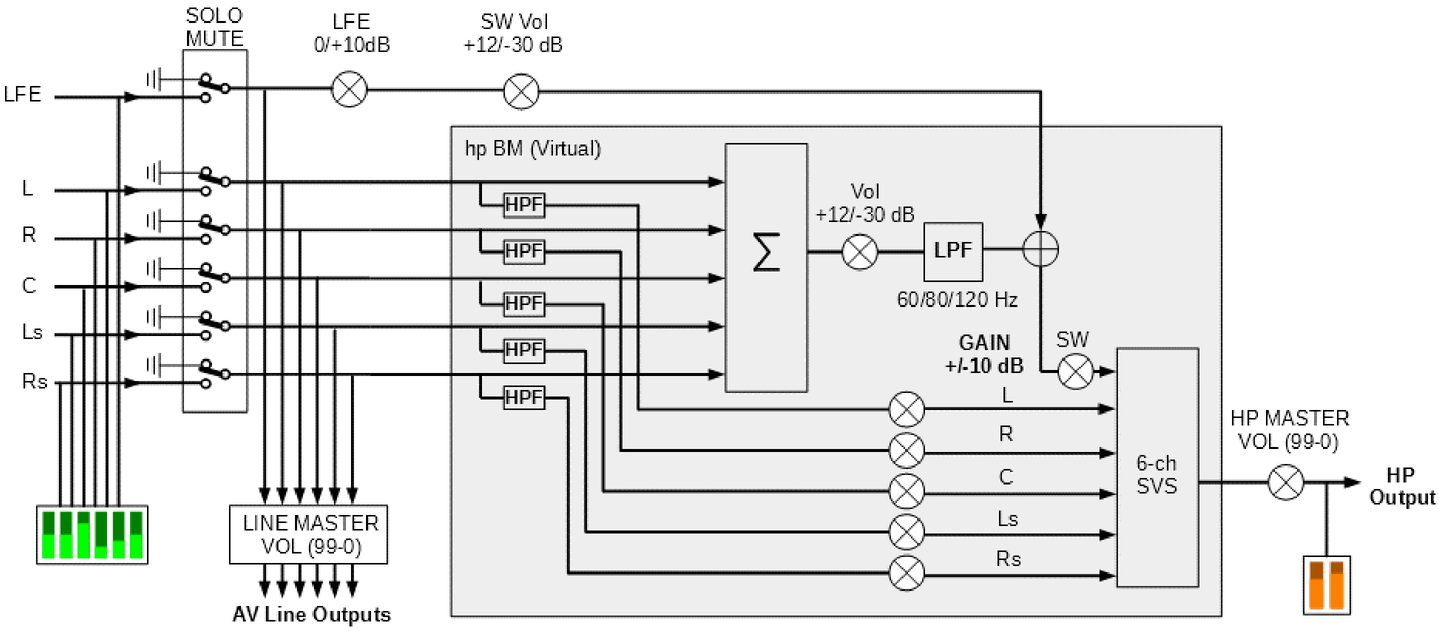
NOTE: For PCM listening rooms there is NO bass management for AV loudspeaker outputs. The AV line output signals are output immediately after the SOLO/MUTE function.

Parameters: 1: hp LFE +10dB (set ON or OFF)

2: hp SW volume (set from +12dB to -30dB)

3: hp BM (set to VIRTUAL) Vol (set from +12dB to -30dB) LPF (set 60/80/120Hz)

4: Gain (set from +10dB to -10dB for individual virtual speakers)

