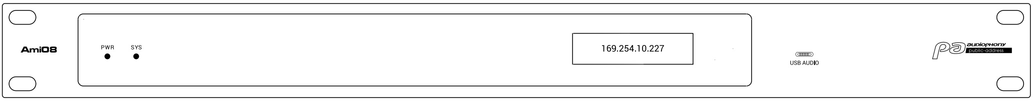
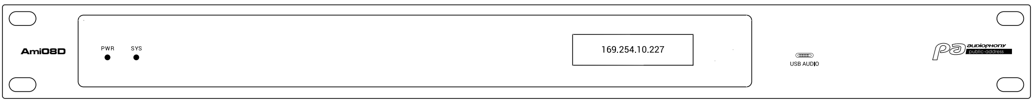



# AmiO8

**8x8 digital audio matrix – DSP – 1U**
**8x8 digital audio matrix – DSP – 1U**




# AmiO8D

**8x8 digital audio matrix + 8x8 Dante – DSP – 1U**
**8x8 digital audio matrix + 8x8 Dante – DSP – 1U**

# USER GUIDE

H11874 / H11875 – Version 1 / August 2026

# Technical and safety advice

**Please read the following technical, safety and environmental instructions carefully before installing and using your equipment.**

## Technical notes

All reasonable measures have been taken in terms of design and engineering to ensure that these matrices always operate satisfactorily within the intended application and environment, and that they provide an adequate level of support to meet all reasonable customer needs and expectations. This support is, however, subject to the following conditions.

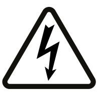
- These panels are Class I appliances and must be installed with a power cable that includes the necessary earthing to comply with Class I safety standards.
- These matrices must always be installed by competent and qualified personnel. Any damage to or malfunction of the matrix resulting from installation or operating errors may result in the withdrawal of support, the warranty or performance guarantees.
- These matrices must not be used in places where minors might have access to them.
- These matrices must only be used as part of audio systems installed and configured by professionals, comprising auxiliary input and output equipment of recognised performance standards and in good working order. Any damage to these matrices or any unsatisfactory performance on their part, resulting from unsuitable or faulty auxiliary input or output equipment, may result in the cancellation of support, the warranty or performance guarantees.
- These matrices are intended for indoor installation and use in a controlled environment (pollution degree PD2), at an ambient temperature of between 0 °C and 40 °C. They are not designed for use at altitudes exceeding 2,000 metres. Installing or using these matrices in environments that do not comply with these limits may result in the cancellation of support, the warranty or performance guarantees.
- Specific warranty terms and conditions are the responsibility of the die supplier.

## Safety and environmental notices

*Note: The symbol depicting a lightning bolt with an arrow inside a triangle is intended to alert the user to the presence of 'dangerous' uninsulated voltage inside the product's casing, the intensity of which may be sufficient to pose a risk of electric shock to people.*

*Note: The exclamation mark inside an equilateral triangle is intended to draw the user's attention to the presence, within this manual, of important instructions relating to safety, operation and maintenance.*

**WARNING! TO AVOID THE RISK OF FIRE OR ELECTRIC SHOCK, DO NOT EXPOSE THIS APPLIANCE TO RAIN OR MOISTURE.**



**Note regarding ambient temperature: if this equipment is used in a confined space or within a multi-rack installation, the internal ambient temperature may exceed the external ambient temperature. In such circumstances, it is important to ensure that the maximum operating temperature specified for the equipment is not exceeded.**

**Reduced airflow: ensure that the rack or any other enclosed structure does not obstruct the flow of cooling air required for the safe and reliable operation of the equipment.**

# Technical and safety advice

## Important safety instructions

- Please read these instructions.
- Please keep these instructions.
- Please observe all warnings.
- Follow all the instructions.
- Do not use this appliance near a source of water.
- Do not immerse the device in water or other liquids.
- Do not use any aerosols, cleaning products, disinfectants or fumigants on, near or inside the equipment.
- Clean with a dry cloth only.
- Do not block any ventilation openings. Install the appliance in accordance with the manufacturer's instructions.
- Do not install it near heat sources such as radiators, air vents, stoves or any other equipment (including amplifiers) that gives off heat.
- To reduce the risk of electric shock, the power lead must be plugged into a mains socket fitted with a safety earth.
- Do not compromise the safety function of the earthed plug. An earthed plug has two pins and a third earth pin. The third pin is provided for your safety. If the plug supplied does not fit your socket, consult an electrician to have the obsolete socket replaced.
- Ensure that the power lead is not trodden on or pinched, particularly near the plugs, sockets and where it exits the appliance.
- Do not unplug the appliance by pulling on the lead; instead, use the plug.
- Only use accessories recommended by the manufacturer.
- Unplug this appliance during a thunderstorm or if it is not going to be used for a long period of time.
- Always have any maintenance carried out by a qualified technician. Servicing is required if the appliance has suffered any damage, for example if the power cord or plug is damaged, if liquid has been spilt or objects have fallen inside the appliance, if the appliance has been exposed to rain or damp, if it is not working normally, or if it has been dropped.
- The appliance's plug, or mains plug, acts as the means of disconnecting it from the mains supply and must remain easily accessible after installation.
- Please comply with all applicable local regulations.
- If you have any doubts or questions regarding the physical installation of equipment, consult a qualified engineer.

## Environmental Statement

This product complies with international directives, including the Directive on the Restriction of the Use of Certain Hazardous Substances (RoHS) in electrical and electronic equipment, the REACH Regulation (Registration, Evaluation, Authorisation and Restriction of Chemicals) and the Directive on the Management of Waste Electrical and Electronic Equipment (WEEE). Please consult your local waste management authorities for information on how to recycle or dispose of this product correctly.

## EC Declaration of Conformity

This product complies with all the essential requirements and other specifications set out in the Directive
- 2014/53/EU (RED)
- 2014/35/EU (LVD)
- 2014/30/EU (EMC)
- 2011/65/EU (RoHS)
The full EU declaration is available at audiophony-pa.com.

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8 – Technical specifications

# 1 – Introduction

The Ami series is equipped with a number of essential features to simplify your audio installations. The DSP-based remote audio equipment is routed, processed and controlled via a computer using the AmiControl software. Signal processing is fully customisable.
AmiControl is a Windows application used to configure and control the Ami series DSP hardware. AmiControl features 16 built-in presets, and the modules and sequences within each preset can be flexibly configured to suit the installer's requirements. Once the configuration is complete, it can be saved for future use. The sequences and settings of AmiControl's built-in processing modules are suitable for most application scenarios without the need for any modification.

Ami Matrices are compatible with a wide range of protocols, such as RS232, RS485, Dante network audio control, etc. They can be paired with wall-mounted controllers from the same series (AmiCTL1, AmiCTL2, AmiD2 and AmiD4). The user interface can be customised so that the end user only has access to the functions essential to their use, all within a personalised environment (on Nexus series interfaces or on their Android or iOS phone).

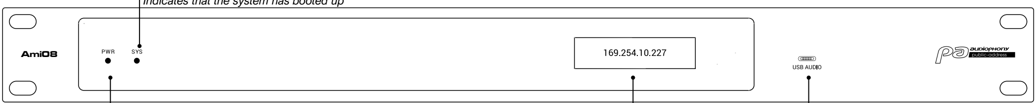
AmiControl provides access to the following functions: audio matrixing, Automix, PEQ (5–12-band) and GEQ (10–31-band) filters, delay, limiter, gate, expander, compression, ANC, AGC, group, etc.

Advanced security features ensure that end users can only access commands authorised by the installer.

# 2 - Description

## 2 - 1 - Introduction to the device

*SYS indicator light*
Indicates that the system has started up
*Indicates that the system has booted up*




*PWR indicator light*
Indicates that the matrix is powered on
*Indicates that the matrix is switched on*

Display
*Display*

USB sound card, 2 IN / 2 OUT
*2-in / 2-out USB sound card*

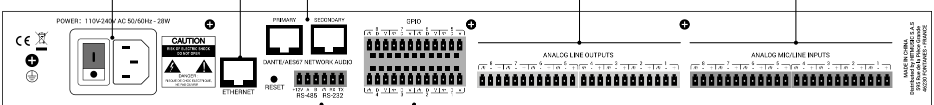
Power input
*Power supply input*

Control network
*Network monitoring*

DANTE network (on Ami08D only)
*DANTE network (Available only on Ami08D)*

Audio outputs / Balanced *audio output*: Euroblock
*Balanced: Eurobloc*

Audio inputs / Balanced *audio inputs*: Euroblock
*Balanced: Eurobloc*

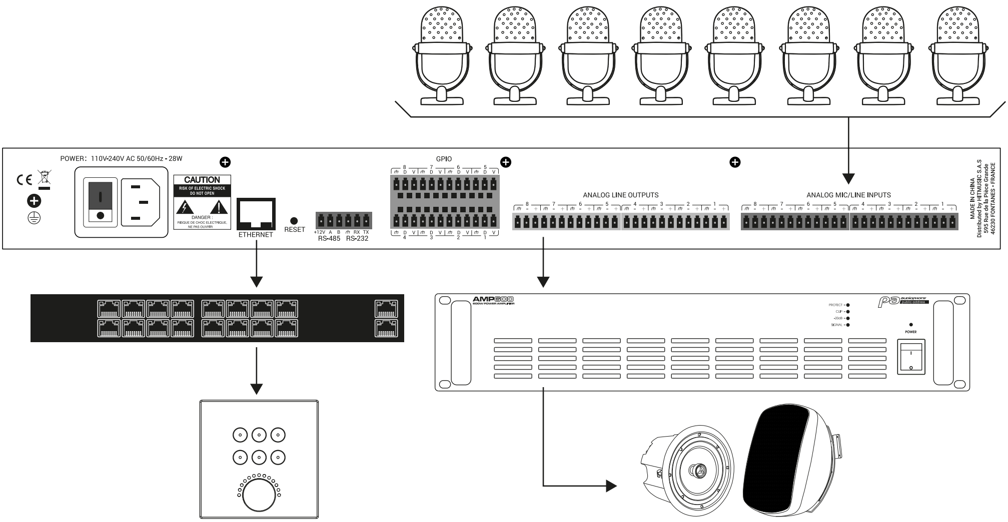



RS-485 and RS-232 interfaces
*RS-485 and RS-232 interfaces*

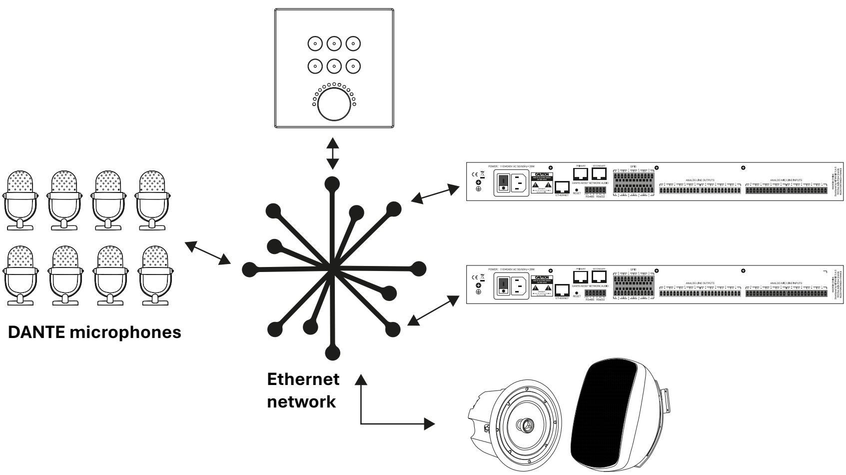
Configurable GPIOs
*Configurable GPIOs*

## 2 - 2 - Typical application of the system

**Conference amplification system**: The processor can be connected to a condenser microphone, a local output amplifier and a loudspeaker in order to enhance and process the signals via the loudspeaker on the output channel. The output signal can also be transmitted to a recording device with a Dante interface, such as a computer fitted with a virtual Dante sound card. Thanks to a simple control panel, volume adjustment, scene recall and other functions can be carried out via the UDP control processor.




**Dante Application**: The Dante network transcends spatial limitations and offers a wide range of application scenarios. It enables all devices supporting the Dante protocol to be connected to the same local network. By cascading two DSPs, it supports access to up to 32 analogue microphones and 32 Dante microphones. Each processor performs a first-level automatic mix on its own microphones, followed by a second-level automatic mix on the mixed signal via the main processor’s automatic mixer, thereby allowing more microphones to be connected to form a larger automatic mixing matrix. This two-level automatic mixing mode meets the requirements of common application scenarios.



# 3 - Installing the software

The matrix has built-in control software, which can be downloaded quickly by accessing the matrix's IP address. Enter the matrix's IP address into the browser's address bar to access the audio processor, locate the download link and download the installation software to the local computer to complete the installation.
The device's default IP address is: 169.254.10.227, subnet mask: 255.255.0.0. Please first add the address of this network segment to the PC so that the device can access it normally. Once the device has started up, enter 'http://169.254.10.227/'.

