AP412

4 × 120W Power Amplifier with DSP

User Manual

Compliance Statements

FEDERAL COMMUNICATIONS COMMISSION INTERFERENCE STATEMENT

This equipment has been tested and found to comply with the limits for a Class B digital service, pursuant to Part 15 of the FCC rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. Any changes or modifications made to this equipment may void the user’s authority to operate this equipment. This equipment generates, uses, and can radiate radio frequency energy. If not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

The device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) this device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

FCC Caution: Any changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate this equipment.

E3914

KCC Statement

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Industry Canada Statement
This Class B digital apparatus complies with Canadian ICES-003.

CAN ICES-003 (B) / NMB-003 (B)

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https://www.audinate.com/legal/patents-and-trademarks

Dante®

RoHS
This product is RoHS compliant.

User Information

Online Registration

Be sure to register your product at our online support center:

International	http://eservice.aten.com

Telephone Support

For telephone support, call this number:

International	886-2-8692-6959
China	86-400-810-0-810
Japan	81-3-5615-5811
Korea	82-2-467-6789
North America	1-888-999-ATEN ext 4988 1-949-428-1111

User Notice

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The manufacturer of this system is not responsible for any radio and/or TV interference caused by unauthorized modifications to this device. It is the responsibility of the user to correct such interference.

The manufacturer is not responsible for any damage incurred in the operation of this system if the correct operational voltage setting was not selected prior to operation. PLEASE VERIFY THAT THE VOLTAGE SETTING IS CORRECT BEFORE USE.

Product Information

For information about all ATEN products and how they can help you connect without limits, visit ATEN on the Web or contact an ATEN Authorized Reseller. Visit ATEN on the Web for a list of locations and telephone numbers:

International	http://www.aten.com
North America	http://www.aten-usa.com

Package Contents

Check to make sure that all the components are in working order. If you encounter any problem, please contact your dealer.

- 1 AP412 4 × 120W Power Amplifier with DSP
- 1 rack mount
- 1 foot pad set (4 pcs)
- 8 3-pin Euroblock connectors with strain relief (3.5 mm)
- 2 4-pin Euroblock connector with screw lock (5.08 mm)
- 1 3-pin Euroblock connector (3.5 mm)
- 1 2-pin Euroblock connector (3.5 mm)
- 1 power cord
- 1 user instructions

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About this Manual

This user manual is provided to help you get the most from the AP412 unit. It covers all aspects of installation, configuration, and operation. Devices and accessories covered in this manual include:

Models	Product Names
AP412	4 × 120W Power Amplifier with DSP
AP901	2-CH Dante Expansion Card for AP Series
AP902	2-CH Mic/Line Pre-AMP Expansion Card for AP Series

An overview of the information found in the manual is provided below.

Chapter 1, *Introduction* introduces you to the AP412 (4 × 120W) Power Amplifier with DSP. Its purpose, features, installation considerations, and panel components are presented and described.

Chapter 2, *Hardware Setup* describes the steps that are necessary to quickly and safely set up your installation.

Chapter 3, *Operation* specifies how to process the web control and configuration for the unit through the Ethernet connection.

Chapter 4, *DSP Effects on Audio Output* describes how DSP processing is applied to the audio output signals, and how each DSP function affects the final output behavior.

Appendix, provides a list of safety instructions and precautions, contact information for ATEN technical support, product specifications, and other technical information.

Note:

- Read this manual thoroughly and follow the installation and operation procedures carefully to prevent any damage to the unit or any connected devices.
- This product may be updated, with features and functions added, improved or removed since the release of this manual. For an up-to-date user manual, visit http://www.aten.com/global/en/

Conventions

This manual uses the following conventions:

`Monospaced`	Indicates text that you should key in.
[ ]	Indicates keys you should press. For example, [Enter] means to press the **Enter** key. If keys need to be chorded, they appear together in the same bracket with a plus sign between them: [Ctrl+Alt].
1.	Numbered lists represent procedures with sequential steps.
♦	Bullet lists provide information, but do not involve sequential steps.
>	Indicates selecting the option (on a menu or dialog box, for example), that comes next. For example, Start > Run means to open the *Start* menu, and then select *Run*.
⚠	Indicates critical information.

Chapter 1
Introduction

Overview

AP412 is a 4-channel DSP-equipped amplifier that delivers 120W per channel and receives balanced / unbalanced mic / line-level signals to drive 4 / 8-ohm low impedance or 70 / 100V line loudspeaker systems. It also supports Lo-Z / Hi-Z in Bridge mode to deliver higher power.

Configurable via intuitive WebGUI / ATEN Audio Wizard, the built-in DSP allows Speaker EQ, 5-band EQ, compressor, delay, limiter, and up to 20 presets for precise tuning and extensive audio system protection. The ground-lift switch helps eliminate unwanted hum noises from power circuit. Immersive sound experience is delivered without compromise on power efficiency with the Class D design. Customizable Auto Standby mode enables AP412 to automatically switch to “sleep” when the signal level falls lower than -40 / -50 dBu for 10 / 15 / 25 minutes.

With the system protection circuit, the amplifier is prevented from damages led by shorted outputs, over / under-voltage, high-frequency overload, and overtemperature. Plus, AP412 will be muted when the heat sink temperature exceeds the limit. The fanless design avoids fan noise and prevents dust from building up inside the chassis.

AP412 becomes compatible with Dante audio when working with AP901 expansion card or receives two more channels of mic / line signals with AP902 expansion card. It can be integrated with ATEN VK Control System via RS-232 / Ethernet for comprehensive control from a distance. The 1U casing is available for rack mounting. AP412 assures reliable amplification with professional audio quality and higher power delivery for larger commercial spaces.

Features

- A 4-channel DSP-equipped amplifier that delivers 120W per channel
- Supports balanced / unbalanced mic / line-level inputs and drives 4 / 8-ohm low impedance or 70 / 100V line (in Bridge mode) loudspeaker systems
- +48V phantom power supply
- Built-in WebGUI-configurable Sigma®DSP allows speaker EQ, 5-band EQ, compressor, delay, limiter, and up to 20presets for precise tuning and system protection
- Class D design assures high power efficiency
- PFC (Power Factor Correction) Power Supply – Fly-back power supply design with PFC ensures efficient operation while reducing energy consumption
- Integrated protection circuit – Protects the system from damages caused by Amp short circuit, Amp output over/under voltage, high-frequency overload, and overtemperature
- Customizable Auto Standby mode – Amplifier automatically switches to "sleep" when the signal level falls lower than -40 / -50 dBu for 10 / 15 / 25 minutes
- Solo function – Allows users to isolate and monitor a particular input / output
- Expansion slot for add-on expansion card – Install AP901 2-CH Dante Expansion Card to make the sound system compatible with Dante network or AP902 2-CH Mic/Line Pre-AMP Expansion Card to receive two more channels of mic / line input signals
- Fanless – Avoids fan noise and prevents dust from building up
- Supports VK Control System via RS-232 and Ethernet connection
- 1U rack mountable enclosure
- Firmware upgradable

Planning the Installation

Requirements

Prepare the following before installing the AP412 unit:

- 1 set of loudspeakers
- 1 or more audio source devices

Connecting Speakers

The AP412 supports both high-impedance (70V/100V) and low-impedance (4 Ω/8 Ω) speaker connections.

Before wiring the speakers, confirm the desired operation mode and ensure the following:

- The total load does not exceed the amplifier's rated output.
- The total impedance is not lower than the amplifier's minimum load.

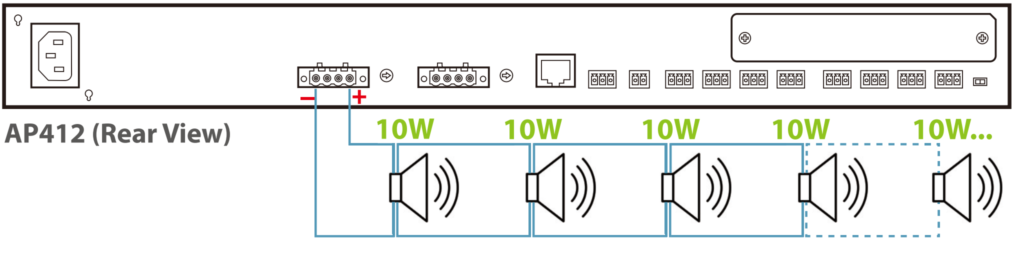
High-Impedance Connections (70V/100V)

In high-impedance installations, multiple speakers equipped with line transformers can be connected in parallel to a single amplifier output.

Each speaker's power tap determines its individual output level, while the amplifier delivers a constant voltage to all connected speakers. This configuration is commonly used in commercial or distributed sound systems where equal voltage but different wattage settings are required.

Always ensure that the sum of all speaker power taps does not exceed the amplifier's maximum rated power per channel.

70V / 100V High-Impedance Wiring Example



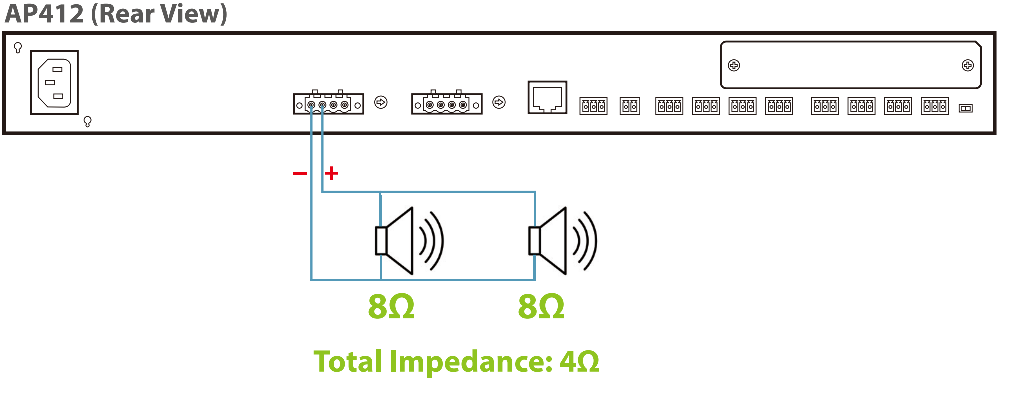
Low-Impedance Connections (4 Ω / 8 Ω)

When using low-impedance speakers, the total load impedance depends on how the speakers are connected.

Parallel Connection

The total impedance is calculated as the reciprocal of the sum of the reciprocals of each speaker's impedance. (For example, two 8 Ω speakers in parallel result in a total impedance of 4 Ω.)

Ensure that the total impedance does not fall below the amplifier's rated output impedance.



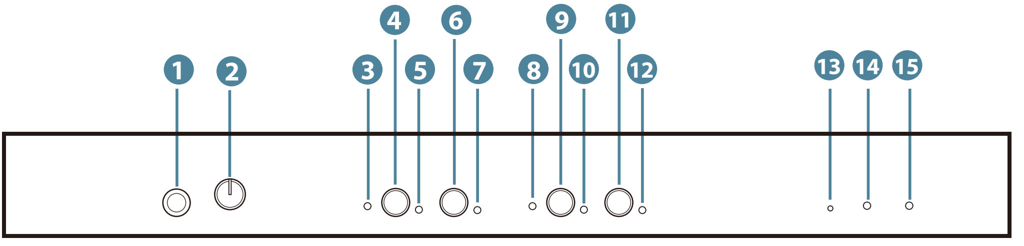
Series Connection

The total impedance equals the sum of each speaker's impedance. However, if one speaker fails or becomes disconnected, the signal to subsequent speakers will be interrupted. This configuration is less common and generally used only when parallel wiring is not possible.

Note: Before powering on the amplifier, verify that the total speaker load is within the rated output range. This includes checking load impedance and total speaker wattage, according to the selected output mode, to prevent damage.