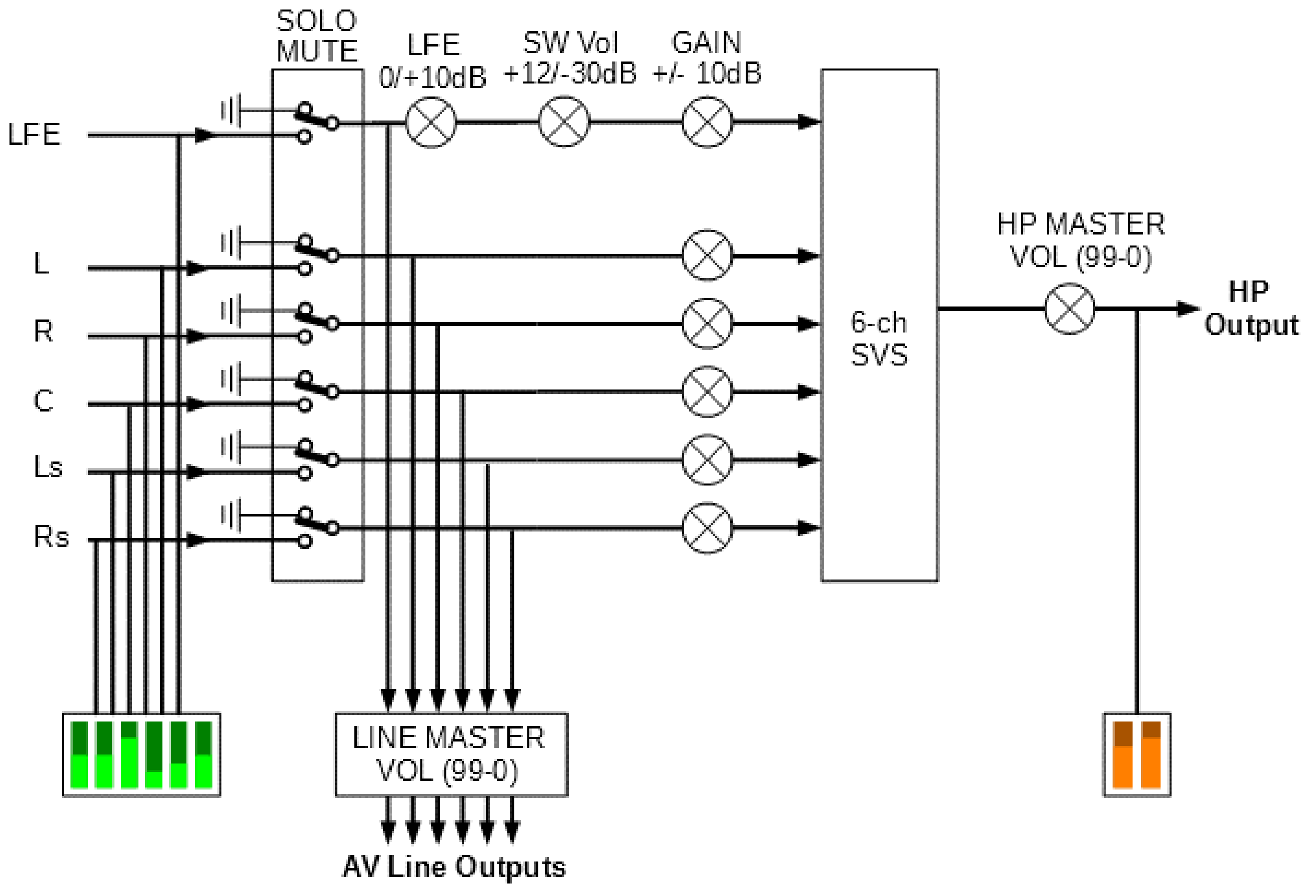


PCM bass management OFF, SVS headphone bass management ON.

26.7 *PCM bass management 'OFF'*

In this scenario bass management is disabled, and the LFE channel is passed to the virtual sub-woofer speaker for headphone rendering.

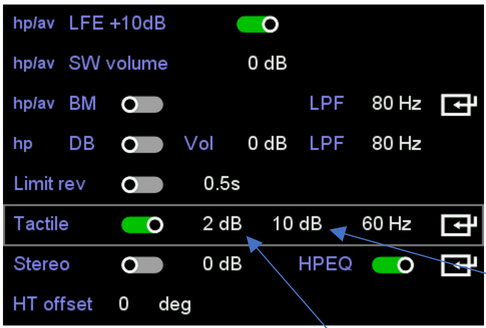
NOTE: For PCM listening rooms there is NO bass management for AV loudspeaker outputs. The AV line output signals are output immediately after the SOLO/MUTE function.