Before installing the software, please ensure that the latest version of Microsoft .NET Framework is installed on your PC.

The latest version of the AmiControl software and the user manual are available to download from audiophony-pa.com in the 'Ami Series Matrices' section.

| | |
|---|---|
| PC-end software | Download |
| Factory reset tool | Download |
| .net framework 4.0 | Download |
| identifyCodeID | 1000 |

**Minimum system requirements:**

- A Windows PC with a 1 GHz processor or faster
- Windows 7 or later
- 1 GB of available storage space
- Resolution 1024 x 768
- 24-bit colour or higher
- 2 GB or more of memory
- Network interface (Ethernet)
- CAT5 cable or existing Ethernet network.

# 4 – Using the software

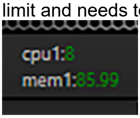
Once the software has been launched, the main menu appears as follows:


Once you have installed the AmiControl software, connect the computer running the software directly to the DSP via a network cable, or ensure that both are connected to the same network.

Open the software and click the ‘  ’ button in the top right-hand corner of the main menu to automatically detect all matrices on the network. Ensure that your computer and the matrix are connected to the same network and are using the correct IP settings. Click ‘Connect’. The user can connect to the matrix of their choice depending on their requirements; once the connection is established, the ‘Sys’ indicator light will flash. A single matrix supports simultaneous connection and control by up to four users.


1 - Allows you to add one or more matrices of the same model or of different models.
2 – A processing module that allows you to modify, replace or delete the processing modules for the input and output channels, and to display their utilisation rates in terms of both processor and memory usage.


When the resource utilisation rate reaches 100 per cent, a red colour appears, indicating that the resource has exceeded the limit and needs to be adjusted again.




3 - Download the modified programme onto the device to enable online editing and downloading.
4 – Input and output channels.
5 - List of devices: all DSP devices on the network are located within the same local area network (LAN), including those in different network segments and VLANs.
6 - Resets the scene to its default settings and restores the saved preset configuration.
7 - Resetting to factory settings. Restores the array to its factory default settings.
8 - Save the preset to store the current configuration in the corresponding preset.
9 - Loads a previously saved configuration preset into the corresponding preset in the matrix.

# 5 - Audio channel settings

There are two ways to adjust the channel settings: firstly, click directly on the input or output channel tracks to access the channel settings interface; secondly, right-click on the channel to bring up the configuration interface. The first method is used for the following channel settings:

## 5 - 1 - Input channels


**Sensitivity**: Microphone gain, 17 levels to choose from (0/3/6/9/12/15/18/21/24/27/30/33/36/39/42/45/48 dB).

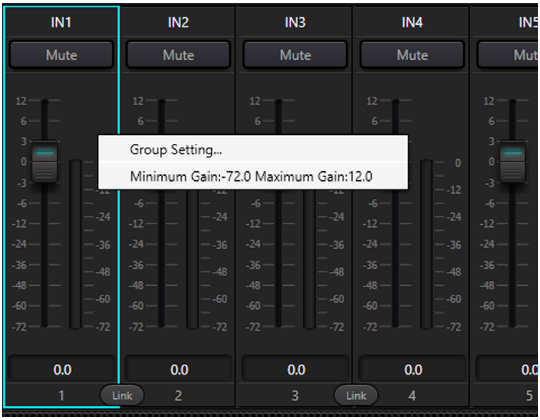
**Phantom**: Phantom power: Supplies power to external condenser microphones; click the button if necessary. Ensure that you do not connect any condenser microphones whilst phantom power is active. Microphones must always be connected whilst the matrix is switched off.

**Sine**: Sine wave: drag the frequency slider (Freq (Hz)) to generate a sine wave at the desired frequency (20–20 kHz). You can adjust the output level (Level (dBFS)) to suit your requirements. Use the fader to adjust it, or click the text field to enter a value.

**White**: White noise: Observe it on the constant-bandwidth frequency spectrograph, which displays a flat frequency spectrum. At this stage, frequency tuning will fail, and the level can be used. Each frequency component of the white noise has an equivalent level.

**Pink**: Pink noise: The power of the frequency components of pink noise is mainly distributed across the mid and low frequency bands. It decreases at a rate of 3 dB per octave across the mid and low frequency bands. In this case, the frequency control does not work and only the level control can be used.

You can also access the following menu by right-clicking on each fader in the main menu.

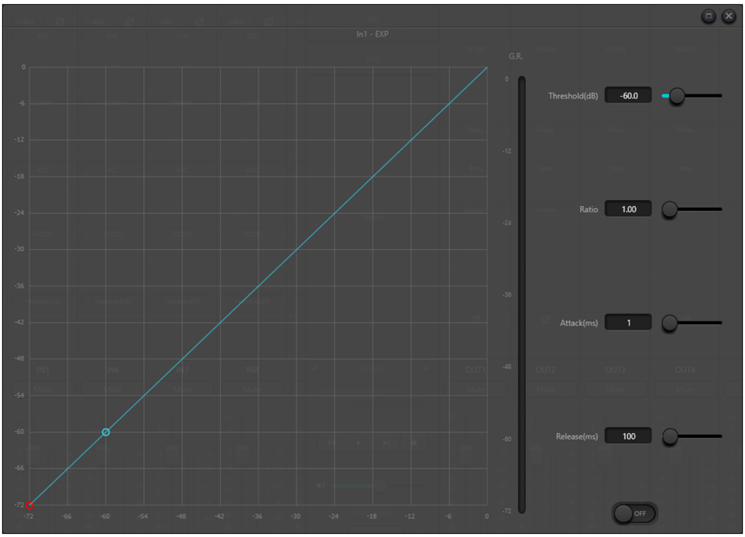



**Group settings**: Opens the group settings interface.

**Minimum Gain / Maximum Gain**: Limit the minimum and maximum gain values for a channel. Once the system is up and running, if you do not want its stability to be affected by external factors, you can set a maximum gain.

## 5 - 2 - Expander

The expander is used to widen the dynamic range of a signal. The expander acts on the parts of the signal below the threshold. The expander can transform a small signal into an even weaker one. Figure 3.2 shows that, when the expansion ratio reaches 1:2, an input signal 20 dB below the threshold produces an output signal 40 dB below the threshold. Thus, as illustrated below, the signal below the threshold is extended downwards and produces a lower level. When an expansion ratio of 1:20 is used, the expander behaves similarly to a noise gate in terms of its transmission characteristics. In fact, a noise gate is an expander with a high expansion ratio.




**Threshold (dB)**: Threshold: The expander can only be activated when the signal exceeds this threshold (allowing the signal to pass through). In practice, the threshold is often set to the level of ambient noise.

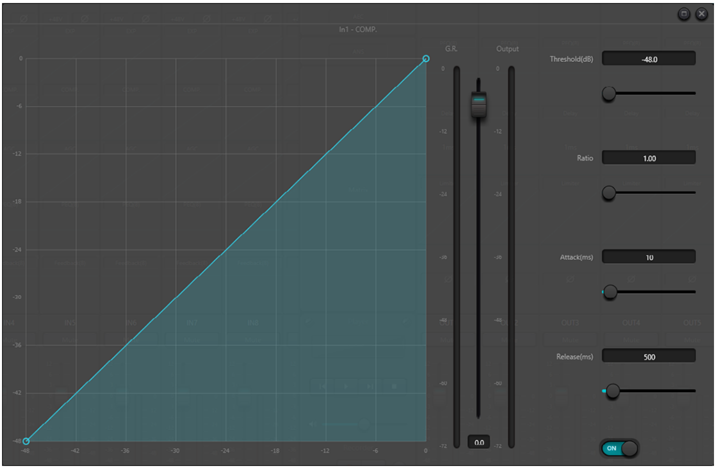
**Ratio**: This refers to the slope below the threshold point on the gain curve. When the slope is set to a high level, the door begins to move.
**Attack (ms)**: Start-up time: This is the time taken to start up the expander when the duration of the input signal exceeds the threshold. A shorter start-up time allows the expander to start up more quickly.
**Release (ms)**: Release time: this is the time taken for the gain to return to a value below the threshold when the input signal is below the threshold.

Whether it is the trigger time or the release time, this simply helps to reduce the rate at which the gain attenuation changes. In other words, the rate at which the gain changes from -40 dB to 0 dB is slowed down by the trigger time. The rise time or release time is not linked to the threshold. If the signal falls below the threshold, the rise time and release time will each have their respective influence on the gain reduction; when the signal level exceeds the threshold, the gain reduction produced by the expander will fade out at the rate controlled by the rise time. When the gain reduction is reduced to 0 dB, the expander stops expanding. Subsequently, when the signal drops back below the threshold, the expander restarts and the release time begins to take effect.

## 5 - 3 - Compressor and limiter

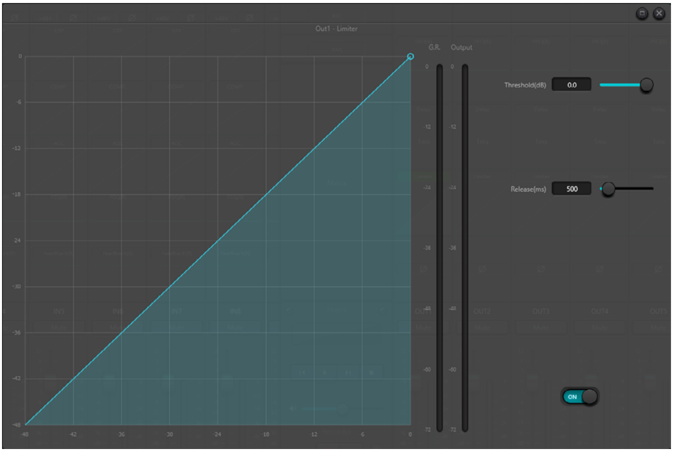
### 5 - 3 - 1 - Compressor

The compressor is used to reduce the dynamic range of the signal above the threshold set by the user and to maintain the dynamic range of the signal below that threshold. The compressor has the following control parameters:




**Threshold (dB)**: Threshold: When the signal level exceeds the threshold, the compressor/limiter begins to reduce the gain. Any signal exceeding the threshold is treated as an overdrive signal, and its level will be reduced. The more the signal exceeds the threshold, the more the level is attenuated.
**Ratio**: This refers to the compression ratio. The ratio determines the degree to which the signal exceeding the threshold level is attenuated. The lower the compression ratio, the more the signal will tend to exceed the threshold. Once the signal exceeds the threshold, the compression ratio determines the ratio between the variation in the input signal and that in the output signal. For example, when the compression ratio is 1:2, if the input signal is 2 dB above the threshold, the excess portion varies by only 1 dB. A compression ratio of 1:1 means that the compressor does not attenuate the signal proportionally. The adjustment range for the compression ratio is between 1 and 20.
**Attack (ms) / Release (ms)**: In order to preserve natural oscillation, it is generally desirable for part of the original signal level to pass through the compression without any (or with minimal) effect. Similarly, if the signal's gain undergoes a sudden and rapid attenuation followed by a rapid recovery, a 'sucking' effect occurs. The compressor's attack and release times are used to prevent this from happening. The attack time determines the rate at which the gain is attenuated, whilst the release time determines the rate at which the gain recovers.
**Attack (ms) / Release (ms)**: Also known as the gain compensation fader. If the compressor reduces the signal level significantly, it may be necessary to increase the output gain to maintain the volume. This increase applies to all parts of the signal and is not linked to the other compressor parameter settings.
**G.R. and output level indicator** : G.R. indicates the compressor's compression level; 'Output' refers to the output level of the signal that has passed through the compression module. The compression level is displayed on an inverted level meter. If the input signal and threshold are set to -6 dB and -30 dB respectively, and the ratio is 2:1, the compression level is 12 dB; the G.R. level meter displays approximately -12 dB and the output level approximately -18 dB.

## 5 - 3 - 2 - Limiter

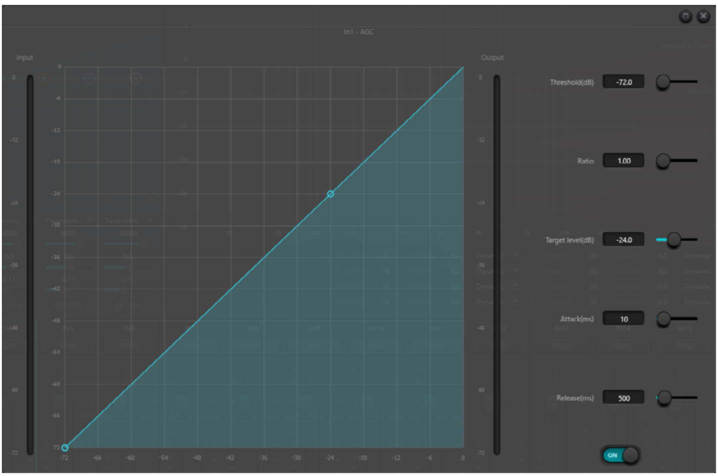



The limiter has just one essential task: to ensure that the signal never exceeds the threshold level. By adjusting the compressor's control parameters, its operating modes will be very similar to those of the limiter. The fundamental operating principle of a limiter is that it focuses specifically on the signal below the threshold level, as well as on how gain reduction is applied before an overshoot signal occurs. The limiting phase comprises two stages of processing: during the first stage, there is minor limiting, but any signal exceeding the threshold will not be processed; during the second stage, if a signal does exceed the threshold, it will be attenuated very abruptly.
The limiter offers only two parameters: the threshold and the release time. In terms of signal processing, the limiter will deal with occasional clipping, whilst the signal level will need to be attenuated in the event of frequent clipping.