Components

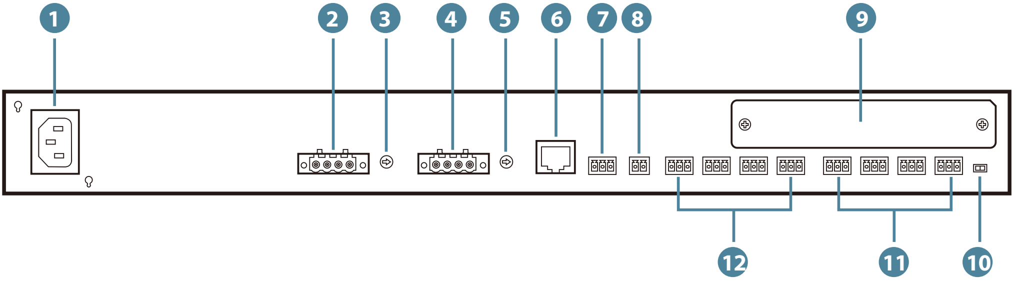
Front View



No.	Component	Description
1	headphone jack	Connects to a headphone with a 1.4" (6.35 mm) TRS connector.
2	volume control rotary potentiometer	Controls the headphone volume.
3	bridge LED for channel A & B	Lights to indicate that channel A and channel B are bridged. Refer to *LED Status*, page 10 for details.
4	volume control for channel A	Adjusts the audio output volume of channel A.
5	signal / clip LED for channel A	Lights to indicate the audio signal status of channel A. Refer to *LED Status*, page 10 for details.
6	volume control for channel B	Adjusts the audio output volume of channel B.
7	signal / clip LED for channel B	Lights to indicate the audio signal status of channel B. Refer to *LED Status*, page 10 for details.
8	bridge LED for channel C & D	Lights to indicate that channel C and channel D are bridged. Refer to *LED Status*, page 10 for details.
9	volume control for channel C	Adjusts the audio output volume of channel C.

No.	Component	Description
10	signal / clip LED for channel C	Lights to indicate the audio signal status of channel C. Refer to *LED Status*, page 10 for details.
11	volume control for channel D	Adjusts the audio output volume of channel D.
12	signal / clip LED for channel D	Lights to indicate the audio signal status of channel D. Refer to *LED Status*, page 10 for details.
13	reset button	Press and hold the button for more than 10 seconds to rest the unit.
14	standby LED	Lights to indicate the unit is in standby mode. Refer to *LED Status*, page 10 for details.
15	power LED	Lights to indicate the unit is powered on. Refer to *LED Status*, page 10 for details.

Rear View

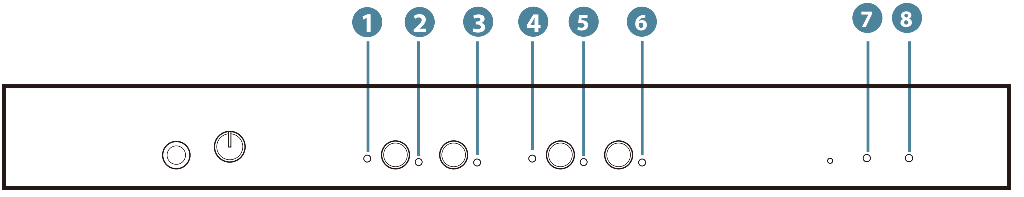


No.	Component	Description
1	power socket	Connects to the power cord.
2	speaker out for channel C & D (4-pin Euroblock connector)	Connects to a set of passive speakers such as AS104 / AS106 / AS108.
3	impedance / bridge switch for channel C & D	Adjusts the impedance for channel C & D between the following options: ◆ 4Ω ◆ 8Ω ◆ 8Ω Lo-Z bridge ◆ 70V ◆ 100V When switching to mute or blank option, channel C & D are muted.
4	speaker out for channel A & B (4-pin Euroblock connector)	Connects to a set of passive speakers such as AS104 / AS106 / AS108.

No.	Component	Description
5	impedance / bridge switch for channel A & B	Adjusts the impedance for channel A & B between the following options: ◆ 4Ω ◆ 8Ω ◆ 8Ω Lo-Z bridge ◆ 70V ◆ 100V When switching to mute or blank option, channel A & B are muted.
6	LAN port	Connects to a network switch or a PC.
7	RS-232 serial port	Connects to an ATEN Control Pad.
8	mute switch port (2-pin Euroblock connector)	Connects to a physical mute button, e.g. a footswitch.
9	expansion slot	Installs the expansion card, such as AP901 / AP902, into the expansion slot for flexible and expandable connectivity.
10	ground / lift switch	Switches to lift to eliminate hum noise from dirty AC electric power circuit.
11	audio mic/line input channels (3-pin Euroblock connectors)	Connects to your audio sources.
12	audio line output channels (3-pin Euroblock connector)	Connects to an active subwoofer or other audio equipment.

LED Status

You can find the unit's LEDs on the front panel as illustrated below. See the table below for details on the LED indication.



No.	LED	Indication	Description
1	bridge LED for channel A & B	lights amber	Channel A and channel B are bridged.
		off	Bridged mode of channel A and channel B is inactive.
2	signal / clip LED for channel A	lights green	The speaker output signal strength is over -50 dBFS.
		lights red	The speaker output signal strength reaches the hard-clipping limit (-3 dBFS).
		off	There is no audio signal.
3	signal / clip LED for channel B	lights green	The speaker output signal strength is over -50 dBFS.
		lights red	The speaker output signal strength reaches the hard-clipping limit (-3 dBFS).
		off	There is no audio signal.
4	bridge LED for channel C & D	lights amber	Channel C and channel D are bridged.
		off	Bridged mode of channel C and channel D is inactive.

No.	LED	Indication	Description
5	signal / clip LED for channel C	lights green	The speaker output signal strength is over -50 dBFS.
		lights red	The speaker output signal strength reaches the hard-clipping limit (-3 dBFS).
		off	There is no audio signal.
6	signal / clip LED for channel D	lights green	The speaker output signal strength is over -50 dBFS.
		lights red	The speaker output signal strength reaches the hard-clipping limit (-3 dBFS).
		off	There is no audio signal.
7	standby LED	lights amber	The unit is in standby mode.
		blinks amber	The unit is in overheat protection mode.
		off	The unit is active and operating normally.
8	power LED	lights green	The unit is powered on.
		blinks green	Firmware upgrade of the unit is in process.
		off	The unit is powered off.

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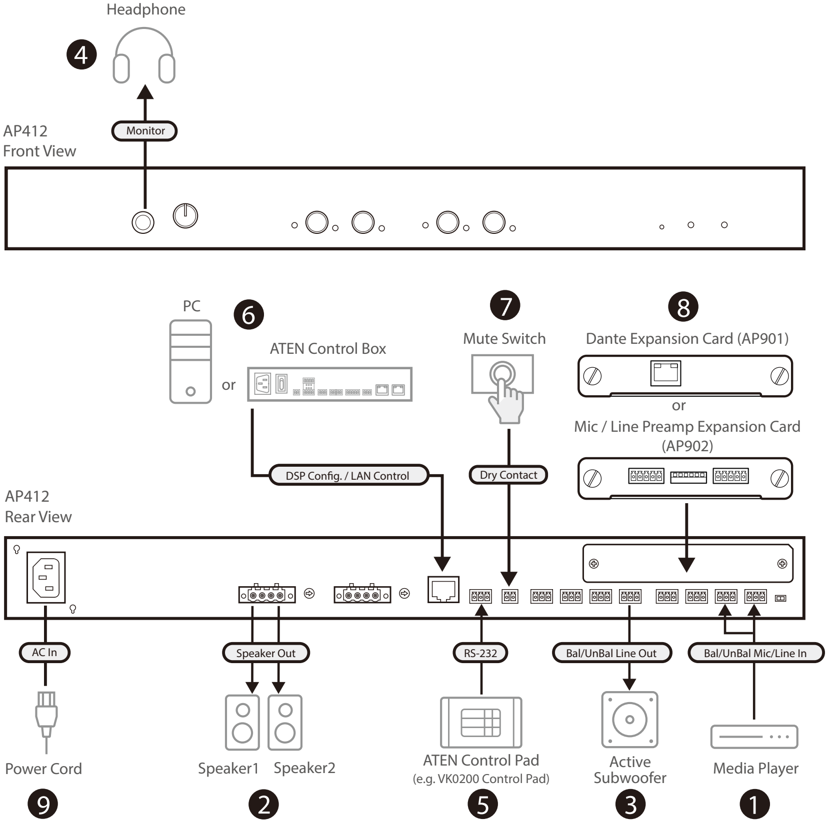
Chapter 2
Hardware Setup

1. Please review the safety information regarding the placement of this device in *Safety Instructions*, page 97.
2. Do not power on the AP412 until all the necessary hardware is connected.

Connecting the AP412 Unit

Follow the steps below to connect the AP412 to audio source devices, a set of loud speakers, and an ATEN controller.



Note: Make sure all the equipment you are connecting to the unit is turned off and disconnected from the power source.

1. Connect your audio sources to the audio input channels.
2. Connect the speakers to the unit's speaker output.

   Note: 1. To avoid damaging the amplifier, please connect the wires to your speakers first, and then connect the other end of the wire to the speaker output channels using the supplied 4-pin Euroblock connector with screw lock (5.08 mm).

   2. Keep the unit and its small parts such as 4-Pin Euroblock connector out of the reach of children. Children can choke or suffocate on the released small parts through detachment or breakage.

3. Connect the subwoofer or other audio equipment to the unit.
4. (Optional) Connect a headphone with a 1/4" (6.35 mm) TRS connector, and adjust the headphone volume using the rotary volume control next to the headphone jack.
5. (Optional) To control the unit using an ATEN Control Pad, connect the Control Pad to the unit's RS-232 port.
6. (Optional) Use the LAN port for firmware upgrades, DSP configuration, or remote management (using a hardware controller).
   - To control the unit using a hardware controller through the Ethernet, e.g. an ATEN Control Box, connect the LAN port to a network switch.
   - To upgrade firmware, configure DSP settings (via the web console), or manage the unit, connect the LAN port to a PC.

   Note: The default IP address is *192.168.0.60*. Use the default login credentials *administrator* and *password* upon first login.

7. (Optional) Connect a physical mute button (e.g., a footswitch) to the mute switch port to disable all audio outputs when the button is triggered.

8. (Optional) To connect the unit to the Dante network or additional mic/line level output device(s), install the Dante expansion card (AP901) or mic/line preamp expansion card (AP902) to the expansion slot in advanced.

Note: 1. AP901 and AP902 are sold separately.

2. Refer to *Installing / Removing the Expansion Card*, page 19 for how to install / remove the expansion card.

9. Connect the supplied power cord to the unit's power socket after powering on all other audio equipment. The unit's power LED lights green to indicate the unit is powered on.

Note: If the AP412 enters standby mode, feed an audio signal with a level higher than –24 dBu to wake the unit, or log in to the web GUI to activate it.

10. Adjust the volume from the unit's front panel.

Expansion Card

The expansion cards listed below are dedicated for use with ATEN's AP DSP Power Amplifier series:

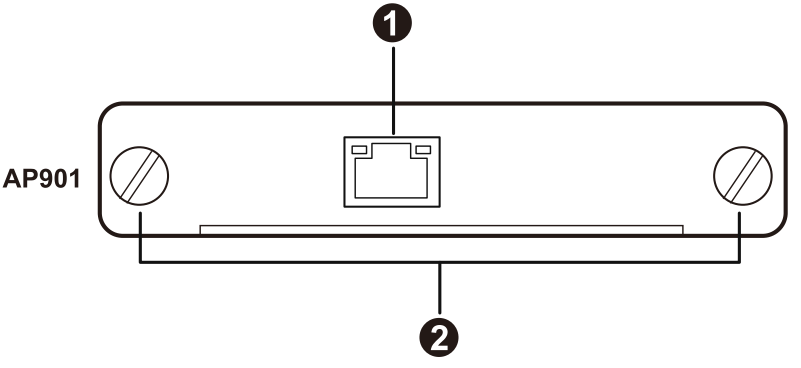
Note: The expansion cards are sold separately. Please contact your ATEN dealer or go to ATEN website for available accessories and product information.