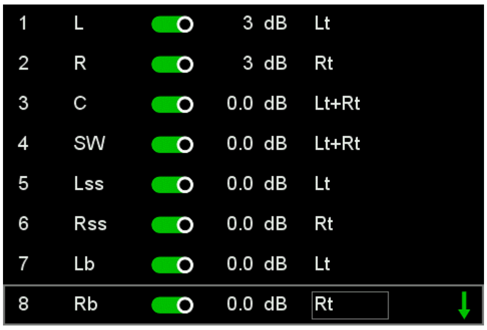
PCM bass management OFF, SVS headphone bass management OFF.

27 Appendix J: Tactile management (all inputs)

The tactile output is a two-channel, low-frequency output intended for vibration transducers to simulate body-conducted sound and acoustic vibration. The tactile signal is created by summing together one or more of the decoded or PCM signals, summing this with the rendered headphone output, and then low pass filtering the final output.



Tactile output enabled. The tactile summation block is reached through the ENTER command



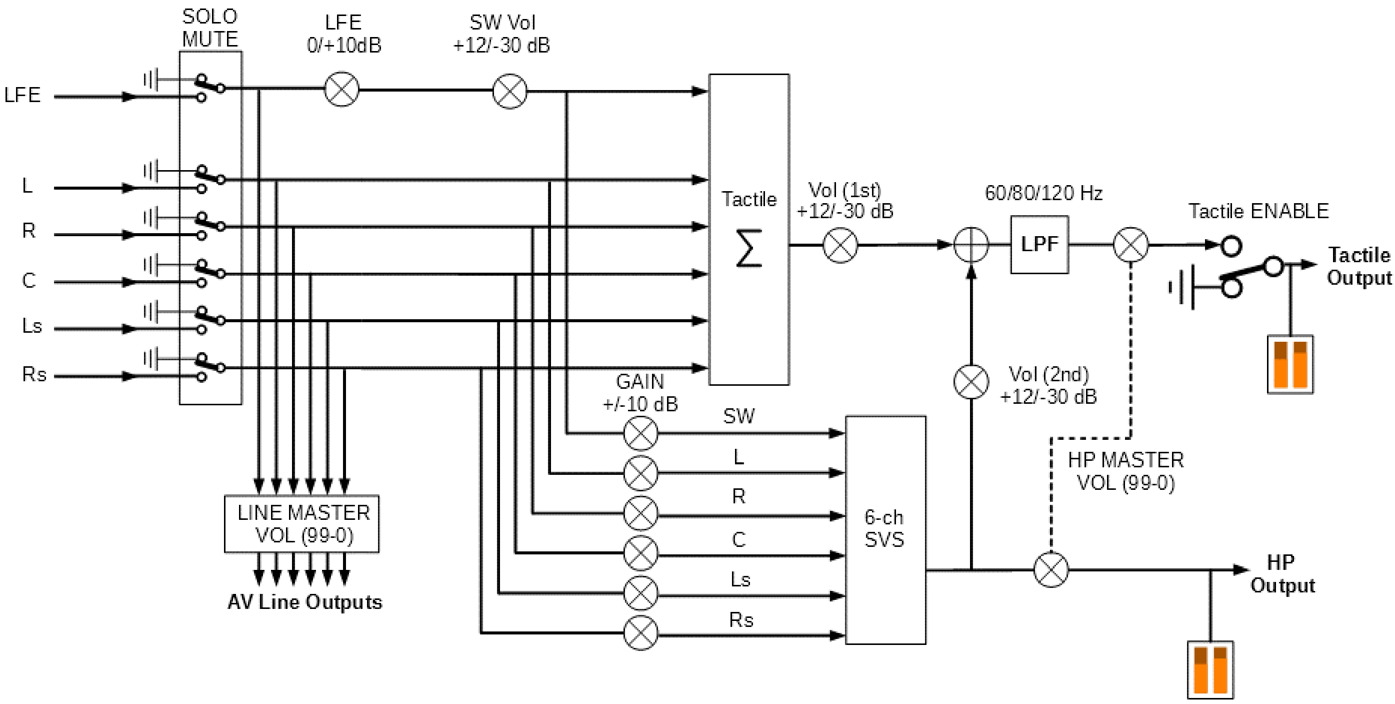
Tactile summation block. Any group of channels, with different gains, can be summed into the left, right or left+right tactile output.