## 5 - 4 - AGC

Automatic gain control (AGC) is a type of compressor. Its threshold is set at a very low level with a medium to slow rise time, a long release time and a low ratio. The aim is to bring a signal of uncertain level up to a target level, whilst preserving the dynamic range. Most automatic gain control systems incorporate silence detection to prevent the loss of gain reduction during periods of silence. This is the only feature that distinguishes automatic gain control from a standard compressor/limiter.

Automatic gain control can be used to normalise the output level of CD players playing background music, entrance music and on-hold music, in order to eliminate level fluctuations in certain PA microphones.

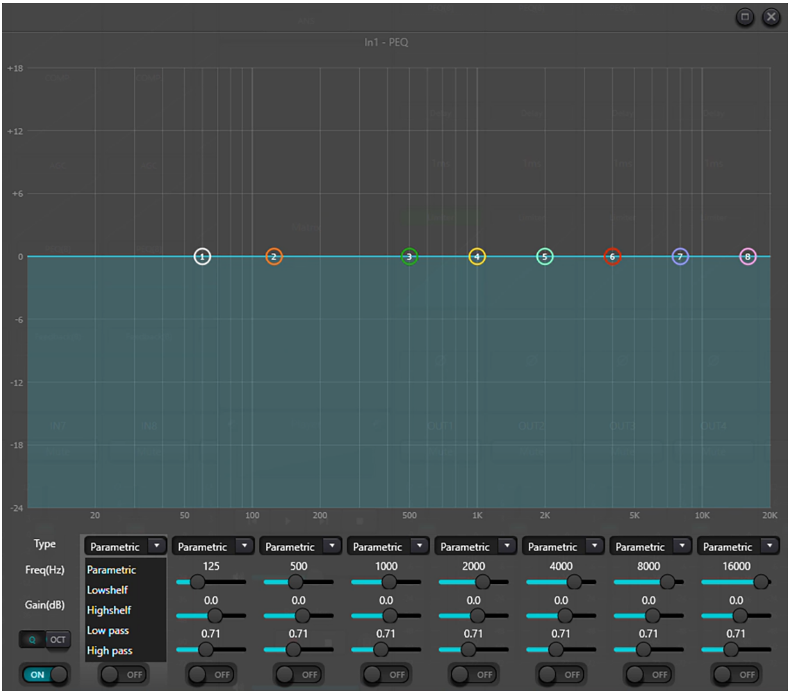


Automatic gain control comprises the following control parameters and switches:

**Threshold (dB)**: Threshold: When the signal level is below the threshold, the input-to-output ratio is 1:1. When the signal level is above the threshold, the input-to-output ratio varies according to the ratio control settings. The threshold is defined as the background noise just above the input signal level.
**Ratio**: This is the ratio of the changes in the level of the input signal above the threshold to the changes in the level of the output signal.
**Target level (dB)**: This is the ratio of the changes in the level of the input signal above the threshold to the changes in the level of the output signal.
**Attack (ms) / Release (ms)**: In order to preserve natural oscillation, it is generally desirable for part of the original signal level to pass through the compression without any (or with only minimal) effect. Similarly, if the signal's gain undergoes a sudden and rapid attenuation followed by a rapid recovery, a 'sucking' effect occurs. The compressor's attack and release times are used to prevent this from happening. The attack time determines the rate at which the gain is attenuated, whilst the release time determines the rate at which the gain recovers.
**Attack (ms)** : Rise time: this is the response time required to bring the level above the threshold.
**Release** : Release time: This is the response time required to regulate the level below the threshold.

## 5 - 5 - PEQ

The equaliser is primarily used to correct frequency ranges that are either too pronounced or lacking, whether broad or narrow. Furthermore, the equaliser can also help us to narrow or broaden the frequency range, or to alter the amount of a particular component within the frequency spectrum. Put simply, the equaliser can be used to alter the tone of the signal.
The equaliser has the following control settings:




Automatic gain control comprises the following control parameters and switches:

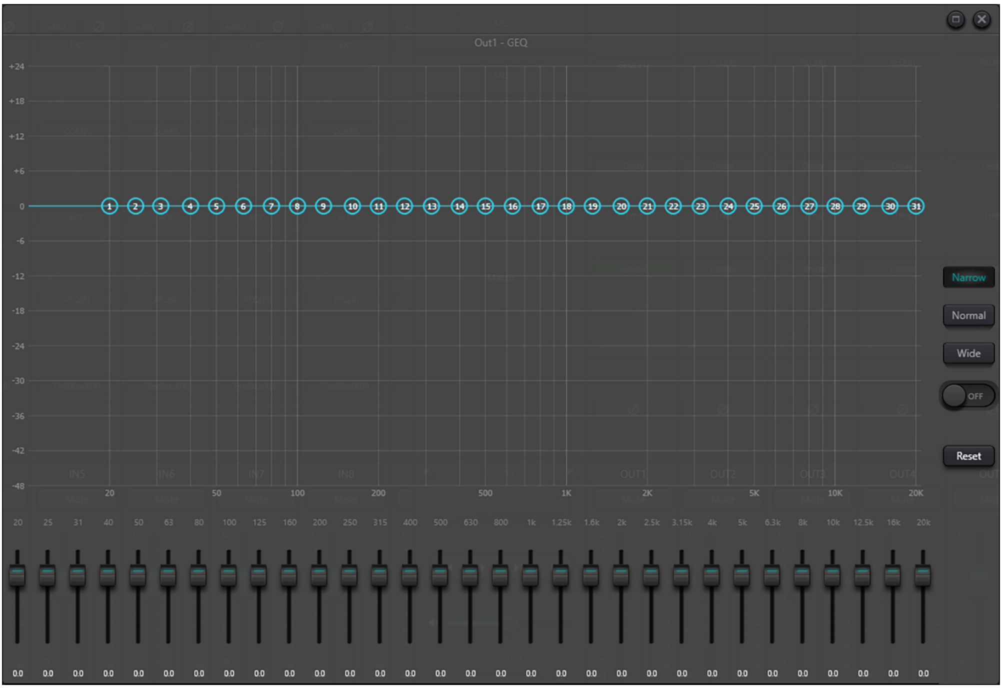
**Type**: The parametric equaliser is selected by default. You can choose between high- and low-shelf filters, and high-pass and low-pass filters. Each filter type offers different shapes to fulfil a variety of functions.

**High-pass and low-pass filters**: The reference frequency of a high-pass or low-pass filter is known as the cut-off frequency. A high-pass or low-pass filter allows frequencies on one side of the cut-off frequency to pass through the filter without

attenuation; at the same time, frequencies on the other side of the cut-off frequency are gradually attenuated. Of these, a high-pass filter allows frequencies above the cut-off frequency to pass through and attenuates frequencies below it. Conversely, a low-pass filter allows frequencies below the cut-off frequency to pass through and attenuates frequencies above it.
**'High & low shelf' filter**: This is also known as a 'shelf' filter. A 'high shelf' filter means that the gain is boosted or attenuated for frequencies above the set frequency. A 'low shelf' filter means that the gain is boosted or attenuated for frequencies below the set frequency. The set frequency is not the 3 dB cut-off frequency, but refers to the centre of the filter's downward or upward slope. The Q factor affects the peak and has a mathematical relationship with it.
**Freq (Hz)**: Frequency: This refers to the centre frequency of the filter.
**Gain (dB)** : This is the value, in decibels, of the amplification or attenuation at the centre frequency.
**Q** : This refers to a filter's quality factor. The Q-factor can be set between 0.02 and 50;
When the filter is a parametric equaliser, the Q-factor refers to the width of the bell-shaped frequency response curve on either side of the cut-off frequency.
When the filter is a 'high & low shelf' filter or a high-pass and low-pass filter, if Q > 0.707, peaks appear in the filter responses. If Q < 0.707, the slope flattens out and attenuation occurs earlier.
Each segment of the equaliser has a switch that allows you to turn the corresponding segment on or off.
Once disabled, the settings will no longer function. The equaliser has a main switch that allows you to enable or disable a module.
Parametric equalisers have 5, 8 or 12 bands.

## 5 - 6 - GEQ

Thanks to constant Q technology, each frequency point is fitted with a push-pull potentiometer. The filter's bandwidth remains unchanged, regardless of which frequency is boosted or cut. The classic professional graphic equaliser divides signals from 20 Hz to 20 kHz into 10, 15, 27 or 31 segments for adjustment.
The graphic equaliser currently offers 10, 15 and 31 bands.

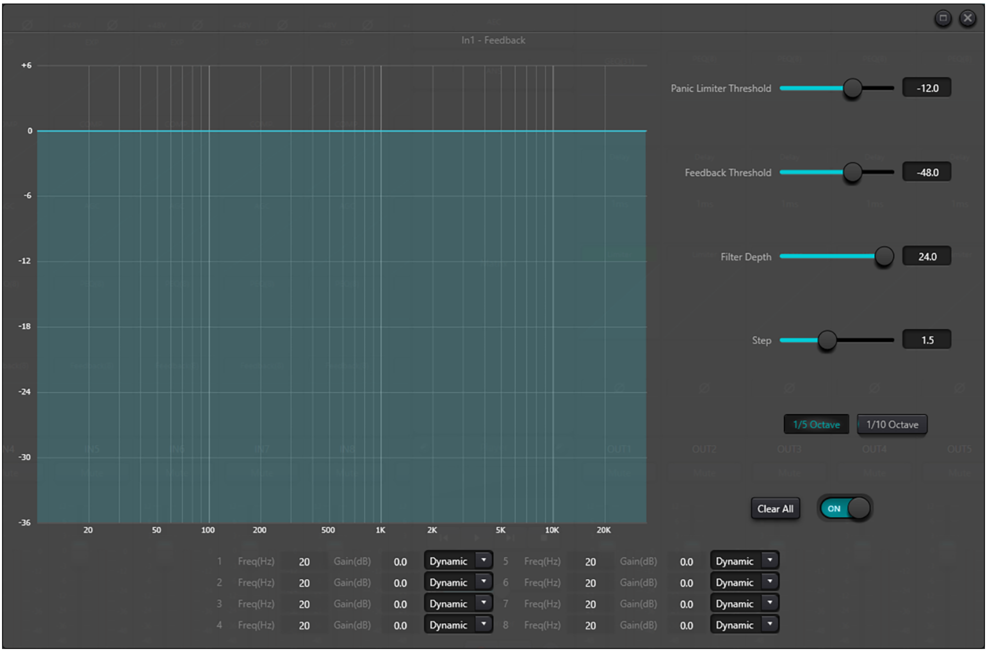


## 5 - 7 - Reducing feedback

When using the feedback suppression module, it is best to combine it with sound systematic designs and practical solutions, without, however, replacing them. Traditional methods should always be employed, such as limiting the number of active microphones, reducing the distance between the sound source and the microphone, positioning the microphone and loudspeaker to minimise feedback, and equalising the room to achieve a flat frequency response. Subsequently, a feedback suppressor may be used to provide additional gain. A feedback suppressor cannot be used to magically resolve system design flaws or improve sound transmission gain in a way that exceeds the system's physical limits.

The feedback suppression module automatically detects and suppresses feedback in the audio system. The module distinguishes feedback from expected sounds based on the characteristics of the signals. When feedback at a particular frequency is detected, a band-stop filter is automatically applied at the point of feedback to attenuate it. On first application, the band-stop filter attenuates the feedback only slightly. If the feedback persists, the band-stop filter will continue to attenuate it in accordance with the predefined settings until the feedback loop disappears or reaches the predefined maximum value.
A number of user parameters can be used to fine-tune the module's effects with precision.

Once the feedback has been eliminated, the filter can be locked to prevent any changes being made whilst the performance is in progress.
The module's settings can be copied to a dedicated band-stop filter module (such as an equaliser). Eight filters are configured as automatic filters in an automatic cycle. In this way, filters intended for temporary use can be removed.
Each channel has a feedback suppression function. Use the mouse to drag the input module and locate the feedback suppression module, or access this module quickly by clicking the shortcut on the right.
If the feedback suppressor needs to be activated, click the activation button to automatically detect the feedback point, then use a narrow-band filter to eliminate it. Each feedback suppression module has 8 narrow-band filters.