- **AP901** 2-CH Dante Expansion Card for AP Series
  - Allows AP power amplifiers to receive 2 inputs of highquality, low-latency digital audio via Dante AoIP solution
  - Enables AP power amplifiers to easily integrate with Dante-enabled devices, such as digital mixers, processors, and media players
  - Supports 24-bit and sampling rate of 48kHz
  - Compatible with Dante Controller software to achieve streamlined workflow
  - Easy to be installed into the AP power amplifiers' expansion slot
  - Firmware upgradable via Dante LAN

- **AP902** 2-CH Mic/Line Pre-AMP Expansion Card for AP Series
  - Provides AP power amplifiers with dual-channel inputs for easier integration with balanced / unbalanced audio signals at mic / line level
  - Adjustable gain level – handles various strength levels of input sources
  - Preamp designed to lower the noise floor and improve the dynamic range for high-quality sound performance
  - Easy to be installed into the AP power amplifiers' expansion slot

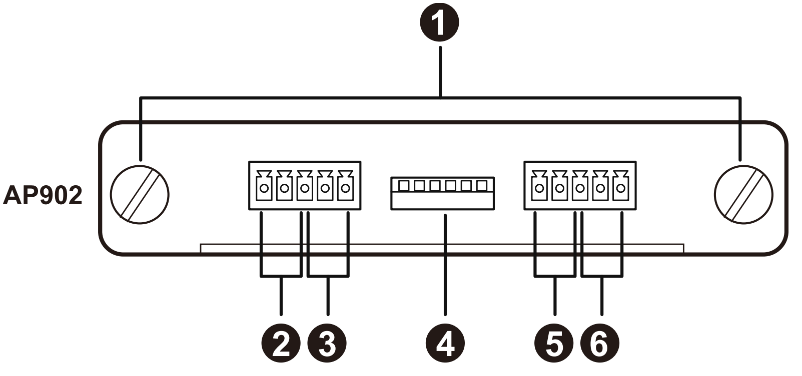
Hardware Overview

AP901



No.	Component	Description
1	Dante link port	Use an Ethernet cable to connect the Dante link port to a network switch. It is strongly recommended that you use a shielded Cat 5e cable (or higher) for better network connectivity.
2	screws	Fasten the expansion card to the slot of the AP DSP Power Amplifier using the screws. Tighten the screw by turning it clockwise while loosen the screw by turning it counterclockwise.

AP902



No.	Component	Description
1	screws	Fasten the expansion card to the slot of the AP DSP Power Amplifier using the screws. Tighten the screw by turning it clockwise while loosen the screw by turning it counterclockwise.
2	line input for channel 2	Connect the audio source devices to the input channel.
3	mic input for channel 2	Connect the audio source devices to the input channel.*
4	gain control switch	Switch on or off the pole(s) to adjust the audio gain level.
5	line input for channel 1	Connect the audio source devices to the input channel.
6	mic input for channel 1	Connect the audio source devices to the input channel.*

Note: 1. To prevent device damage, do not feed a line level signal to mic input channel.

2. Do not connect Mic and Line in the same channel at the same time.

Installing / Removing the Expansion Card

Make sure that you turn off the AP412 power amplifier and disconnect the unit from the power source before installing or removing the expansion card.

Installing

To install the expansion card into the AP412 power amplifier:

1. Remove the expansion slot plate on the rear side of the AP412 power amplifier.

2. Insert the expansion card into the amplifier's expansion slot.

   Note: Give a little push to get the expansion card fully seated in the slot.

3. Apply force to each screw head till it is in place, and then tighten the screw by turning it clockwise.

Removing

Note: For safety purpose, please power off the AP412 power amplifier and then wait for 50 seconds before removing the expansion card.

To remove the expansion card:

1. Alternately loosen each screw.

2. Hold the two screws and then gently pull out the expansion card.

3. Use the expansion slot plate to cover the blank slot, and secure it with the screws.

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Chapter 3
Operation

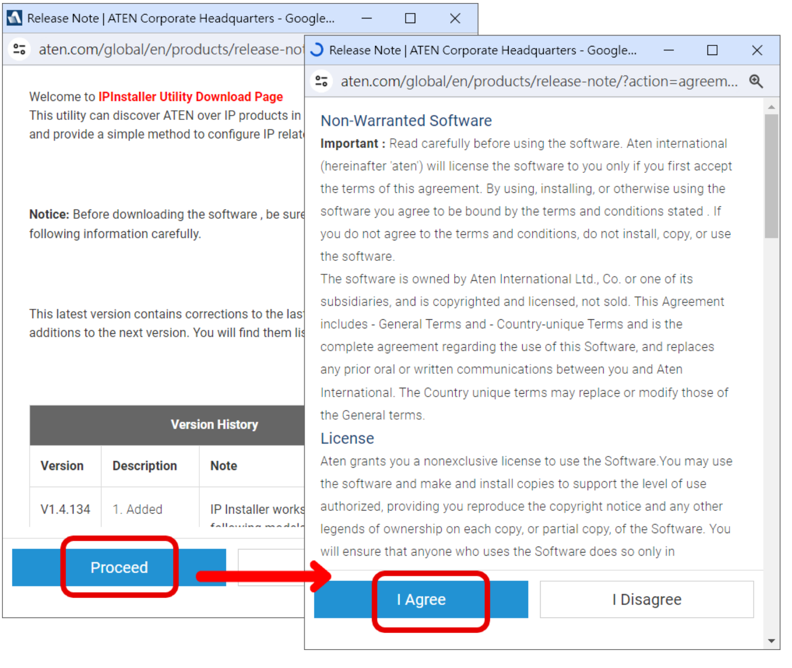
Browser Web Control

The AP412 power amplifier offers the web control and configuration through the Ethernet connection. The default IP address of the unit's web console is *192.168.0.60* without DHCP environment. If you connect the AP412 unit to a router, use the IP address that DHCP server allocates.

DHCP-assigned IP Address

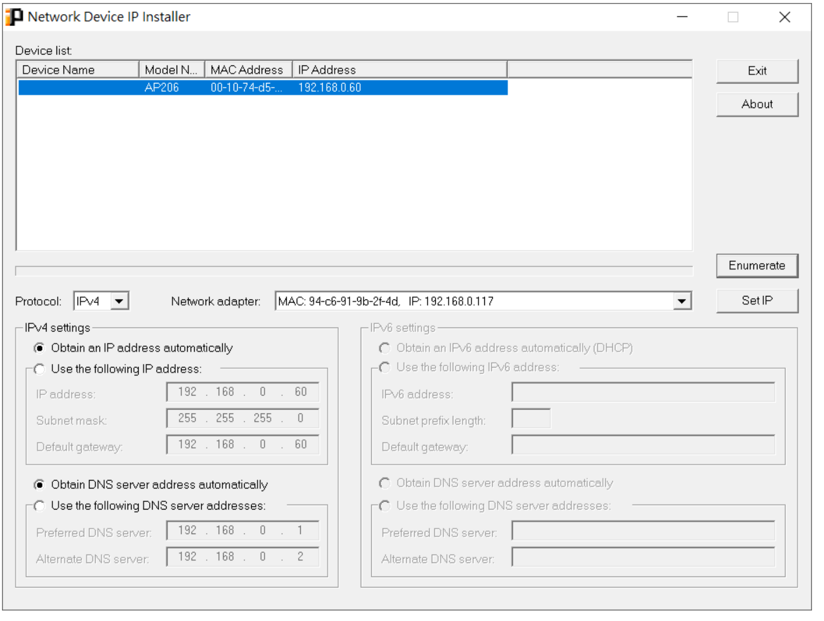
To get the DHCP-assign IP address, do the followings:

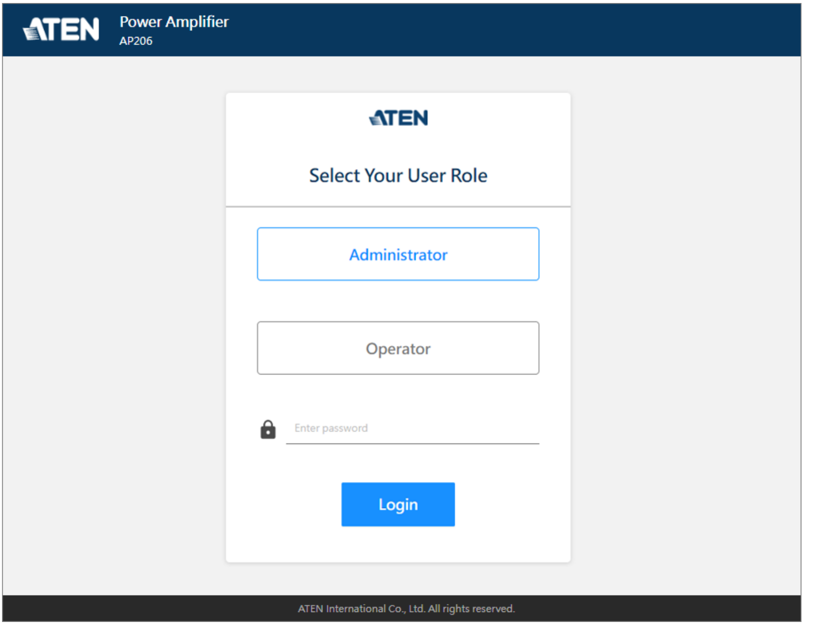
1. Download the **IP Installer** from the *Support and Download* tab of the product page.



2. Unzip the .zip file of the IP Installer and then run the .exe file.

3. Obtain the DHCP-assigned IP address of the unit from the **Device List**, and use this IP address to access the unit's web GUI.



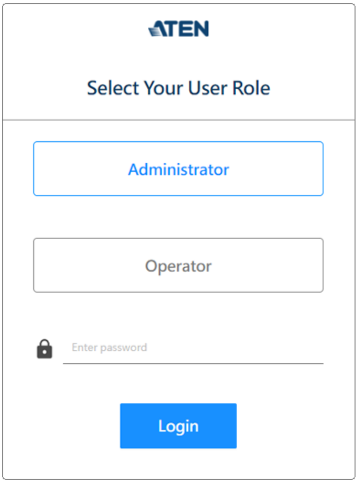


Note: All functions available in the web GUI can also be configured through the ATEN **Audio Wizard** application.
ATEN Audio Wizard, downloadable from the product page on the ATEN website, provides the same operational capabilities and parameter controls as the web GUI.

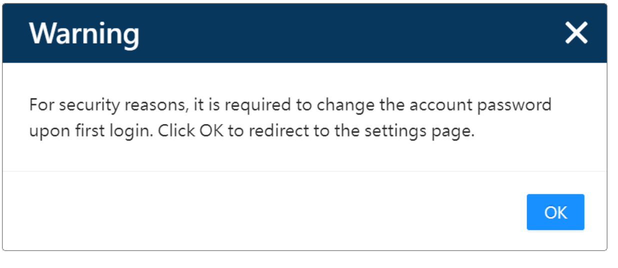
Login

To control the unit from a web control:

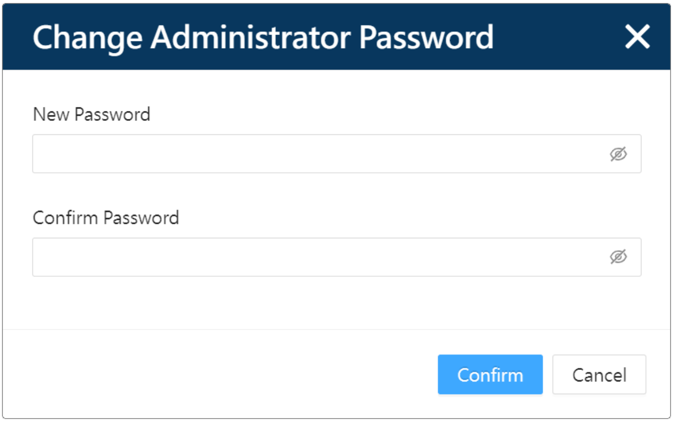
1. Start up the supported web browser, and then input the unit's IP address into the address bar.
2. The login page shows up. Upon first login, select **Administrator** as the user role type, and use the default password, *password*.