For the tactile parameters listed below the first Vol is the volume set immediately following the tactile summation block, the second Vol is the volume of the SVS headphone signal summed into the tactile output, and LPF is the corner frequency of the low-pass filter operating on the final output.

Parameters: 1: Tactile (set ON) Vol (set from +12dB to -30dB) Vol (set from +12dB to -30dB) LPF (set 60/80/120Hz)
2. Tactile summation block (use ENTER command). Within the tactile mixing block each input channel can be enabled or disabled, the gain of each enabled signal can be set from +10db to -6dB, and the signal can be sent to either the left, right or left+right tactile outputs.

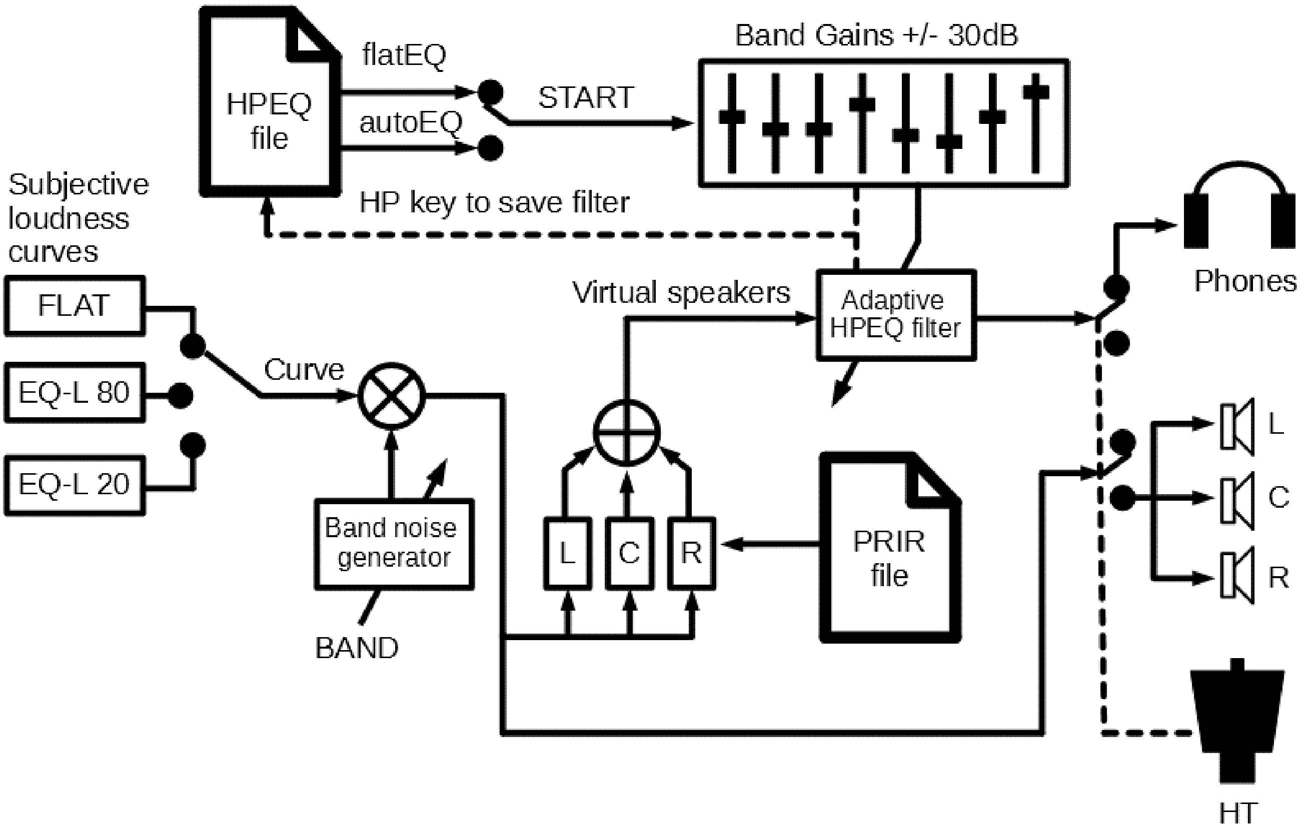
***NOTE**: When enabled, the tactile signal is output from the Tactile Out phono connectors on the back panel of the Realiser A16.*