The feedback suppression module has the following adjustable settings:

**Panic limiter threshold**: According to this setting, 'any level above the threshold is considered feedback'. When a signal level exceeds the feedback threshold, one of the following occurs:
- (a) the output gain is temporarily reduced to control the rate of feedback.
- (b) the output level is limited to prevent loss of control.
- (c) the filter's sensitivity is increased to enable faster detection and response.

Once the output level falls below the threshold, the gain is restored and the sensitivity returns to normal. This value corresponds to the peak value of the signal within the digital range. If the value is set to 0, this function is disabled.

**Feedback threshold**: According to this setting, 'any level below the threshold is not considered feedback at all'. This may prevent the module from detecting feedback in soft music or caused by low-level noise.

**'Filter depth** ': This refers to the maximum attenuation of a single filter. A low setting can prevent the filter or band-stop filter from causing excessive damage to the signal. This may result in less effective feedback control, particularly in a system with a narrow resonance.

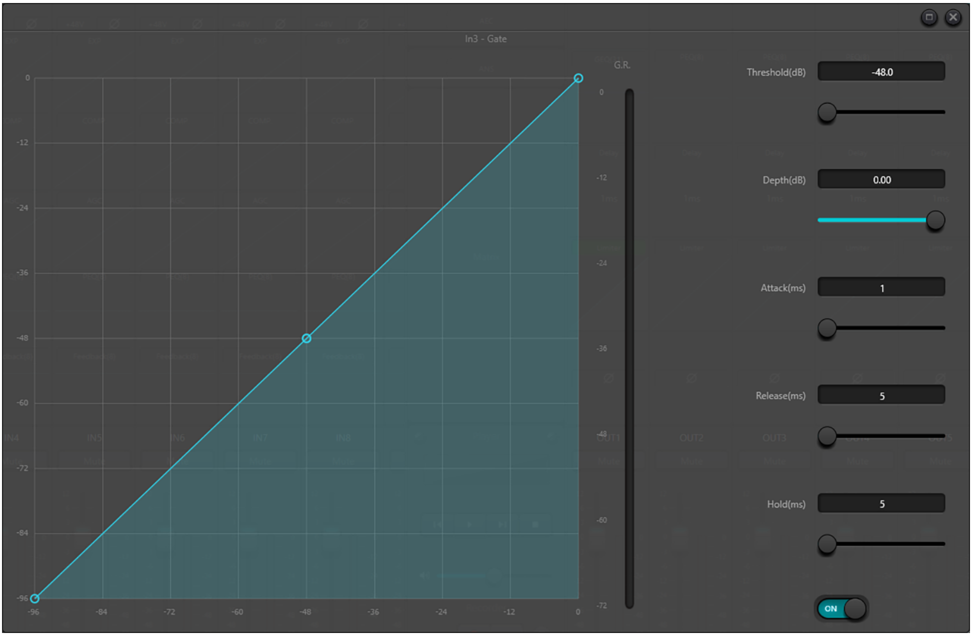
**Bandwidth**: 1/10 and 1/5 of an octave can be selected. A constant Q-factor is used. The filter will not widen as the depth increases. It is recommended to use the filter in a speech environment. In the event of frequent feedback, the bandwidth is set to 1/5 octave as this offers a wider bandwidth and greater influence.

**Preset** : There are four built-in presets: 'large music hall', 'small music hall', 'large vocal hall' and 'small vocal hall'. These four presets correspond to the default settings used in most applications.

**Auto mode for the band-stop filter** : Auto mode is enabled for the band-stop filter. Once all eight filters have been used, a new feedback loop is detected and the module checks the 'auto' filter and uses it to suppress this new feedback loop. Each band-stop filter has three modes: auto, manual and fixed. When manual mode is selected for the filter, the gain can also be adjusted manually. When fixed mode is selected, the filter operates continuously and will not be occupied by new feedback points; it continues to operate even after a restart. If you wish to save these feedback settings, please click the preset save button.

**Clear All** : Click this button to clear all filters instantly. This will clear all previously detected feedback points. This operation is usually carried out when the feedback module is put back into service.
The feedback suppressor can be used as a tool during system commissioning to identify feedback points, or as a preventative measure during normal operation. If you wish to achieve a higher system transmission gain and better feedback suppression, it is recommended that you carry out troubleshooting by following the steps below:
(a) Reduce the system gain and use the 'Clear' button to reset all filter settings.
(b) Configure the settings for the feedback suppression module. Also reduce the panic threshold to lower the level of feedback.
(c) Switch on all microphones and slowly increase the system gain until feedback occurs. Stop increasing the system gain when feedback occurs.
(d) Wait for the feedback suppression module to take effect; once the feedback has stopped, continue to increase the gain.
(e) Repeat the process until the system reaches the required gain or until all filters have been fully distributed
(f) Set the panic threshold to the maximum level just above the expected feedback-free signal.
At this stage, if necessary, you can set each filter to fixed mode or save the dynamic state to manage any feedback during the performance. Furthermore, you can copy the filter to the band-stop filter module (such as an equaliser). In this way, you can increase the filtering capacity.
If a loudspeaker is among the equipment being used, it is recommended that a compressor/limiter module be used for additional protection. You can set an appropriate limiter to ensure that the loudspeaker is not damaged, even if all the band-stop filters are saturated or if the feedback suppressor fails to control feedback effectively, for example in the event of excessive system gain.

## 5 - 8 - Noise Gate




The main purpose of a noise gate is to attenuate signals below the threshold, and this attenuated signal is generally noise.

**Threshold**: Above the cut-off point, below the attenuation point.

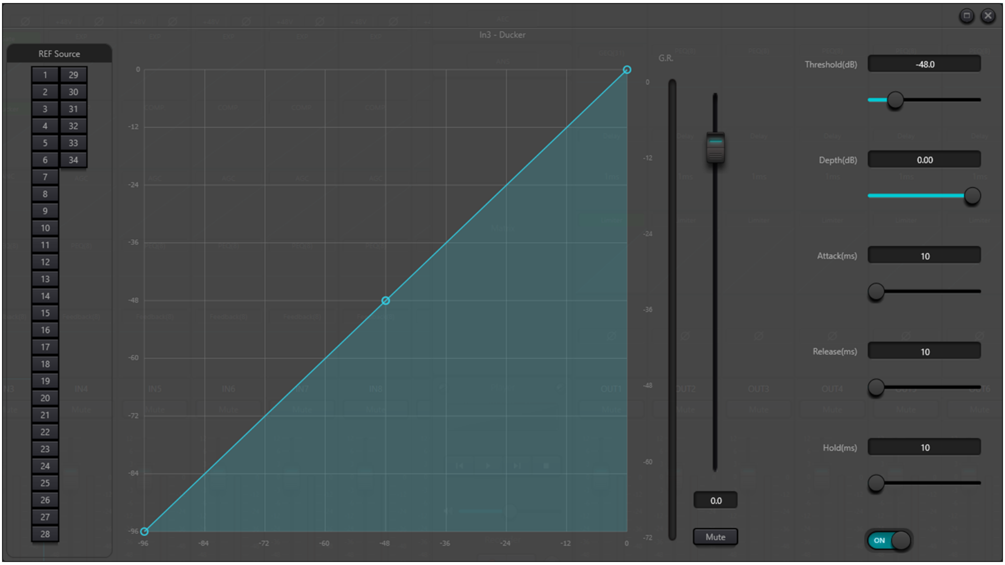
**Depth**: Attenuation, which determines the extent to which the signal below the threshold is attenuated.

**Attack (ms)**: The attack time refers to the speed at which the noise gate opens.

**Release (ms)**: The release time is the opposite of the attack time; it refers to the speed at which the noise gate closes.

**Hold (ms)** : The hold time works as follows: when the signal falls below the noise gate threshold, it would normally begin to attenuate immediately. However, if you set a hold time of 10 ms, the noise gate will maintain the original output level for 10 ms, and it is only after these 10 ms that the signal will begin to attenuate.

## 5–9 – Ducker




When the level of one channel exceeds the specified threshold, the level of the other channel is attenuated.

**Threshold**: The reference signal begins to attenuate above the threshold and returns to its initial level below the threshold.

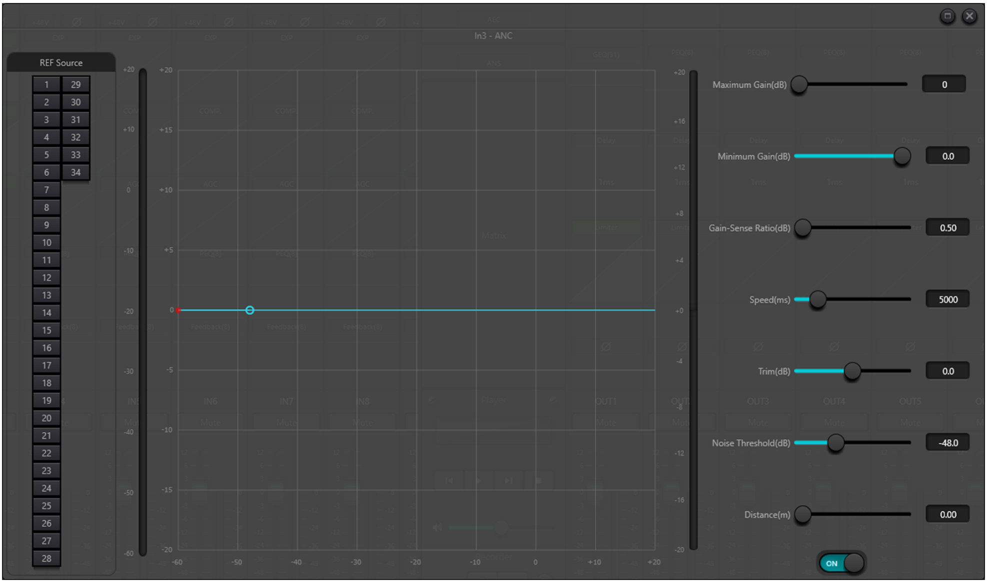
**Depth**: The reduced value of the avoidance signal.

**Attack (ms)**: The attack time refers to the speed at which the noise gate opens.

**Release (ms)**: Once the reference signal falls below the threshold, the avoidance signal returns to its original level.

**Hold (ms)** : The hold time is the duration for which the attenuation on the attenuation channel persists after the control signal has fallen below the threshold.

## 5–10 – ANC




Automatically adjusts the output volume based on the detection and processing of ambient noise.

**Maximum gain**: The highest value that can be set.

**Minimum Gain**: The lowest gain setting that can be selected.

**Depth**: The reduced value of the avoidance signal.

**Gain sense ratio**: The ratio between the increase and the attenuation.

Speed: Once the reference signal falls below the threshold, the avoidance signal returns to its original level.

**Trim**: Gain.

**Noise threshold**: Higher than the initial gain, lower than the reduction.

**Distance**: The distance between the reference and local signals.

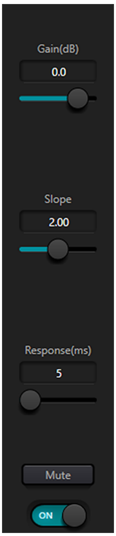
## 5 - 11 - Automixer

In a conference room, if several microphones are set to the same gain level and only one person is speaking, the sound from the microphone may not be clear. The other microphones will pick up background noise and reverberation from the room. When these signals are mixed with the normal microphone signals, the quality of the audio output is significantly reduced and the entire amplification system can easily produce a whistling sound, resulting in a loss of sound transmission gain. To resolve this issue, the other unused microphones must be switched off. The AutoMixer can switch them off more quickly than manual operation.

The processor features a gain-sharing AutoMixer. It supports audio signal output of up to 32 channels.
Each channel in the automatic mixing matrix has a direct output, which is affected only by the channel's mute switch, and not by the automatic gain or the channel fader. Channels requiring a fixed volume, such as those for background music, must remain at a constant level without being controlled by the AutoMixer. For example, the microphone should normally remain unmuted. In this case, its gain will not be affected by the AutoMixer. At this stage, users can adjust the channel's output directly in the output matrix routing and disable the channel's AutoMixer button. Its gain will not be adjusted, and the gains of the other channels will not be affected by the signal level of this channel.

The AutoMixer module comprises two groups of control parameters: the main control parameters and the channel control parameters.

(1) Key control parameters



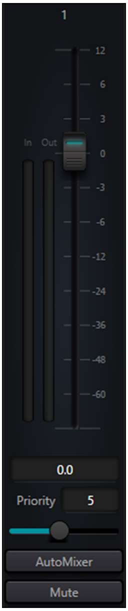

Click the button at the bottom to open or close the AutoMixer.

**Gain (dB)**: Controls the main output volume of AutoMixer.

**Slope**: the slope control affects the attenuation of low-level signals. The higher the slope, the greater the attenuation of low-level channels. The slope control and the ratio control of
The expanders all work in the same way. It is recommended to set the value to 2.0 or thereabouts. If set to 1.0, the effect is equivalent to disabling AutoMixer on all channels. If set to 3.0, this will result in a more significant gain adjustment, which may produce an unnatural effect. The higher the value, the more the channel is opened and the greater the total attenuation. When the slope is set to 2.0, an ideal gain split can be achieved, so this is the recommended value.