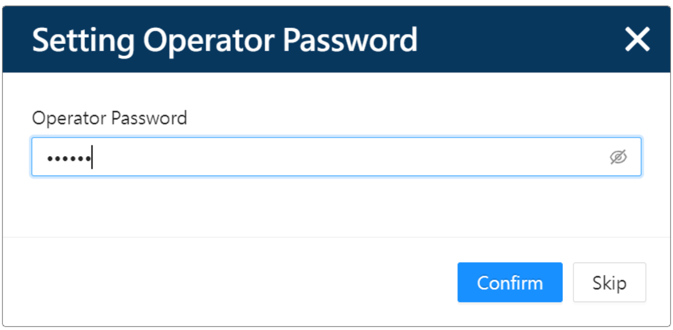
3. It requires to change the password for **Administrator**. Follow the on-screen instructions to complete.

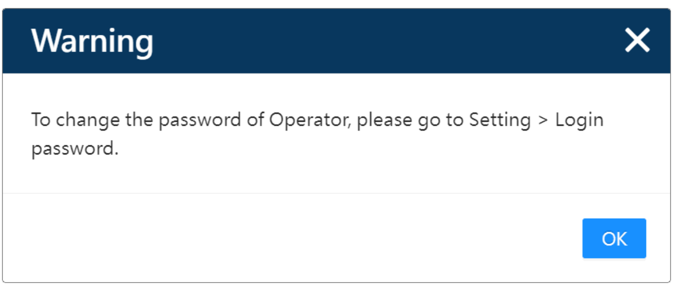


a) Enter the new password for **Administrator**, and confirm by entering the password again. Click on the Confirm button to continue.



b) Change the password for **Operator**. You can skip this steps and change the Operator's password later on.

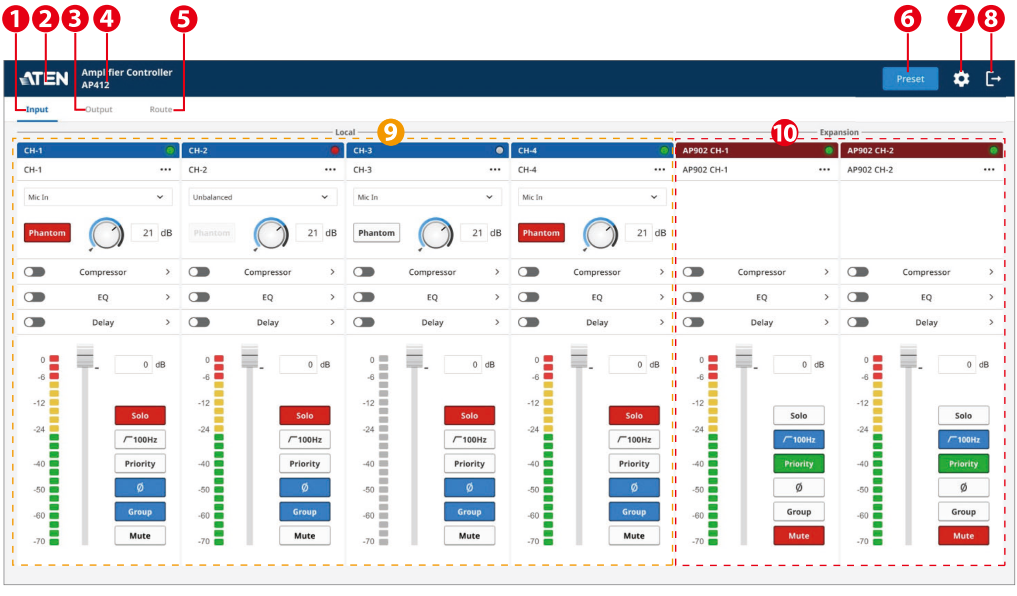




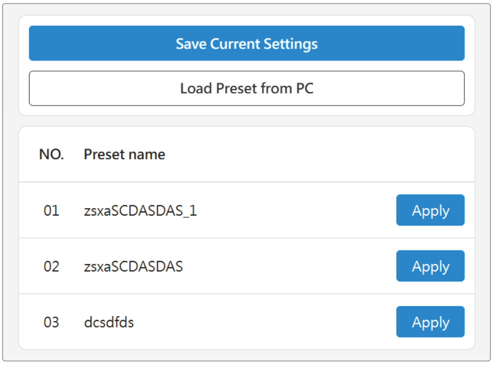
c) After successfully changed the password, you will be redirected to the login page again. Use the new password to log in to the unit's web console.

DSP Configuration

Digital Signal Processing (DSP) performs sound manipulation with filtering, equalizing, limiting, and enhancing the audio signal. After logging in or waking up the unit from the standby mode, you will enter the DSP configuration screen as illustrated below:



No.	Item	Description
1	input tab	Listing the input channels and the configuration panel of each channel for user to configure the processing to be applied to the input audio signal. See *Input Tab*, page 27.
2	ATEN	Click on the ATEN logo that takes you straight to ATEN's official website.
3	output tab	Listing the output channels and the configuration panel of each channel for user to configure the processing to be applied to the output audio signal. See *Output Tab*, page 42.
4	model name	The model name of the connected unit
5	route tab	Configuring the signal patching matrix. See Route Tab, page 53.

No.	Item	Description
6	preset configuration	A button for display preset configuration menu which contains the following:  ◆ **Save Current Settings**: A button for saving the current configurations to be a preset ◆ **Load Preset from PC**: A button for importing the present file (.bin format) that saved in the PC ◆ **preset menu**: A list of all the saved preset(s) for user to choose and apply See *Preset Management*, page 63 for details.
7	settings button	A button for switching to the system settings screen. See *System Settings*, page 66. **Note:** This function button is only available for **Administrator**.
8	exit button	A button for logging out the web GUI
9	local channel operation area	Showing the audio signal configuration panel for the local channels.
10	expansion channel operation area	Showing the audio signal configuration panel for the expansion channels (available only when an expansion card is installed).

Input Tab

The Input tab allows users to monitor and configure audio input signals before they are processed by the patching matrix and output stages. From this tab, users can adjust input gain levels, apply input-side DSP functions, and manage signal behavior to ensure clean and stable signal flow into the system.

The Input tab is divided into two sections based on input source types:

- **Local** input channels:

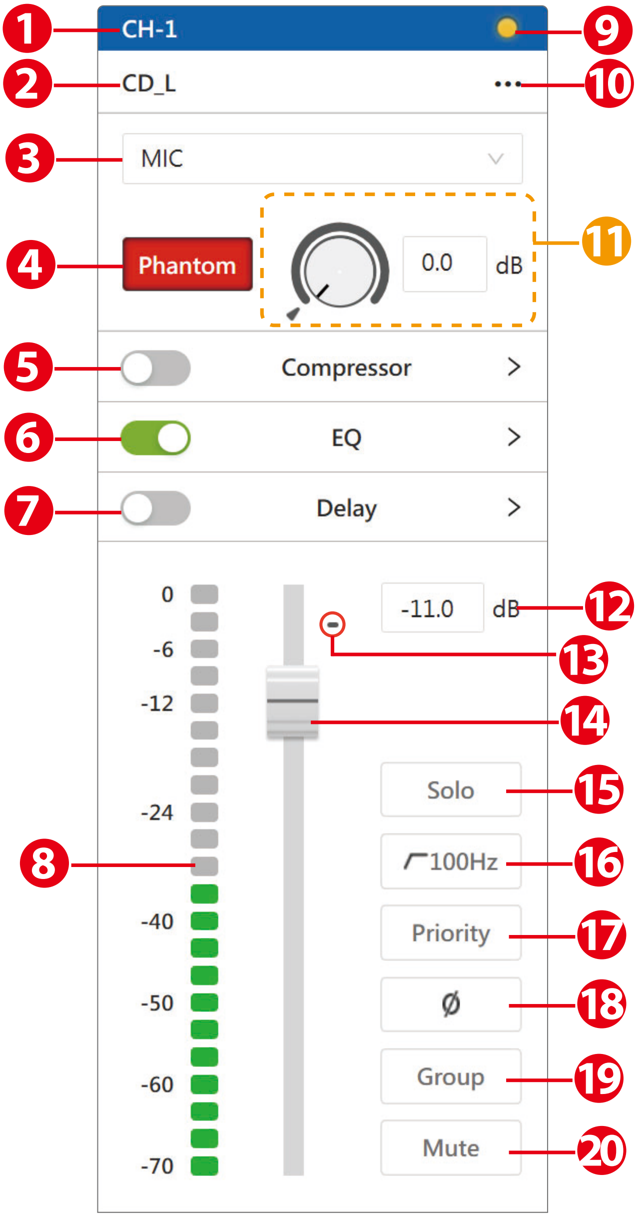
  For configuring audio sources connected directly to the unit's physical input ports, such as microphones or line-level devices.

- **Expansion** input channels:

  For configuring input channels provided by installed expansion cards, enabling additional audio inputs to be integrated into the system.

Each input channel provides real-time signal monitoring and independent DSP control, allowing audio signals to be conditioned at the source before patching and output-side processing.

In the **Input** tab, users can use the following DSP functions to configure the audio signal for each channel:



No.	Item	Description	Default
1	channel ID	Displays the ID of this channel. Please note that the ID is unalterable.	N/A
2	channel name	Displays the name for this channel. To change the channel name, click the more button and select **Rename** to continue.	as channel ID
3	input signal mode	Select the input mode according to the connected audio source. ◆ **MIC**: For microphone-level input sources ◆ **Balanced** / **Unbalanced**: For line-level input sources using balanced or unbalanced connections The available gain range varies depending on the selected input mode: ◆ **MIC**: 0 dB to +36 dB ◆ **Unbalanced**: -6 dB to +10 dB ◆ **Balanced**: -6 dB to +4 dB Every time the input signal mode is changed, the gain value is automatically set to the minimum.	Balanced
4	Phantom	Phantom power is available only when the input signal mode is set to **MIC**. When enabled, phantom power supplies operating voltage for condenser microphones and does not affect audio quality.	Disabled / Off
5	compressor switch	Switch on or off to apply or withdraw the compressor settings to this channel. Click on the function name to open the popup for further configuration. See Compressor Configuration, page	Off
6	equalizer switch	Switch on or off to apply or withdraw the EQ settings to the audio signal. Click on the function name to open the popup for further configuration. See *Equalizer Configuration*, page 38.	Off