***NOTE**: The tactile function does not work when the A16 is in test mode (TEST key). Please use regular audio to evaluate the Tactile output.*



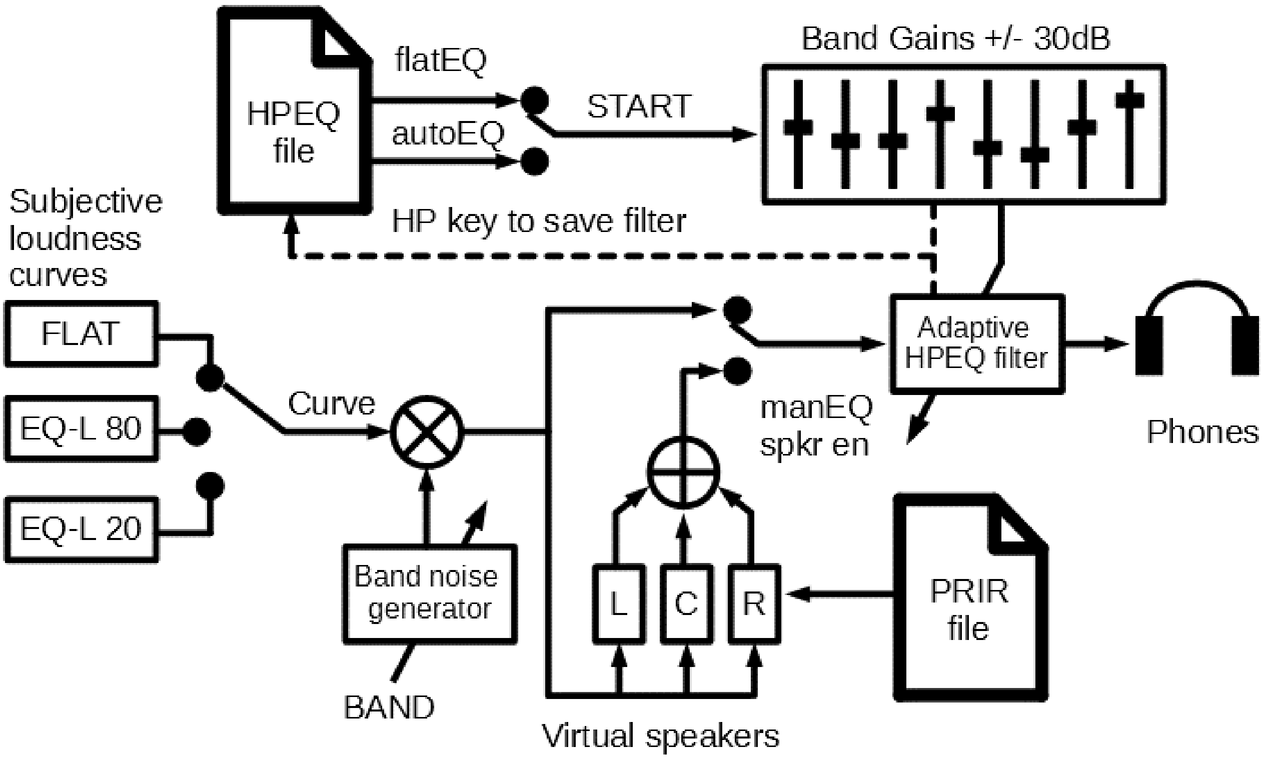
Generation of the Tactile Output signal.