**Response (ms)**: Response time: A faster response time helps to prevent the beginnings of sentences from being cut off. A slower response time ensures smooth operation. Experience shows that the best results are achieved when the response time is between 100 ms and
1,000 ms. The automatic gain control is designed so that the microphones switch on more quickly than they switch off. Consequently, the first few letters of speeches will not be cut off even if the response time is 100 ms. If it is set to several seconds, the AutoMixer's response hold time will be longer, and the previously active channel will remain open for several seconds.

(2) Channel control settings




Click the button at the bottom to open or close the AutoMixer.

**AutoMixer**: Each channel has an AutoMixer on/off button, which must be switched on for the channels to participate in the AutoMixer. It can also be switched off, in which case the channel will not participate in the AutoMixer.

**Mute**: Mute: Both the mute switch and the channel fader are subject to automatic gain control. If the channel level is higher, the gain on the other channels may also be reduced, even if the channel mute is activated.

**Gain**: Adjusting the gain fader can increase or decrease the volume level in the AutoMixer.

**Priority**: The priority setting can give higher priority to high-priority channels over low-priority channels, which will affect the AutoMixer algorithm. The priority setting ranges from 0 to 10. The higher the value, the higher the priority.

Both the channel mute and fader are located downstream of the automatic gain control. Any adjustments made to these two parameters will not affect the operation of the AutoMixer. For example, if the level of one channel is higher, the level gain of the other channels may also be reduced, even when the channel's mute function is activated. You should activate the channel mute and deactivate the AutoMixer to cut the signal and prevent it from affecting the AutoMixer. The mute button for each channel must be activated and the output muted directly whilst mixing the audio. The channel faders also control the audio mix level and the direct output level of the channels. Click on the text box and enter a value in dB to adjust the channel level precisely.

Priority control allows high-priority channels to override low-priority channels, which will affect the algorithm in AutoMixer. The priority value can be set from 0 (lowest priority) to 10 (highest priority), with the default being 5 (standard priority). Users can use the slider or click on the text field to enter a specific priority value between 0 and 10 in order to adjust the priority. A higher value means a higher priority.
If two channels have the same signal level, the channel with the higher priority will receive a higher automatic gain. If there is a difference of one priority unit between them, the channel with the higher priority will receive an additional 2 dB of mix gain (assuming that the slope for both channels is set to 2.0). For example, if the priorities of channels 1 and 2 are set to 6 and 3 respectively, and the input levels of these two channels are identical, channel 1 will receive an additional 6 dB of audio mix gain compared to channel 2. When using this feature, it should be noted that adjusting the slope of the main control parameters will also influence the difference in audio mix gain resulting from the channel priority weighting. If the slope is set to 3.0, a difference of one priority unit will result in a gain difference of 4 dB. If all channels have the same priority, their priority settings should be set to 5.

Note: In some cases, users should exercise great caution when using extreme differences in priority between channels, such as priorities of 0 and 10. If channels with very high priority detect signals such as background music coming from a loudspeaker, they may mask channels with lower priority, even if the former are not in use. This phenomenon is exacerbated by a steeper slope. If the problem occurs during installation and commissioning, users may consider installing a noise gate or an expander between the AutoMixers on the channels with the highest priority. At the same time, they should set the threshold to a level that will not be triggered by the noise gate or expander.

## 5–12 – AEC

Acoustic Echo Cancellation (AEC) is a digital audio signal processing technology. It is used in audio and video conferencing when participants in the local conference room are speaking with one or more people located some distance away. The AEC programme improves the speech intelligibility of the remote speaker by cancelling out the acoustic echo generated in the local room.

The echo cancellation module for remote calls can be used to amplify remote voice signals locally and reduce interference caused by acoustic echo. Its basic operating principle involves simulating an echo channel, estimating the echo likely to be generated by the remote signals, and then subtracting the estimated signal from the input signal from the microphones; in this way, no echo is generated in the input voice signal, thereby achieving the objective of echo cancellation.

The DSP controller contains only one echo cancellation module. Two mixers for local input and remote output are preset to allow multi-channel signals to be included in the echo cancellation, as shown in the figure. One parameter can be adjusted: Non-linear filter (NLP): three types – Conservative, Moderate and Aggressive – can be selected to determine the level of echo suppression.


Version for the Ami08 matrix

Version for the Ami08D matrix


Note: The echo cancellation module settings must be used in conjunction with the signal router in the matrix module.

## 5–13 YEARS OLD

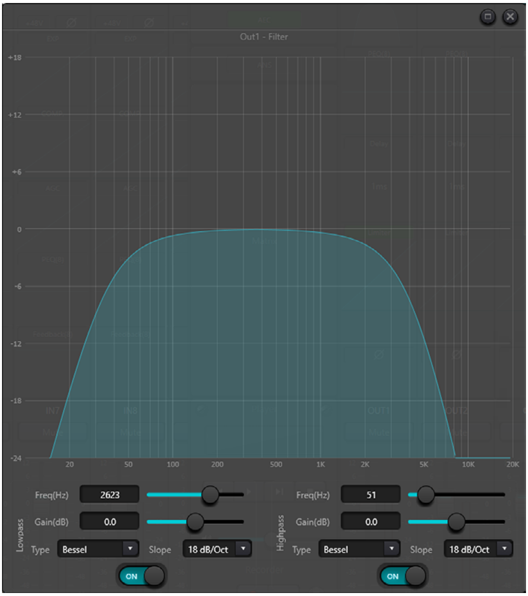
The noise-cancelling module effectively removes non-vocal sounds. It is capable of distinguishing the human voice from non-vocal sounds and treating the latter as noise. After processing, in theory only the human voice remains in an audio file containing both the human voice and noise.
The DSP controller contains only one echo cancellation module. The multi-channel mixers are preset to perform multi-channel noise cancellation, as shown in the figure.


Attenuation level: there are three levels in total, namely Low (6 dB), Medium (10 dB) and High (15 dB). The dB here refers to the reduction in noise measured in decibels. The higher the value, the greater the degradation of the voice, which is inevitable.

## 5 - 14 - Matrix

The matrix has two functions: routing and audio mixing. As shown in the figure, the horizontal axis indicates the input channel and the vertical axis the output channel. The default setting is a one-to-one correspondence between the input and the output. If you need to mix the audio from channels 1 and 2 and then route it back to channel 1, simply click ‘1’ in both the horizontal and vertical directions at output channel 1. If inputs 1 and 2 are set to automatic mixing, the output will not be affected. Similarly, after configuring the automatic mix, echo cancellation and noise suppression module, the user must also configure the matrix to ensure the correct signal routing.

## 5 - 15 - High-pass and low-pass filters




Each output channel has high-pass and low-pass modules. Each filter has four types of parameters, as follows:

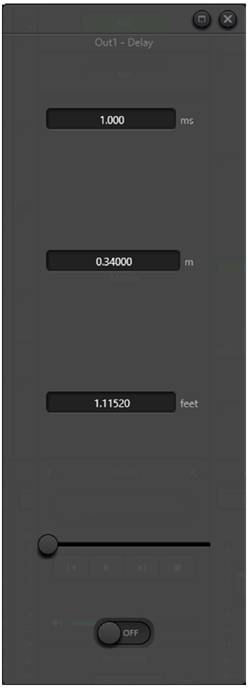
**Freq(Hz)**: Cut-off frequency of the filters. The cut-off frequency of Bessel and Butterworth filters is set to -3 dB, and that of the Linkwitz-Riley filter at -6 dB.

**Gain (dB)**: The gain setting affects the amplification and attenuation of the passband.
**Type**: There are three types of filter: Bessel, Butterworth and Linkwitz-Riley. The Butterworth filter has the flattest frequency response.

**Slope**: This refers to the attenuation values in the transition region of the filters. There are a total of 8 attenuation values: 6, 12, 18, 24, 30, 36, 42 and 48 dB/oct. For example, 24 dB/oct indicates that the attenuation range is 24 dB for each octave difference in frequency within the transition zone.

Users can click on the activation button at the bottom to enable the high-pass or low-pass filter.

## 5 - 16 - Delay




**Enable button**: Enables the delay module selected in the modules and inserts it into the audio signal path to increase the fixed delay time of the signals.

**ms/m/feet**: Sets the delay time. The value must be between 1 and 1,200 milliseconds. 'Metres' and 'feet' are alternative units to milliseconds.

## 5 - 17 - Exit

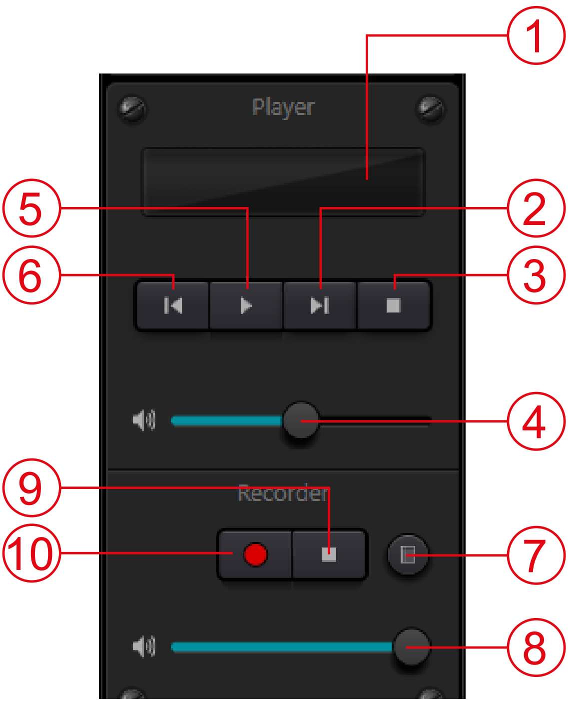

**Phase**: Reverses the phase of the audio signal by 180 degrees.

**Mute**: Turns the mute function on or off.

Similarly, users can use the button on the right to configure the output channel menu, which can be done as required.

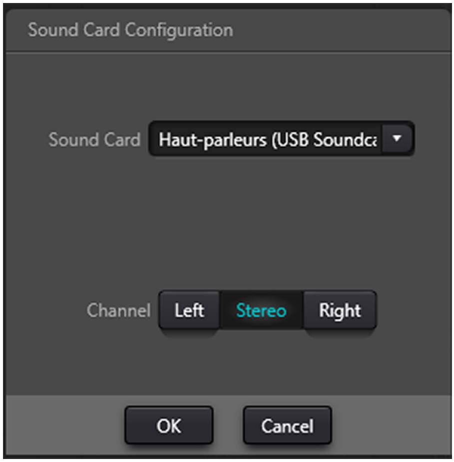

## 5 - 18 - USB sound card

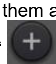
The USB sound card is used for two purposes: recording and broadcasting, as well as teleconferencing via personal computers. After passing through the echo and noise cancellation modules, the USB audio can be easily used for teleconferencing. The USB broadcasting function on the software interface can only be used for recording and broadcasting.




1 – Information about the track; double-click to access the playlist.
2 – Next track.
3 – Pause.
4 - Adjusting the volume.
5 - Play.
6 – Previous track.
7 - List of tracks recorded.
8 - Adjusting the recording volume.
9 - Stop recording.
10 - Start recording.

A USB cable with two Type-C connectors is used to connect the DSP processor to the host computer. When connecting for the first time, the message 'New hardware detected' will appear on the screen and the driver will install automatically. Once installation is complete, the USB sound card will appear in the computer's list of sound cards, as shown in the figure. Users can select the USB sound card from the sound card settings within the software's playlist.




Users can manage music files in a playlist and save them as a playlist. They can also open them directly the next time they use the device. As shown in the illustration, click on '' at the bottom of the playlist to open the file folder and select the tracks to play; click on '' to clear the playlist, or on '' to access the sound card configuration interface.