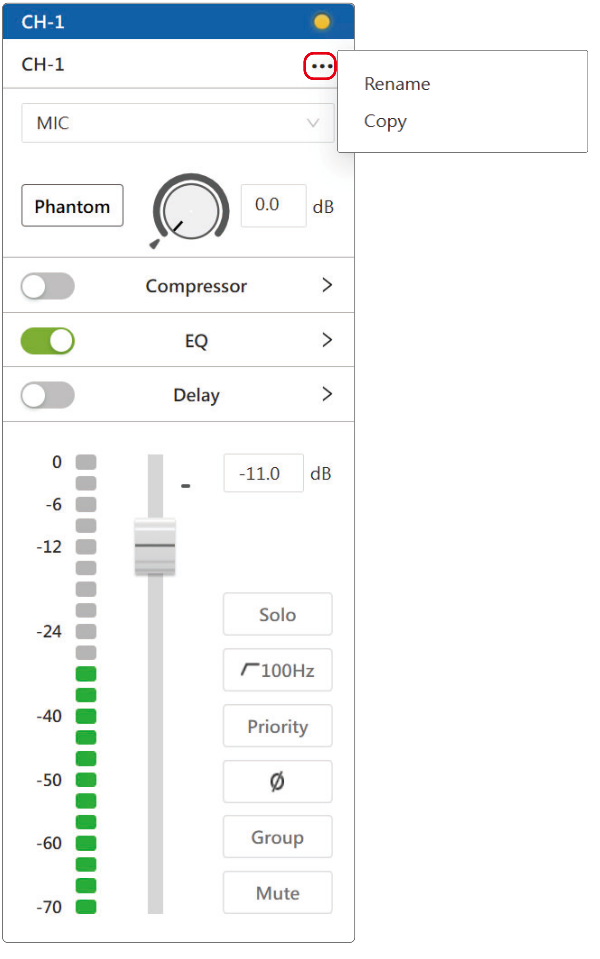
No.	Item	Description	Default
7	delay switch	Switch on or off to apply or withdraw the delay settings to the audio signal. Click on the function name to open the popup for further configuration. See *Delay Configuration*, page 41.	Off
8	signal level meter	Shows the audio signal level in decibels (dB) after preamp and DSP processing. This meter reflects the signal level after the input signal has passed through the preamp stage (input signal mode, phantom power, and gain control), and the downstream DSP processing, including Compressor, EQ, Delay, HPF, Priority, Phase, Group, and Mute settings.	N/A
9	input signal level	Indicates the input signal level in real time after preamp processing only. This indicator displays the signal level after the signal passes through the preamp stage (input signal mode, phantom power, and gain control), before any DSP processing is applied. The indicator colors represent the input signal strength as follows: ◆  ◆ **Local**: no signal; <-50 dBu ◆ **Expansion**: no signal; <-70 dBFS ◆  ◆ **Local**: >-5 dBu and ≥-50 dBu ◆ **Expansion**: <-12 dBFS and ≥-70 dBFS ◆  ◆ **Local**: >-5 dBu to +13 dBu ◆ **Expansion**: -12 dBFS to -3 dBFS ◆  ◆ **Local**: ◆ Peak level: > +13 dBu ◆ Maximum peak level: +19 dBu ◆ **Expansion**: peak; >-3 dBFS	N/A

No.	Item	Description	Default
10	more button	Click on the more button to open the option menu for further settings: ◆ **Rename**: Input a new name for this channel. ◆ **Copy**: Copy the settings of this channel and apply them to the selected channel(s).	N/A
11	gain control	The gain control panel contains the following: ◆ Gain Control Knob: Adjusts the channel gain level by dragging the knob. ◆ Gain Level: Displays the current gain value. Users can also enter a value manually. ◆ Gain Reference Marker: Indicates the 0 dB reference point of the gain control knob. The marker position varies depending on the selected input signal mode. **Note:** The 0 dB reference point does not necessarily correspond to the minimum position of the gain control knob and varies according to the selected input signal mode.	0.0 dB
12	channel fader level	Shows the gain or attenuation level of the audio signal after processing.	0.0 dB
13	Fader Zero Reference Marker	Indicates the 0 dB reference position of the channel fader. The solid marker represents the reference point where the fader value is 0 dB.	N/A
14	channel fader	Adjust the gain or attenuation level of the audio signal to be output. Drag the fader to change the value in decibel. The volume value also displays in the channel fader level field next to the channel fader.	0.0 dB

No.	Item	Description	Default
15	solo	Allows users to monitor this channel through headphones connected to the AP412 front panel. When **Solo** is enabled on any channel, the signal bypasses the channel fader, mute, phase, and priority settings, and is monitored only through the Compressor, EQ, Delay, and HPF processing stages. As a result, the user can still hear the original input signal even if the channel fader is set to minimum or the channel is muted.	Disabled
16	high-pass filter	Enable this function to remove the signal below the cut-off frequency of 100 Hz, -18 dB/OCT.	Disabled
17	priority button	Click the button to set the audio signal input to this channel to have the priority. Other input channels will be attenuated/suppressed by 20 dB.	Disabled
18	phase button	Click the button to invert the polarity of the phase. Disabling this function means that the phase polarity is normal.	Disabled
19	fader group button	Enable the group function to add this channel to the linked channels to simultaneously control the volume levels.	Disabled
20	mute button	Click on the mute button to enable or disable the mute function for this channel.	Disabled

Renaming & Copying a Channel

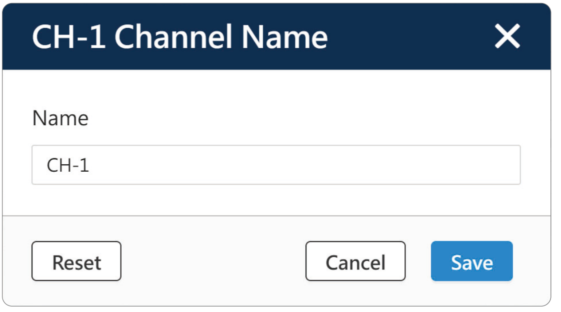
Click the more button next to the channel name to open the option menu. From the menu, you can rename the channel or copy its settings to other channels.



Rename

To change the channel name:

1. Click **Rename** to open the **Channel Name** dialog box.



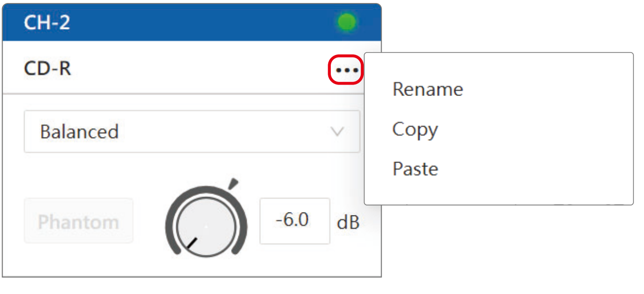
2. Enter a new channel name.
3. Click **Save** to apply the change.
    - **Reset**: Reset the channel name to the factory default.
    - **Cancel**: Discard the change.
    - **Save**: Save the new channel name. The change takes effect immediately.

Copy

The **Copy** function allows you to copy the settings of one channel and apply them to other channel(s).

To copy and apply channel settings:

1. Open the option menu of the channel you'd like to copy, and click **Copy**.
2. Open the option menu of the target channel. A **Paste** option will appear.



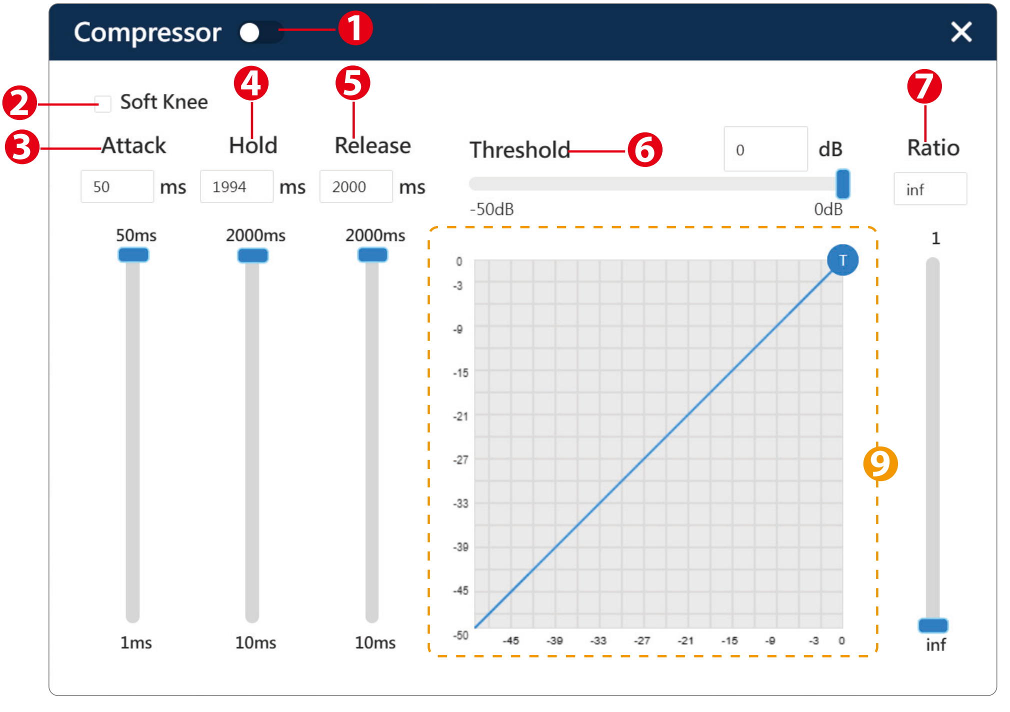
3. Click **Paste** to apply the copied channel settings.

Compressor Configuration

The compressor controls the dynamic range of the input audio signal by applying level reduction when the signal level exceeds a defined threshold.

Compression can be enabled for the selected input channel, and the parameters that determine how the signal is processed before patching and output-side processing can be adjusted.

The following diagram and table describe the available compressor controls and their default settings.



No.	Item	Description	Default
1	compressor switch	Switch it on or off to apply or withdraw the compressor settings to this channel.	Off

No.	Item	Description	Default
2	soft knee setting	Uses soft knee processing to smooth the compressor effect. The compressor gives a linear or non-linear response curve according to the knee settings that controls how the compressor effects on an audio signal. ◆ **Hard Knee**: A hard knee response is applied when the **Soft Knee** option is disabled. Compression starts immediately once the signal exceeds the threshold, resulting in a more abrupt and clearly defined level reduction. This setting is suitable when precise dynamic control or a more aggressive compression effect is required. ◆ **Soft Knee**: Check the checkbox to enable a soft knee setting. Level reduction starts earlier, before the signal reaches the threshold. This results in smoother and more natural level control.	Off
3	attack setting	Defines how quickly the compressor responds once the signal exceeds the threshold. Specify the value in milliseconds by entering a value or dragging the slider.	1 ms
4	hold setting	Sets the time for which the existing level reduction is retained after the input signal falls below the threshold, delaying the start of the release phase.	10 ms
5	release setting	Defines how quickly the applied level reduction is released after the hold period ends, allowing the signal level to return gradually toward its uncompressed state.	10 ms

No.	Item	Description	Default
6	threshold setting	The compression is applied once the audio signal crosses or approaches the threshold, and it stops at the full ratio value/amount. Specify a decibel value by manually inputting the value in the field or dragging the slider. **Note:** A hard knee setting starts the compression process right after the signal reaches the threshold while a soft knee setting applies the compression as the audio signal approaches the threshold.	0 dBFS
7	ratio setting	Set the ratio amount that dictates the reduced output audio signal. Specify a value by manually inputting the value in the field or dragging the slider.	1.0
8	compression graph	Display the graph that illustrates the compression curve according to your settings about threshold, ratio, and knee. The graph displays the input level on the x-axis and the output level on the y-axis, both in decibels (dB). The T in the curve graph marks out the threshold, and the R is the ratio. You can drag the T or the R in the curve graph to change the value of the threshold and ratio.	N/A

Equalizer Configuration

The equalizer (EQ) allows adjustment of specific frequency bands of the input audio signal by controlling frequency, gain, and Q. By boosting or attenuating selected bands, EQ is used to correct tonal balance, emphasize or suppress certain frequency ranges, and adapt the input signal to different source characteristics.

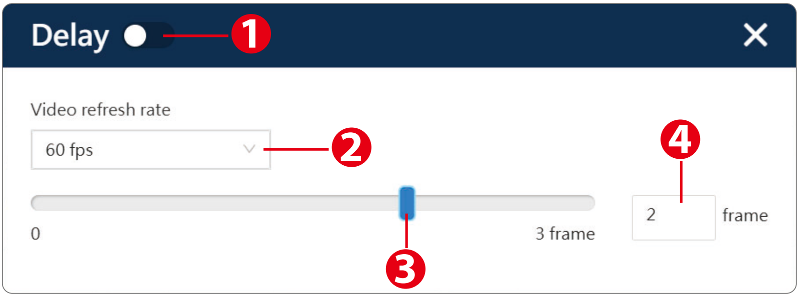
Each EQ band can be adjusted independently. Changes take effect immediately on the input signal and are reflected in the EQ graph, providing direct visual feedback for frequency response shaping.