28.1 Manual headphone EQ using an external loudspeaker as reference



Manual headphone EQ using an external loudspeaker as reference.

28.2 Manual headphone EQ using an equal loudness curve

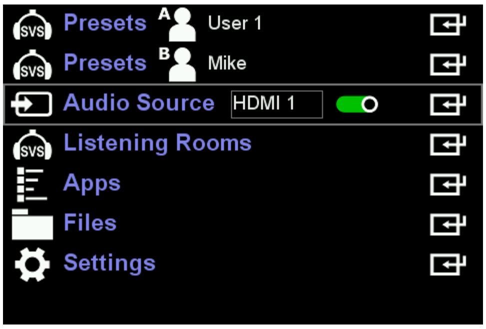


Manual headphone EQ using an equal loudness curve.

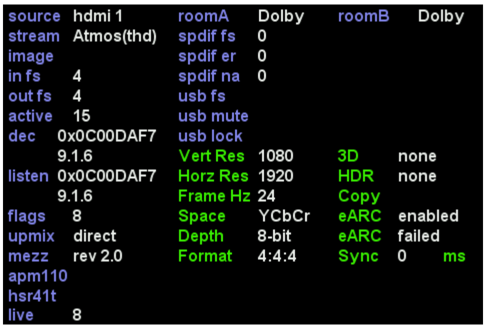
29 Appendix M: Diagnostic displays

29.1 Audio source diagnostics

The audio source diagnostics screen is activated from the Home Page menu using the ENTER command while selecting the Audio Source option. Press the BACK key to return to the Home Page menu.



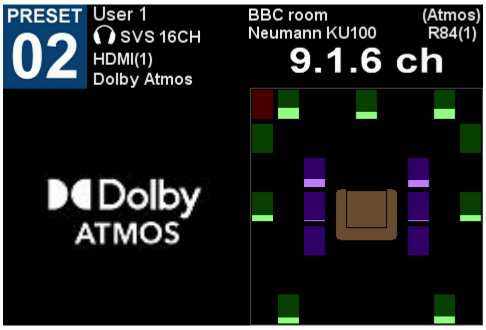
Activating the audio source diagnostics display through the Audio Source option on the Home Page menu.



Audio source diagnostics screen.

29.2 Preset speaker map information

The preset speaker map has two associated diagnostics screens, one providing information on the listening mode and the second detailing the linkage between the multichannel line outputs and the speaker labels.

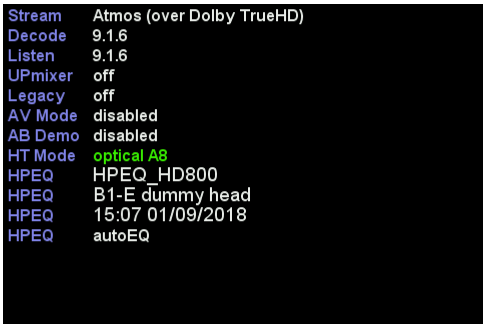


Preset speaker map display for User A

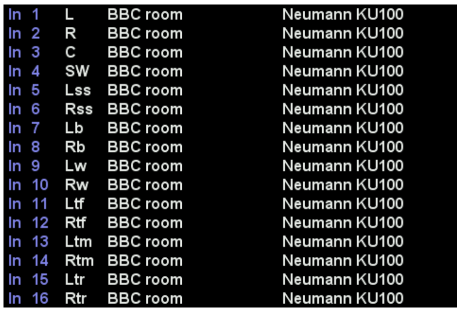
29.2.1 Listening mode and multichannel line outputs

The listening mode diagnostics screen is activated from the preset speaker map display through the UP ARROW key on the remote control. The display will automatically revert to the speaker map display after a few seconds or can be manually returned using the DOWN ARROW key. User A and User B have separate listening mode diagnostics.

The multichannel line output diagnostics screen is activated from the preset speaker map display through the DOWN ARROW icon key on the remote control. The display will automatically revert to the speaker map display after a few seconds or can be manually returned using the UP ARROW icon key.



Listening mode diagnostics for User A



Multichannel line output diagnostics.

30 Appendix N: Trademarks

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