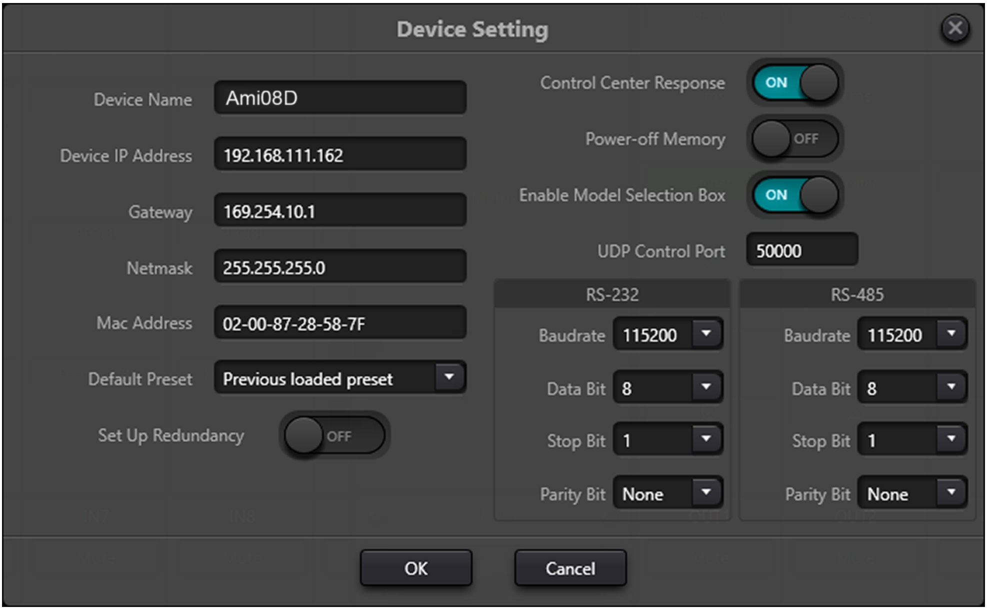
# 6 - Device Settings menu

## 6 - 1 - File menu


In offline mode, click on the file dialogue box that appears and open an existing default document with the *.amidsp extension, or right-click on the default document to open AmiControl.

'Save as...' allows you to save the application's presets to the local hard drive to make copying and storage easier.

## 6 - 2 - Device settings




Information such as the device name, network address and serial port baud rate can be configured in the device settings. The maximum length of the device name is 16 characters.

Default start-up: Two start-up preset modes are available. The first involves selecting one of the 16 presets as the start-up preset. The unit will start up using this preset each time. The second involves selecting the previously loaded preset and using the last preset active before the power cut as the preset for the next start-up.

Set Up Redundancy: With two identical matrices, it is possible to set up a redundancy (backup) system. The slave matrix will receive exactly the same commands.

## 6 - 3 - GPIO settings

Open the GPIO settings software interface. The device has a total of 8 GPIOs, which can be configured independently as inputs or outputs.
The input GPIOs offer the following options: Preset, router, gain, mute, control and analogue-to-digital gain.
The output GPIOs offer the following options: Preset, Level, Mute and Command.



### 6 - 3 - 1 - Configuring the GPIO inputs

| | | |
|---|---|---|
| Preset | <br> | Trigger types: High-level trigger / Low-level trigger<br><br>Preset: When the input control type of the hardware GPIO port matches the trigger type defined in software, the matrix switches to this preset. |
| Routing | <br> | Trigger types: The same as those listed above<br><br>Input and output: Select the mix of input channels that corresponds to the output channel.<br><br>Performs the mix/unmix action when the trigger condition is met. |
| Gain | <br> | Trigger types: the same as those listed above<br><br>Channels: select the input or output channel<br><br>Step size: increase the step size in dB in line with the original gain achieved by the channel. |

| | | |
|---|---|---|
| Turn Mute on / off |  | Trigger types: The same as those listed above<br><br>Channels: Select the input or output channel |
| Order |  | Trigger types: The same as those listed above<br><br>Command: When the conditions defined by the trigger type are met, the matrix sends the command code via RS232. |
| Analogue-to-digital gain |  | Analogue-to-digital gain is very useful when connecting an external potentiometer. It allows you to adjust the gain of the input or output channel. It is similar to a rotary encoder. The difference between the two is that the potentiometer is analogue and controls voltage and current, whilst the encoder is digital and transmits the binary codes 0 and 1. |

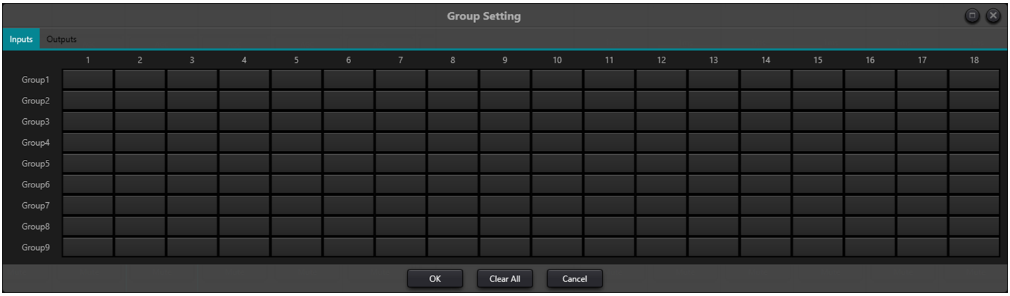
**6 - 3 - 2 - Configuring GPIO outputs**

| | | |
|---|---|---|
| Preset |  | Output types: high level/low level<br><br>Preset: The corresponding GPIO pin outputs a high or low level when switching to this preset. |
| Level |  | Output types: high level/low level<br><br>Channel: Designated input or output channel<br><br>Level: The GPIO outputs produce a high or low level when the level of the relevant channel reaches the predefined threshold. |

| | | |
|---|---|---|
| Mute |  | Output types: high level/low level<br><br>Channel: Designated input/output channel. The predefined high/low level is output when the channel is muted. Conversely, the opposite level will be output when the mute is deactivated. |

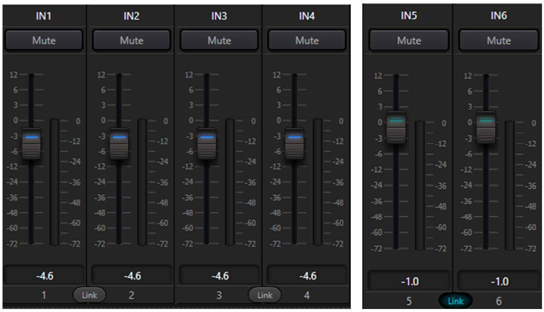
## 6 - 4 - Group settings

The group settings interface comprises two tabs: Input and Output. A maximum of 9 groups for the Ami08D and 5 groups for the Ami08 can be defined for each tab. A channel can only belong to one group. Within the same group, volume control and channel muting are synchronised. The module’s other settings are not synchronised, which is the main difference between this feature and the Link function.



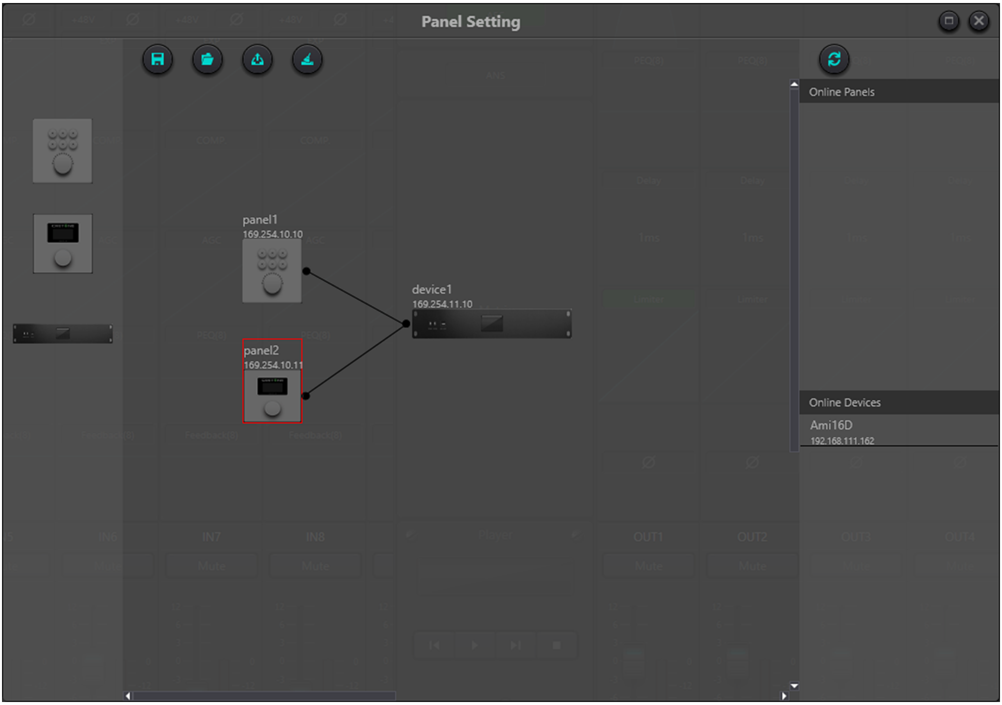

There are a total of 5 or 9 groups. The maximum number of channels per device can be selected for each group. The maximum number of channels is determined by the type of device you have purchased. The channels are grouped together in a group, which will be colour-coded in the main interface.

Relationship between groups and LINK: A channel that is part of a group will not participate in LINK, which means that the group takes precedence over LINK. The difference between groups and LINK is that groups can only control the gain and muting of channels, whilst LINK links all settings at channel level.



## 6 - 5 - Connecting and configuring the AmiCTL1 and AmiCTL2 wall-mounted panels

The connection settings for wall-mounted panels cover two types of panel: button panels (AmiCTL1) and OLED panels (AmiCTL2). Use cables to connect several physical panels to the DSP unit, then configure each control on the DSP unit via the panel by easily setting up the panel.




**Offline device**:
This option is suitable for offline editing. First, the installer configures the panel settings locally, then uploads them to the online panel. It is, of course, possible to edit the panel directly online. Drag the offline device from the online panel column into the panel design area, then double-click to edit it.

**Connecting devices online:**
In Panel Setting, you will see the controllers connected to your network (under Online Panels).
Select the controller you wish to configure. First of all, you must configure its network settings. Click on 'Set IP', assign it an available IP address and enter the same subnet mask and gateway as the matrix.

All you then need to do is connect the controller to the matrix virtually…

Please note that there is a small circle on both the panel and the matrix. Click on the circle, then draw a line and select the target matrix; this establishes the connection between the two elements.

Double-click on the panel in the design area to access the panel configuration interface. The configuration of two panels is described below. Once configuration is complete, click on the ' ' download icon on the toolbar to download the panel configuration to the hardware.

### 6 - 5 - 1 - OLED panels: AmiCTL2

The AmiCTL2 panel consists of a 1.3-inch OLED screen and a rotary push-button. The OLED display provides access to five types of menu: You can configure the controller menu (Gain, Mute, Preset, controls or matrix). Double-click the AmiCTL2 controller in the design area of the Panel Setting module to access its detailed settings, as shown below.



How to use the panel:

Click on ‘Add menu’ to bring up the menu selection box, select the relevant menu and confirm. Once you have finished configuring the software menu, click on the download icon on the toolbar to download the configuration to the controller.

1. The panel name and IP address are displayed on the main interface; turn the knob to the left or right to change menus.
2. Press the button on the rotary control: The second line of the menu interface starts to flash, indicating that edit mode is active.
3. Turn the knob to the left or right to change the value.
4. Press the rotary knob button again to exit edit mode and return to menu mode.

Access to AmiCTL2 can be protected by a 4-digit password.
Two passwords can be set. The first is for settings and the second is for control.

**Password setting**: 4 digits
To access this menu, press and hold the button (for 10 seconds) and then enter the default password (4816). Select the ‘Password Set’ menu, then click ‘Yes’ to change your password.

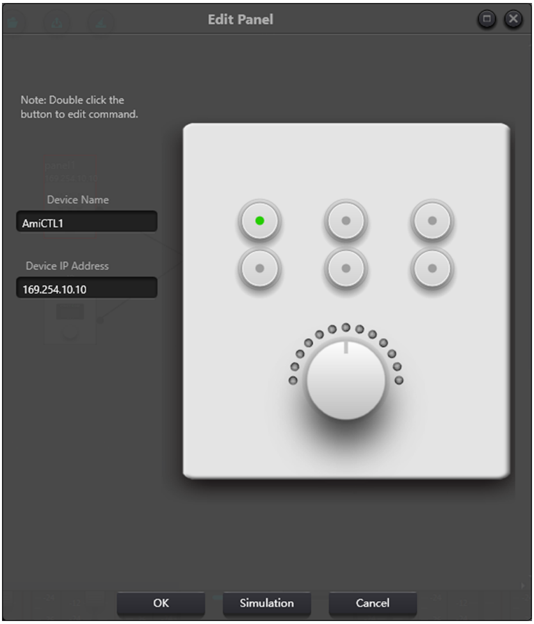
**Verification password**: 4 digits
If the installer ticks the ‘enable password’ box in ‘edit panel’, the 4-digit code will be required on the AmiCTL2 controller to access its control functions.
This password can be changed directly in the System Config menu of the AmiCTL2.
To access this menu, press and hold (for 10 seconds) and then enter the default password (4816). Select the ‘Ctrl Password’ menu to enable it and enter your new password.

If you enter the wrong password by mistake, you can use the master password: 4816.

### 6 - 5 - 2 - Keypad panels: AmiCTL1

AmiCTL1 features 6 buttons and a rotary encoder with a push function. The knob is used to adjust the gain, and the 6 buttons can be programmed to perform various functions. There are five types of button functions, including volume control, mute, presets and commands. Double-click on the button you wish to programme to select its function and mapping.
Similarly, once programming is complete, users can use the emulation button to check whether the configuration is correct.