No.	Item	Description	Default
1	EQ switch	Enables or disables the equalizer to boost or suppress the audio frequency bands.	Off
2	option menu	Select the EQ mode from the drop-down menu to apply the EQ settings. **Note:** EQ parameters are editable only when **Custom** mode is selected.	Custom
3	reset button	Restores all EQ parameters to their default values.	N/A
4	Q setting	Defines the bandwidth of the selected frequency band by adjusting the Q (quality factor). The Q factor represents the ratio of the center frequency to the bandwidth. A higher Q value results in a narrower bandwidth, while a lower Q value produces a wider bandwidth. Adjust the Q value by dragging the slider or entering a value manually.	1
5	gain setting	Adjusts the gain applied to the selected frequency band. ◆ Positive values boost the target frequency. ◆ Negative values attenuate the target frequency. Set the gain by dragging the slider or entering a value manually.	0 dB
6	frequency setting	Specifies the frequency to be adjusted within the range of 20 Hz to 20 kHz. Adjust the frequency by dragging the slider or entering a value directly in the frequency field.	200 Hz

No.	Item	Description	Default
7	EQ graph	Displays a visual representation of the EQ settings. ◆ Click **HPF** in the graph and drag to adjust the high-pass filter frequency. ◆ For bands 1–4, click a band node to select it, then drag to adjust its frequency, gain, and Q.	◆ Band 1: 200 Hz ◆ Band 2: 1 kHz ◆ Band 3: 5 kHz ◆ Band 4: 10 kHz

Delay Configuration

The function **Delay** for input audio signal is designed for synchronization of video and audio, so called "Lip Sync.". In most case, audio processing is faster than video, that is why audio should be delayed.



No.	Item	Description	Default
1	delay switch	Switch it on to enable the equalizer to boost or suppress the audio frequency bands.	Off
2	video refresh rate	Click to select the video refresh rate of the video.	60 fps
3	slider	Drag the slider to configure the frame delay.	0 (frame)
4	frame offset setting	Enter the number of frames that offset.	0 (frame)

Output Tab

The **Output** tab allows users to monitor and configure audio signals after they have been processed by the input stage and patching matrix. From this tab, users can adjust output-level parameters and apply output-side DSP functions to ensure the audio signal is properly delivered to connected speakers or downstream audio equipment.

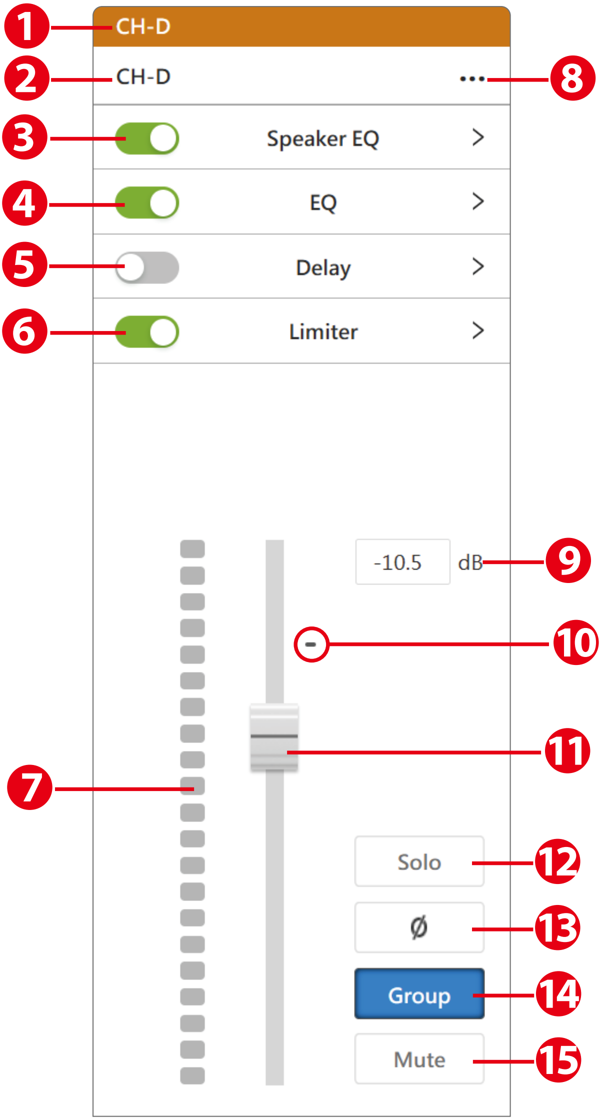
The **Output** tab is divided into two sections based on output signal types:

- **Speaker Out** channels:
  for configuring amplified speaker outputs
- **Line Out** channels:
  for configuring line-level outputs to external audio devices

Each output channel provides real-time signal monitoring and independent control to support flexible system tuning and output management.

Speaker Out Processing

The **Speaker Out** section provides controls for managing speaker-level audio signals after patching and input-side processing. Output DSP functions and output levels can be adjusted before the signals are delivered to the speakers connected to the unit's speaker output channels.

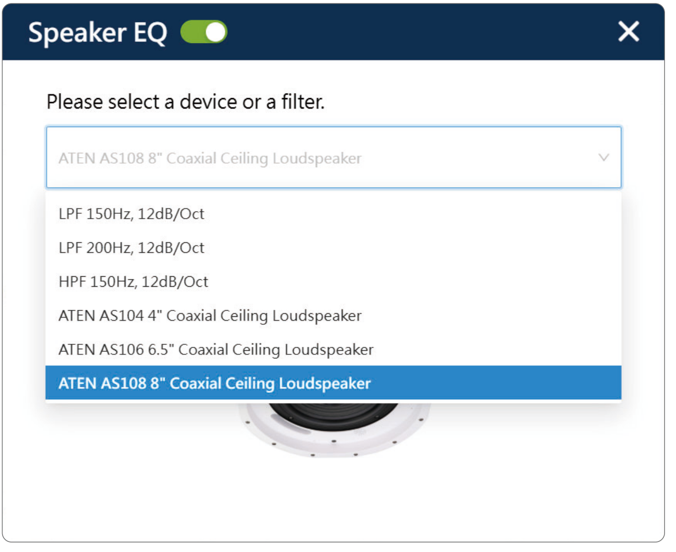


No.	Item	Description	Default
1	channel ID	Displays the ID of this channel. Please note that the ID is unalterable.	N/A
2	channel name	Displays the name for this channel. To change the channel name, click the more button and select Rename to continue.	as channel ID
3	speaker EQ	Select the speaker that connects to the speaker out channel.	Off
4	equalizer switch	Enable or disable the equalizer (EQ) for the speaker output signal. Click the function name to open the EQ configuration window. See *Equalizer Configuration*, page 47.	Off
5	delay switch	Enable or disable delay processing for the speaker output signal. Click the function name to open the delay configuration window. See *Delay Configuration*, page 50.	Off
6	limiter switch	Enable or disable the limiter to apply output level protection for the speaker output signal. Click the function name to open the limiter configuration window. See *Limiter Configuration*, page 52.	Off
7	signal level meter	Displays the real-time speaker output signal level in decibels (dB).	N/A
8	more button	Click on the more button to open the option menu for further settings: ◆ **Rename**: Input a new name for this channel. ◆ **Copy**: Copy the settings of this channel and apply them to the selected channel(s). See *Renaming & Copying a Channel*, page 33.	N/A
9	channel fader level	Displays the current gain or attenuation value of the speaker output signal in decibels (dB).	-50 dB

No.	Item	Description	Default
10	Fader Zero Reference Marker	Indicates the 0 dB (unity gain) reference position of the channel fader. The solid marker represents the reference point where the fader value is 0 dB.	N/A
11	channel fader	Adjusts the gain or attenuation level of the speaker output signal. Drag the fader to change the level in decibels (dB). The corresponding value is shown in the channel fader level field.	-50 dB
12	solo	Allows monitoring of this channel through headphones connected to the AP412 front panel. When **Solo** is enabled, the monitored signal includes all output-stage DSP processing, such as EQ, Delay, and Limiter, and is also affected by the channel fader, mute, phase, and priority settings. If the channel is muted, no audio from this channel will be heard, even when **Solo** is enabled.	Disabled
13	phase button	Inverts the signal polarity for this channel. When disabled, the signal polarity remains normal.	Disabled
14	fader group button	Enables group control by linking this channel with other grouped channels, allowing their volume levels to be adjusted simultaneously.	Disabled
15	mute button	Mutes or unmutes the audio output signal of this channel. If a physical mute switch is installed and activated, all audio output channels are muted. In this state, the channels cannot be unmuted through the AP412 web GUI.	Disabled

Speaker EQ

On the popup window, you can do the following:



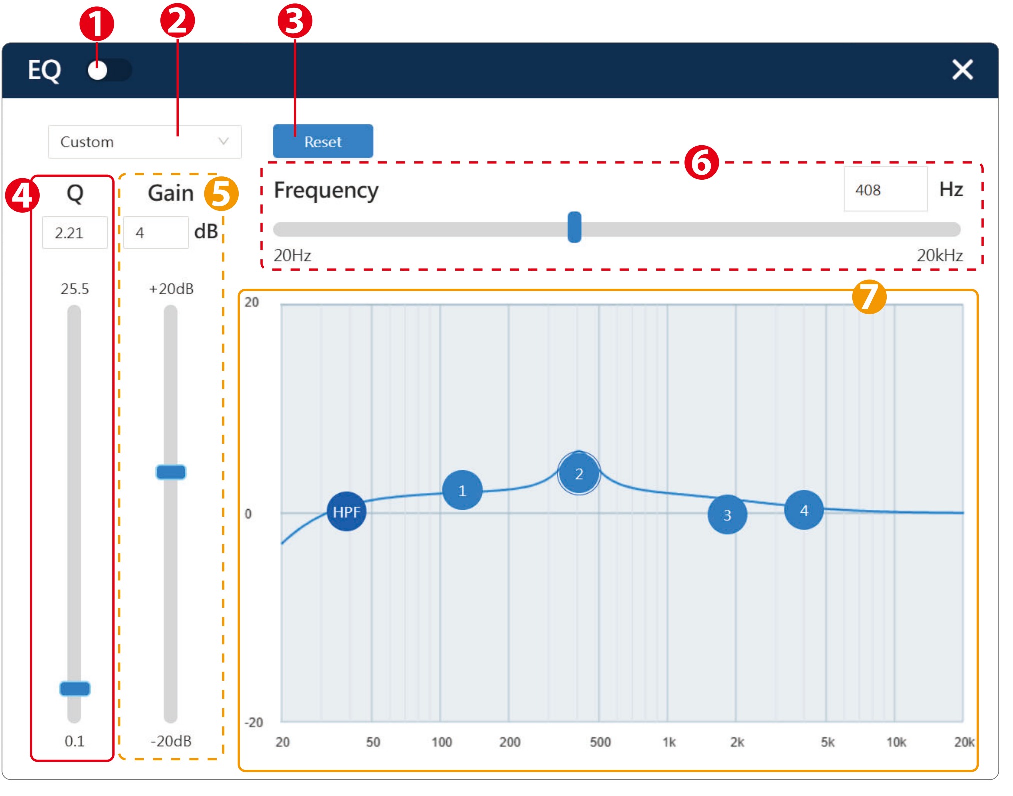
- Turn on the switch to enable the optimization of the audio signal to be output to the connected speaker.
- Select the filter to apply:
  - If this speaker output channel connects with an ATEN loudspeaker, directly select the model. The system will automatically apply the optimization.
  - If you use a self-supplied audio output device, select a filter that is suitable for your output device to apply.

Filter	Description
LPF 150 Hz, 12 dB/Oct	LPF are often used with subwoofers. Choose between 150 Hz and 200 Hz based on your connected subwoofer, and apply the low pass filter to filter out the high-frequency sounds from an audio signal.
LPF 200 Hz, 12 dB/Oct	
HPF 150 Hz, 12 dB/Oct	Apply the high pass filter to filter out the lowfrequency content of the signal to increase the clarity and definition.

Equalizer Configuration

The **EQ** (equalizer) is used to adjust the output frequency balance for the selected speaker channel. Settings are applied by enabling EQ, selecting a mode, and adjusting frequency bands directly in the EQ graph or control fields.

Detailed parameter definitions and default values are shown in the figure and table below.



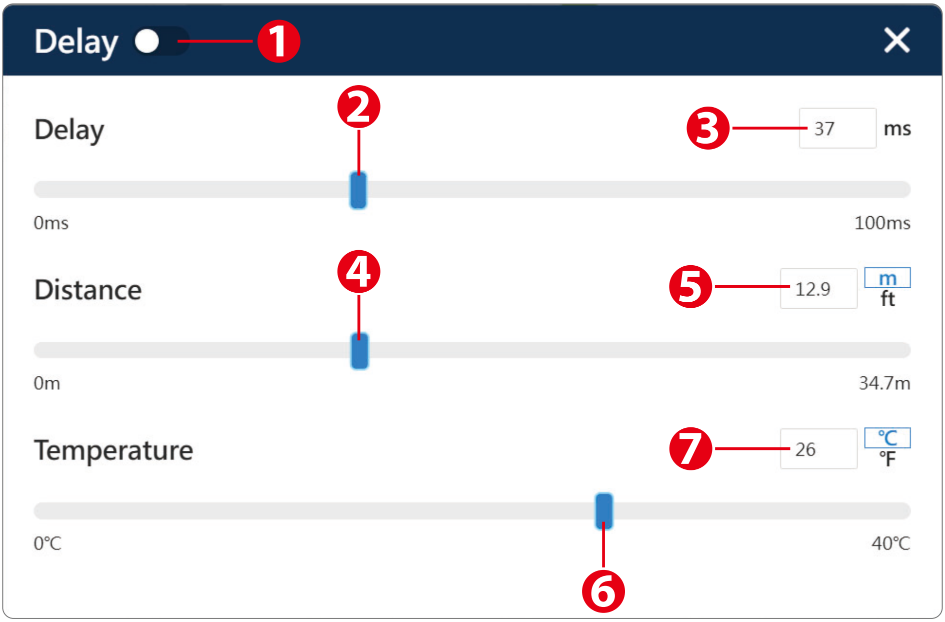
No.	Item	Description	Default
1	EQ switch	Enable or disable the equalizer for the selected channel. When enabled, the EQ can be used to boost or attenuate specific frequency bands.	Off
2	option menu	Select an EQ mode from the drop-down list. **Note:** EQ parameters are editable only when **Custom** mode is selected.	Custom
3	reset button	Restore all EQ parameters to their default values.	N/A
4	Q setting	Adjust the bandwidth of the selected frequency band by setting the Q (quality factor). The Q factor represents the ratio between the center frequency and its bandwidth: ◆ A higher Q value results in a narrower affected frequency range. ◆ A lower Q value results in a wider affected frequency range. Set the Q value using the slider or by entering a numeric value.	1
5	gain setting	Adjust the gain applied to the selected frequency band. ◆ Positive values boost the selected frequency range. ◆ Negative values attenuate the selected frequency range. Set the gain using the slider or by entering a numeric value.	0 dB
6	frequency setting	Specify the frequency to be adjusted within the range of 20 Hz to 20 kHz. The frequency can be set by moving the slider or by entering a value directly.	200 Hz

No.	Item	Description	Default
7	EQ graph	Displays the overall EQ curve based on the current settings. ◆ Click and drag the HPF node to set the cutoff frequency. ◆ Click a band node (Band 1–4) to select it, then drag the node to adjust its frequency and gain. ◆ The Q value of the selected band is reflected in the width of the curve.	◆ Band 1: 200 Hz ◆ Band 2: 1 kHz ◆ Band 3: 5 kHz ◆ Band 4: 10 kHz

Delay Configuration

The Delay function provides time alignment control for each speaker output channel.

Delay can be specified directly in milliseconds, or calculated automatically based on the configured speaker distance and ambient temperature. These settings allow per-channel adjustment of output timing during system configuration.



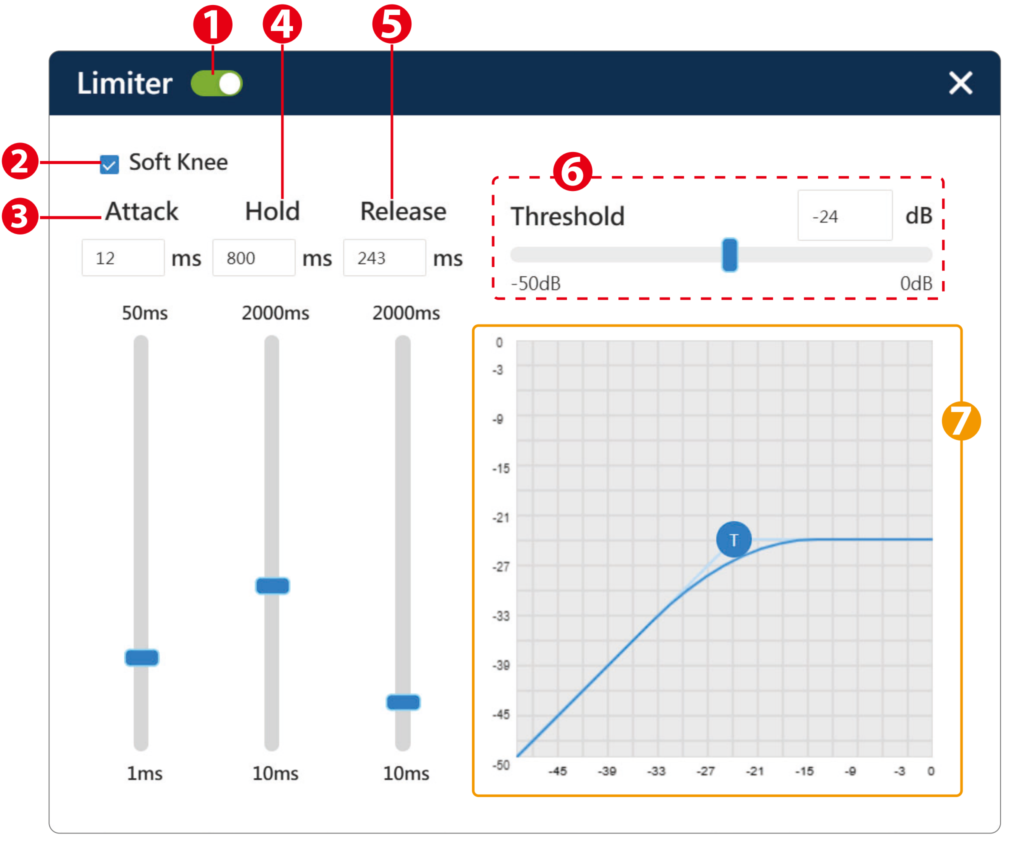
No.	Item	Description	Default
1	delay settings switch	Enable or disable the delay function for the current channel.	Off
2	delay time slider	Adjust the delay time in milliseconds by dragging the slider. The current delay time is displayed in the delay time value field.	0 ms

No.	Item	Description	Default
3	delay time value	Displays the delay time in milliseconds. You can manually enter a value in this field to set the delay time precisely.	0 ms
4	distance slider	Set the speaker distance by dragging the slider. The corresponding distance value is displayed in the distance value field.	0.0 m
5	distance value	Displays the distance between the speaker and the listening position. To modify the value: a) Enter a numeric value directly in the field. b) Select the unit of length between meters (m) and feet (ft).	0.0 m
6	temperature slider	Set the ambient temperature by dragging the slider. The corresponding temperature value is displayed in the temperature value field.	20 °C
7	temperature value	Displays the ambient temperature used for delay calculation. To modify the value: a) Enter a numeric value directly in the field. b) Select the temperature scale between Celsius (°C) and Fahrenheit (°F).	20 °C

Limiter Configuration

The limiter defines an upper output level for the speaker output channel. When enabled, the limiter constrains signal peaks based on the configured threshold and response parameters, preventing the output level from exceeding the specified limit.

Limiter behavior is adjusted through knee, attack, hold, release, and threshold settings. These parameters determine how quickly and how smoothly level reduction is applied as the signal approaches or exceeds the limit. Together, they allow precise control over peak handling while maintaining stable and predictable speaker output .

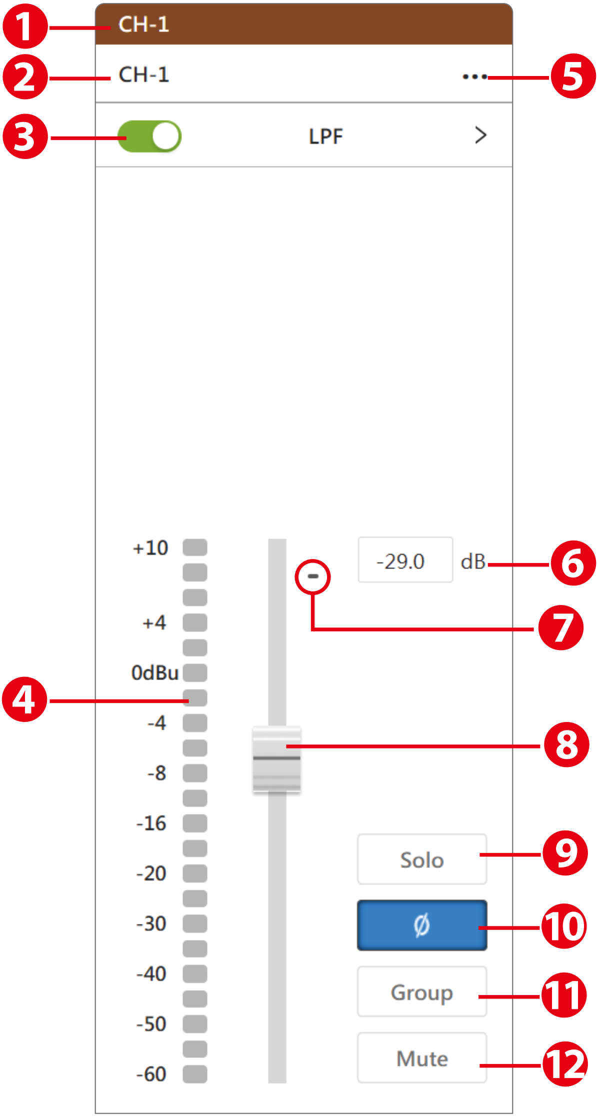


No.	Item	Description	Default
1	limiter switch	Enables or disables the limiter for the selected speaker output channel. When enabled, output level protection is applied based on the configured limiter parameters.	Off
2	soft knee	Sets how the limiter applies level reduction as the signal approaches the threshold. The limiter can operate with either a hard knee or soft knee response, affecting how abruptly or smoothly the output level is controlled. ◆ **Hard Knee**: A hard knee response is applied when the **Soft Knee** option is disabled. Limiting starts immediately once the signal reaches the threshold, resulting in a sharper cutoff curve in the graph. This produces a more immediate and assertive level reduction, delivering a tighter and more punchy output sound. ◆ **Soft Knee**: When **Soft Knee** is enabled, level reduction is applied gradually as the signal approaches the threshold, forming a smoother, rounded curve in the graph. This results in more natural and transparent level control, producing a smoother and more refined output sound.	On
3	attack setting	Sets how quickly the limiter starts reducing the output level after the signal exceeds the threshold. Shorter values apply limiting more rapidly.	1 ms
4	hold setting	Sets the duration for which the limiter maintains full level reduction after the signal falls below the threshold, before the release phase begins.	10 ms