How to use the panel:




1. The key indicator remains lit, indicating that the key is set to the mute function.
2. If the button's indicator light is flashing, this indicates that the button is configured with the gain function. The configured button also allows you to use and adjust the channel's gain. The 13 LEDs around the button indicate the gain levels. They light up or go out depending on the gain level. When all 13 LEDs are off, the gain is -72 dB, whilst when they are all on, the gain is 12 dB.
3. A sudden flash when the key is pressed indicates that it is configured with the preset or control function.

## 6 - 6 - User interface settings

The DSP controller offers a customisable user interface, enabling the installer to create bespoke interfaces that can be modified by system integrators and used by on-site technicians or end users who are not familiar with the technology. It supports wireless remote control via iPad, tablet and mobile phone, and runs on Windows, Android and iOS.

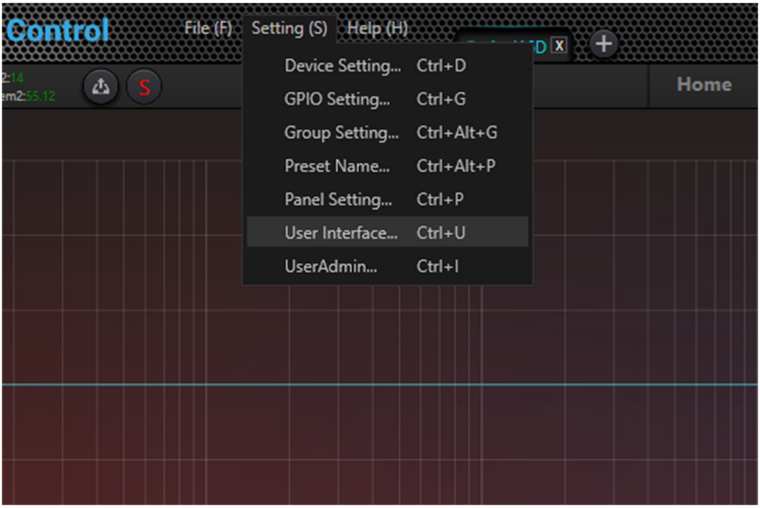
### 6 - 6 - 1 - Downloading the app

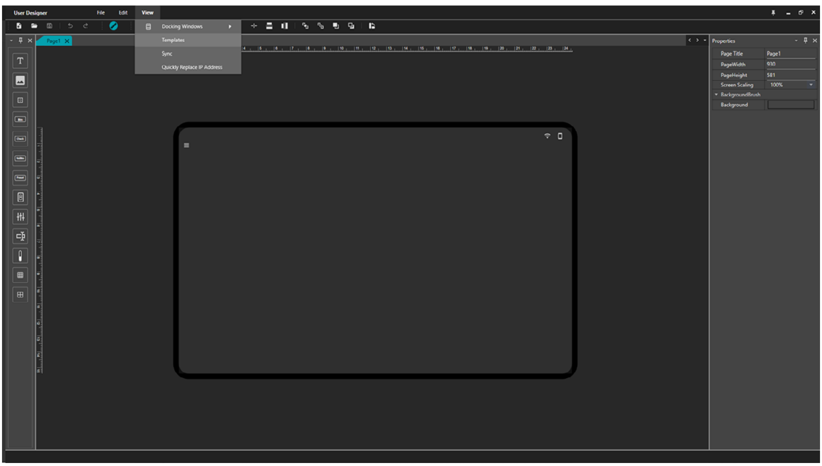
iOS: Search for the AmiControl app in the App Store and install it.
Android system: Search for the AmiControl app on the Google Play Store or download the installation file (.APK) from our website.

It should be noted that the application interface is blank after installation and that the relevant content must be edited using the User Designer module in the Ami Control PC software before being downloaded to the terminal.

### 6 - 6 - 2 - Modifying the user interface

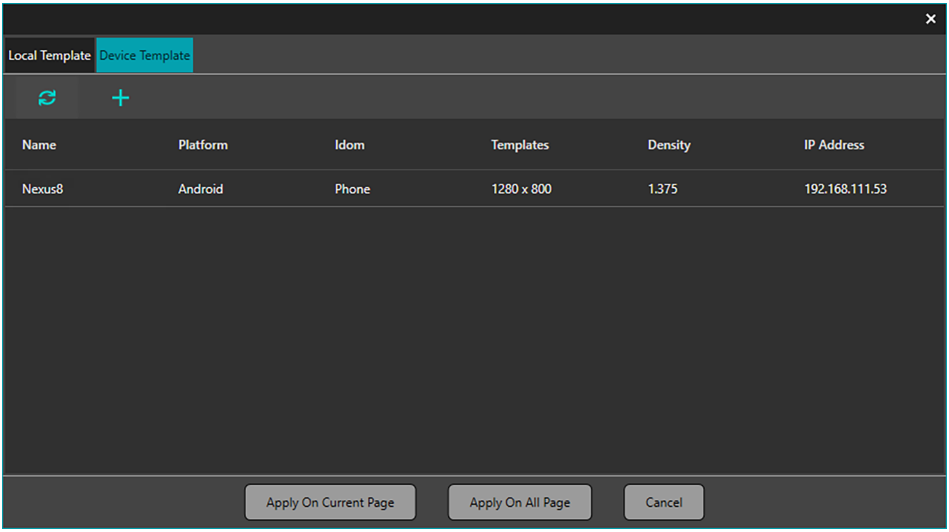
Access the editing area via the AmiControl software settings (Settings (S)) – User interface.



### 6 - 6 - 3 - Selecting the model

We recommend matching the screen resolution and density using the Device Template tool and the online templates. Click on 'Device Template' to automatically search for information about devices online. Please ensure that your phone or iPad is on the same local network as your computer and that the app is already open.



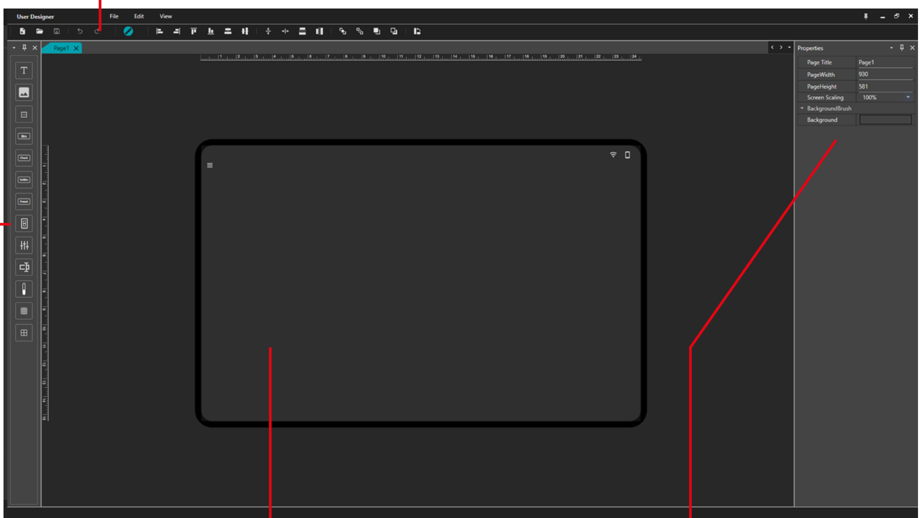

**Template**
**Device**: Allows you to create screen templates by specifying the device type, resolution and screen density
**Device**: Scans the network to find devices with the Ami Control app open.
All devices connected to the same network will appear on the list. Simply select the device and its specifications will be automatically applied to the relevant page(s) in User Designer.

### 6 - 6 - 4 - Editing functions

User Interface Designer

Top bar elements: Project management and layout.

Sidebar elements: Suite of graphic tools.



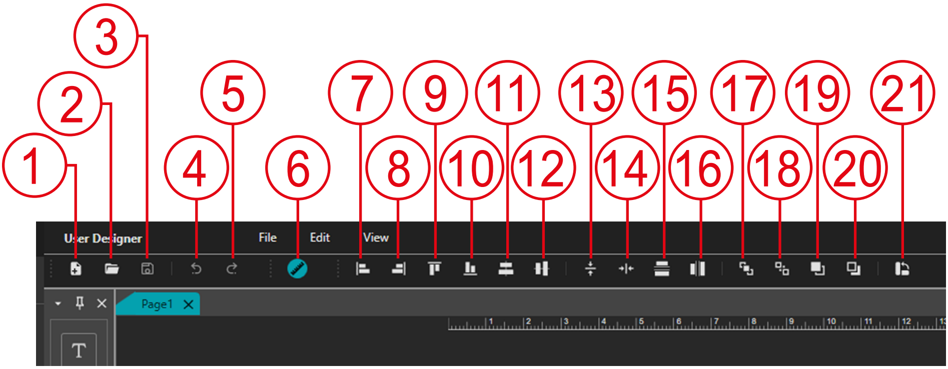

Graphic area showing the project currently being created.

Property of the selected element

**Sidebar elements**
Functions can be added by dragging and dropping

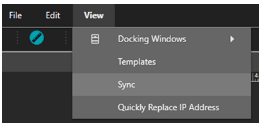
| | |
|---|---|
|  | **Text Button**: Adding text or a title<br>Options for customisation: Font size and colour, insert size and colour |
|  | **Background button**: Add an image (jpeg, jpg, gif, bmp or png)<br>Options for customisation: Frame size and colour. |
|  | **Frame button**: Add a frame<br>You can change the size and colour of the frame and the background colour. |
|  | **Btn button**: Press to send custom commands. Set parameters such as the button’s size and colour in the properties bar, and choose from three different protocol formats in the command bar;<br>Protocol format: RS232, RS485, UDP. The command format is hexadecimal. |
|  | **Check button**: Adds a system control button (mute, phase or matrix) or a customised control (UDP, RS232 or RS485) with two states.<br>You can change: the size and colour of the frame, the background colour, and the size and colour of the text. |
|  | **VolBtn button**: Adds a volume + or volume – button assigned to an input or output.<br>You can customise: the size and colour of the frame, the background colour, and the size and colour of the text. |
|  | **Preset button**: A drop-down menu has been added for selecting presets.<br>Options to customise: Button size and the available presets. |
|  | **Channel button**: Add a channel (fader, VU meter and mute).<br>Selecting the input or output channel.<br>You can change: the size and colour of the button. |
|  | **Fader button**: Adds a fader (gain).<br>Selecting the input or output channel.<br>You can change: the size and colour of the button. |
|  | **Volume Input Button**: A volume input button has been added.<br>Selecting the input or output channel.<br>You can customise: the size and colour of the button, as well as the text colour. |
|  | **Peak meter button**: Adds a peak meter.<br>Selecting the input or output channel.<br>Options for adjustment: Size, upper and lower alarm levels. |
|  | **Matrix button**: Add a matrix (4x4 to 64x64).<br>Selection of the routing for each button, with the option to add UDP, RS232 or RS485 commands.<br>You can customise: the name, size and colour of the buttons and text. |
|  | **Static matrix button** : Add a static matrix (4x4 to 64x64).<br>Selecting the routing for each button; only the matrix gain will be displayed.<br>Options for customisation: Size, colour of the buttons and gain. |

**Top bar elements**




**1** - Add a new page
**2** - Open a project
**3** - Save the project
**4** - Undo the last action
**5** - Redo the last action
**6** - Turn the grid on or off
**7** - Align the selected items to the left
**8** - Align the selected items to the right
**9** - Align the selected items to the top
**10** - Align the selected items to the bottom
**11** - Centre the selected items vertically
**12** - Align the selected items horizontally
**13** - Arrange the selected items vertically without any spacing between them
**14** - Spread the selected items horizontally without leaving any space between them
**15** - Space the selected items evenly vertically
**16** - Space the selected items evenly horizontally
**17** - Highlight
**18** - Move to the back
**19** - Bring the selection back to the front
**20** - Send the selection to the back
**21** - Change the page orientation (landscape or portrait)

**View menu**

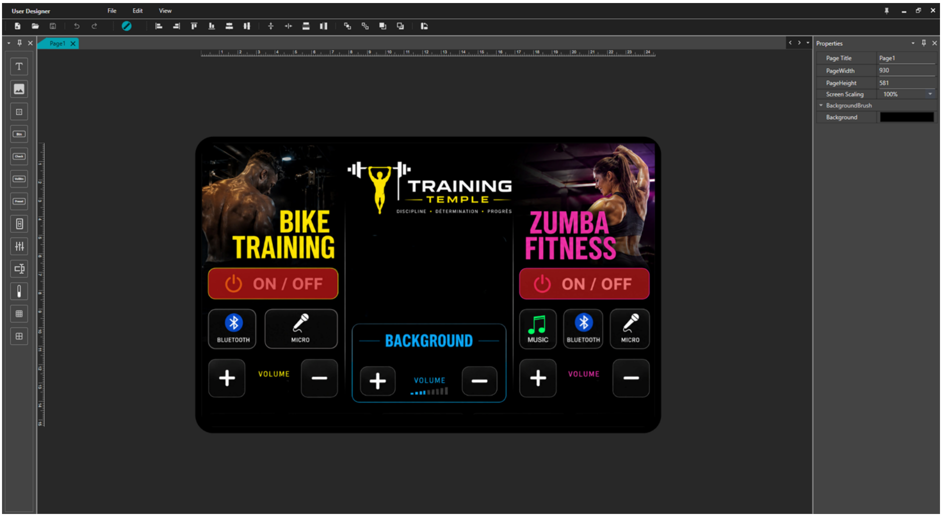



**Template**
**Device**: Allows you to create screen templates by specifying the device type, resolution and screen density
**Device**: Scans the network to find devices with the Ami Control app open.
All devices connected to the same network will appear in the list. Simply select the device and its specifications will be automatically applied to the User Designer page(s).

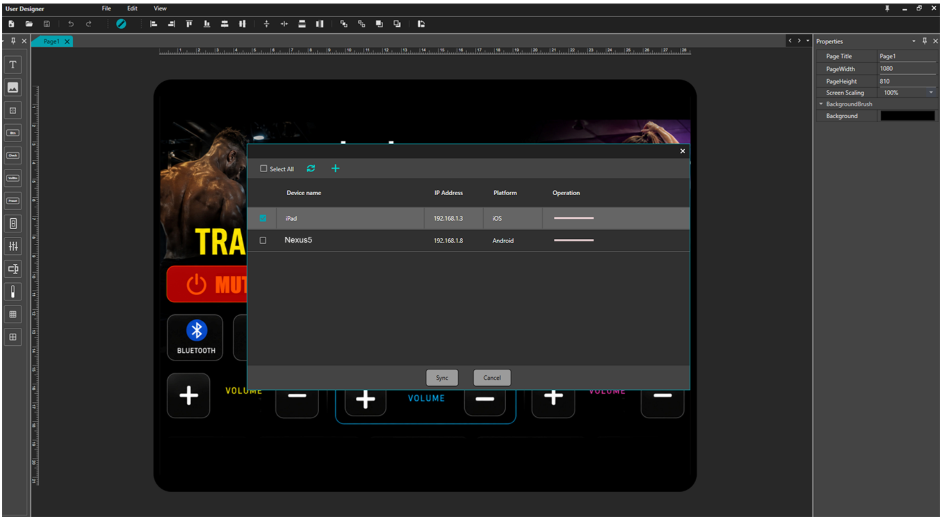
**Sync**:
Once the user interface has been created, select the device you wish to synchronise (on the same network and with the app open)
And start the process.

**Quick IP address replacement**: Allows you to replace the IP addresses of all buttons or functions on the page(s) with a single click.

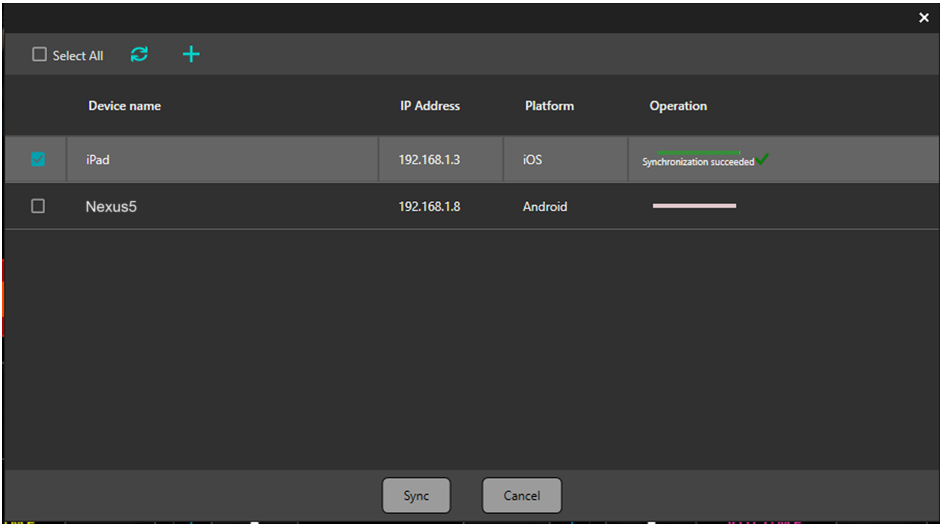
## Example of a user interface layout




## Download the user interface



**Download successful**




Please note that the mobile device and the matrix must be on the same local area network and the same network segment, and that the app must be open. The control attribute must be set to the correct IP address of the matrix. If you are unable to control it, please exit the app and try again, or restart the wireless router.

# 7 - DANTE configuration

## 7 - 1 - Introduction to the DANTE system

Dante technology, launched by Audinate, offers a high-performance digital multimedia network designed to meet professional requirements for sound amplification and audio/video installation, as well as the demands for high sound quality and high performance in broadcast and recording systems.
Dante aims to fully utilise the capabilities of current and future network equipment. It offers a multimedia transmission mechanism capable of overcoming the design constraints of many traditional audio networks. Dante facilitates the deployment of a stable and flexible digital audio network, whilst offering virtually unlimited performance. The Dante network can be designed with data rates combining Gbps and 100 Mbps; it supports audio signals with different sample rates and bit depths, and even allows network zones to be designed with variable latencies.

Dante is based on the Internet Protocol, not just on Ethernet. Dante uses the standard IP protocol over Ethernet, enabling it to operate on existing, low-cost IT network hardware. At the same time, thanks to QoS standards, Dante can share the existing network with other data streams and IT traffic.
Dante offers precise, sample-accurate synchronisation as well as the low latency required by professional audio. The core of the Dante network is independent of other synchronised audio streams and enables perfectly synchronised playback across different audio channels, devices and networks, and even across multiple switch hops.

Dante makes networking a truly ‘plug-and-play’ process and enables automatic device discovery and configuration. Dante-compatible devices automatically define their own network configuration and identify each other and their channels on the network. This simplifies complex, error-prone configuration procedures and eliminates the need for ‘magic numbers’. Network devices, along with their input and output signals, can be renamed to make them easier to understand.

Dante is not limited to the configuration and transmission of audio channels. It also provides, via its IP network, a mechanism for sending or receiving control and monitoring information, including device-specific information and commands defined and developed by the manufacturers of those devices. Building on its solid foundations and an existing, constantly evolving network standard, Dante offers cutting-edge technology, without which it would not be possible to use it in other types of digital audio transmission. From the outset, Dante was designed for Gigabit networks. Furthermore, the current version of Dante incorporates the emerging AVB network standard. The ongoing evolution of its network technology is an integral part of Dante’s development.

Dante technology can be used for hardware and software installation, as well as for APIs relating to design and development. For further details, please visit the Audinate website: www.audinate.com.

**Features:**
Based on current IP network technologies, notably the IEEE 802.3 and UDP/IP standards, it utilises existing Ethernet hardware.
Uses standard VoIP (Voice over IP) Quality of Service (QoS) to integrate with existing networks.
Incorporates Ethernet network speeds and enhances them from 100 Mbit to 1 Gb.

The DSP’s digital audio resolution is 24-bit, with a sampling rate of 48 kHz. Dante allows both the sampling rate and bit depth to be seamlessly integrated simultaneously on the same network.
The Dante technology in DSP devices supports network latency as low as 0.25 milliseconds. For device detection, the ER ensures plug-and-play operation. Automatic configuration. Tag-based routing. Data streams that can be renamed.

The number of Dante channels included in the Ami matrix depends on the type of device purchased. There are numerous versions available, including the Ami04D, Ami08D and Ami16D.

Use the DANTE Controller software, downloaded from Audinate, to connect a PC or Mac directly to the Dante network.

The DANTE network is configured using the DANTE Controller software. By default, the DANTE card is set to DHCP.

## 7 - 2 - System requirements for Dante

CAT6 cables are used for all internal connections within the Dante system.
If bandwidth control is implemented on the same network, 30 per cent of the available bandwidth must be reserved. Using this reservation method, a 100 Mbit link can handle up to 48 × 48 channels. A 1 Gbps link can handle up to 512 × 512 channels when the sampling rate is 48 kHz.

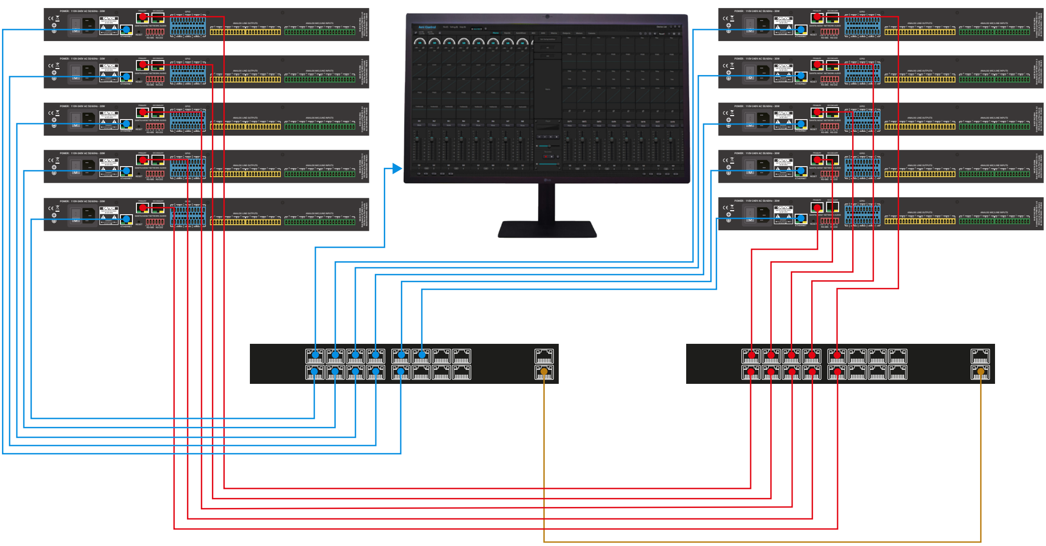
Serial connections and uplinks must be Gbps. Repeaters are not supported.

## 7 - 3 - Dante network design

There are two network diagrams that can be used for a standard Dante topology.

**10 non-redundant units or fewer**
For systems configured with up to 10**non**-redundant units, please connect your PC to an Ethernet port, daisy-chain the remaining Ethernet ports, and then connect the Dante ports to these same daisy-chains without needing to use a Dante switch or carry out any special configuration. All units must operate in switch port mode.

**More**than **10 units or a**distance **of more than 100 metres between them**
For systems with more than 10 units or where the distance between them exceeds 100 metres, please connect your PC or the Ethernet ports of all units to an Ethernet switch, and the main Dante port to a second Ethernet switch. All units must operate in switch port mode.



# 8 - Technical specifications

| | Ami08 | Ami08D |
|---|---|---|
| Processor<br>*Processor* | ADI SHARC 21489 (x 1) | |
| Analogue inputs<br>*Analogue inputs* | 8 | 8 |
| DANTE digital inputs<br>*DANTE Digital inputs* | 0 | 8 |
| Analogue outputs<br>*Analogue outputs* | 8 | 8 |
| DANTE digital outputs<br>*DANTE analogue outputs* | 0 | 8 |
| Sampling rate/Digitisation bit depth<br>*Sampling rate/Digitisation bit* | 48 K / 24-bit | |
| Input gain<br>*Input gain* | 0/3/6/9/12/15/18/21/24/27/30/33/36/39/42/45/48 dBu | |
| Phantom power<br>*Phantom power* | 48 V (10 mA max.) | |
| Frequency response<br>*Frequency response* | 20–20 kHz ±0.3 dB | |
| Maximum input level<br>*Maximum input level* | +18 dBu | |
| Maximum output level<br>*Maximum output level* | +18 dBu | |
| Total harmonic distortion<br>*Total Harmonic Distortion* | < 0.003% @ 4 dBu | |
| Dynamic input range<br>*Input dynamic range* | 110 dB | |
| Dynamic output range<br>*Output dynamic range* | 112 dB | |
| Background noise (A-weighted)<br>*Background Noise (A-weighted)* | -91 dB | |
| Common-mode rejection ratio at 60 Hz<br>*Common-mode rejection ratio at 60 Hz* | 80 dB | |
| Channel isolation at 1 kHz<br>*Channel isolation @1 kHz* | 108 dB | |
| Input impedance (balanced)<br>*Input impedance (balanced)* | 5.4kΩ | |
| Output impedance (balanced)<br>*Output impedance (balanced)* | 600 Ω | |
| System delay<br>*System delay* | < 3 ms | |
| Supply voltage<br>*Operating power* | AC 110–240 V 50–60 Hz | |
| Maximum consumption<br>*Maximum Power Consumption* | 28 W | 30 W |
| Operating temperature range<br>*Operating temperature range* | 0 - 40° | |
| Dimensions (Width x Depth x Height) *Dimensions (Width x Depth x Height)* | 482 x 260 x 45 mm | |
| Weight<br>*Weight* | 2.8 kg | |