No.	Item	Description	Default
5	release setting	Sets how quickly the limiter restores the output level to normal after level reduction is released.	10 ms
6	threshold setting	Sets the output level at which the limiter begins reducing signal level to prevent excessive output. **Note:** A hard knee setting starts the limiting process right after the signal reaches the threshold while a soft knee setting applies the limiting as the audio signal approaches the threshold.	-25 dB
7	limiter graph	Displays the limiter curve based on the current threshold and knee settings. The graph visually represents how signal levels are constrained once they reach the defined limit.	N/A

Line Out Processing

The Line Out section provides controls for managing line-level audio signals before they are sent to external audio devices. Users can apply output processing functions and adjust output levels for line-level signals delivered through the unit's line output channels:



No.	Item	Description	Default
1	channel ID	Displays the ID of this channel. Please note that the ID is unalterable.	N/A
2	channel name	Displays the name for this channel. To change the channel name, click the more button and select **Rename** to continue.	as channel ID
3	low-pass filter switch	Switch it on to enable the low-pass filter to cut the high frequency of the audio signal. See *Low-pass Filter (LPF) Configuration*, page 58.	Off
4	signal level meter	Show the audio signal levels in dBu.	N/A
5	more button	Click on the more button to open the option menu for further settings: ◆ **Rename**: Input a new name for this channel. ◆ **Copy**: Copy the settings of this channel and apply them to the selected channel(s). See *Renaming & Copying a Channel*, page 33.	N/A
6	channel fader level	Show the volume level of the audio signal to be output.	-80 dB
7	Fader Zero Reference Marker	Indicates the 0 dB (unity gain) reference position of the channel fader. The solid marker represents the reference point where the fader value is 0 dB.	N/A
8	channel fader	Adjust the volume level of the audio signal to be output. Drag the fader to change the value in decibel. The volume value also displays in the channel fader level field next to the channel fader.	-80 dB

No.	Item	Description	Default
9	solo	Allows monitoring of this channel through headphones connected to the AP412 front panel. When **Solo** is enabled on any channel, the signal reflects the post-processing output and is affected by the channel fader, mute, phase, and group settings. If the channel is muted, no audio from this channel will be heard, even when Solo is enabled.	Disabled
10	phase button	Click the button to invert the polarity of the phase. Disabling this function means that the phase polarity is normal.	Disabled
11	fader group button	Enable the group function to add this channel to the linked channels to simultaneously control the volume levels.	Disabled
12	mute button	Click on the mute button to enable or disable the mute function for this channel.	Disabled

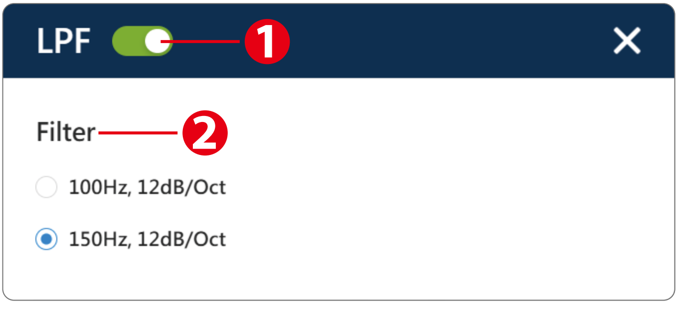
Low-pass Filter (LPF) Configuration

The low-pass filter (LPF) is available for line out channels to limit the output frequency range by attenuating content above a fixed cutoff point. This setting is typically applied when the line output is routed to low-frequency signal paths or external audio equipment that does not require high-frequency content.

LPF configuration is performed per line out channel and consists of enabling or disabling the filter and selecting a predefined cutoff frequency.

- **LPF Settings**

Use the LPF popup window to configure the low-pass filtering behavior for the selected line out channel:



1. **LPF Switch**

   Enables or disables the low-pass filter for the selected line out channel.

2. **Filter Selection**

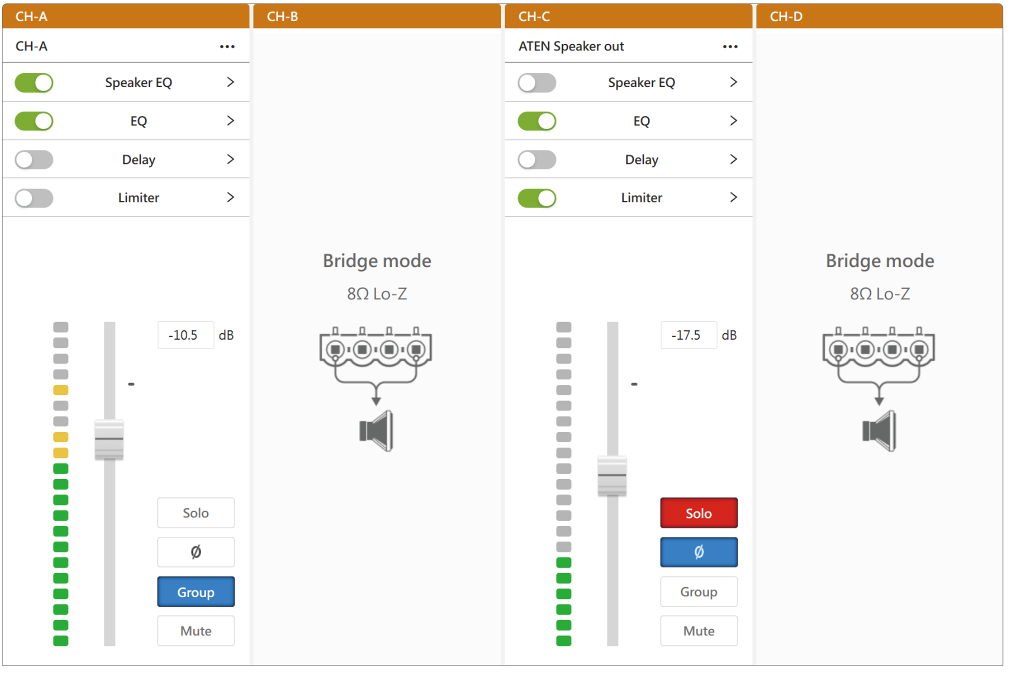
   Select a predefined cutoff frequency and slope using the radio buttons:

   - 100 Hz, 12 dB/Oct
   - 150 Hz, 12 dB/Oct

   The selected option determines the frequency above which audio content is progressively attenuated.

Bridge Mode

Bridge mode allows two amplifier output channels to be combined and operate together as a single, higher-power output channel.



When Bridge mode is enabled, the paired channels work in a push-pull configuration, effectively increasing the output voltage. As a result, the amplifier can deliver greater output power to a single speaker load or support high-impedance (Hi-Z) speaker systems.

On the AP412, the available bridge output types include 8Ω Lo-Z, 70V Hi-Z, and 100V Hi-Z. These bridge modes are determined by the device's hardware configuration (see *impedance / bridge switch for channel A & B*, page 9 and *impedance / bridge switch for channel C & D*, page 8). The web GUI displays the current Bridge mode status for reference, but does not provide controls for changing the bridge configuration.

As a result, Bridge mode is commonly used when:

- Higher output power is required
- Driving low-impedance (Lo-Z) speakers at higher wattage
- Driving high-impedance (Hi-Z) distributed speaker systems, such as 70 V or 100 V speaker lines

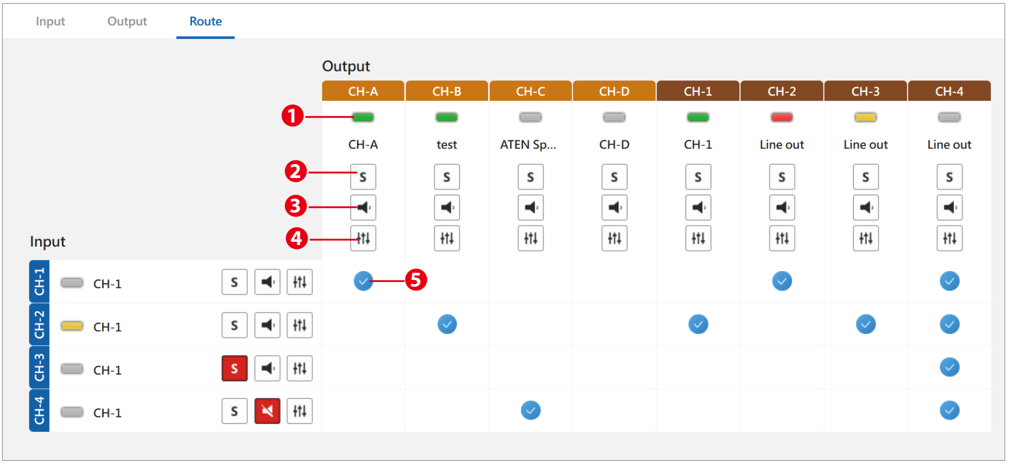
When Bridge mode is enabled, the bridged channel pair operates as a single output. Individual channel controls and routing options for the bridged channels may be limited or unavailable.

Always verify the speaker wiring, load impedance, and bridge output type before enabling Bridge mode to prevent improper operation or potential damage to the amplifier and connected speakers.

Route Tab

The **Route** tab provides controls for patching audio signals between input and output channels. Signals from any input channel can be assigned to one or more output channels, allowing flexible signal assignment across the system.

Note: Only **Administrator** can access to the route matrix page.

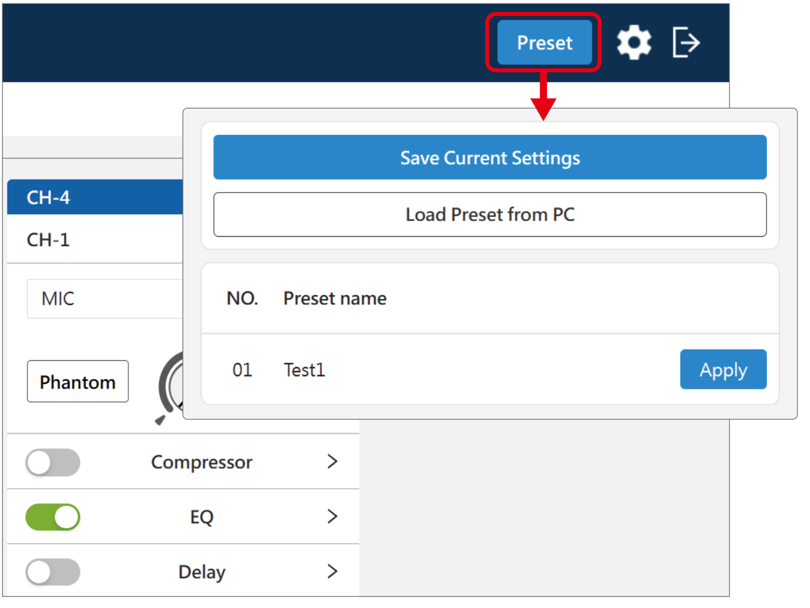


No.	Item	Description
1	signal status	Indicates the current signal status of the channel using color-coded indicators (green / yellow / red), reflecting the real-time signal level.
2	Solo	Enables headphone monitoring for this channel via the headphone jack on the AP412 front panel. When **Solo** is enabled, only one channel can be monitored at a time. The monitored signal bypasses channel-level controls such as fader, mute, phase, and priority, and is routed through the DSP processing stages only. This allows users to monitor the original signal even if the channel fader is set to minimum or the channel is muted.

No.	Item	Description
3	mute	Mutes or unmutes the selected channel. This setting is synchronized with the mute status on the **Input** / **Output** tab.
4	channel DSP	Displays the current DSP processing status of the channel. Click the button to expand the DSP status view. Enabled DSP functions are shown in blue, while disabled functions are shown in gray. 
5	crosspoint	In the graphical crossbar, each crosspoint represents a patching connection between an input and an output. Click a crosspoint to enable the signal connection; click it again to disable the connection.

Preset Management

All the settings you configured on the input tab, output tab,and route tab can be saved as a preset. You can easily switch to other set of settings by applying an existing preset.

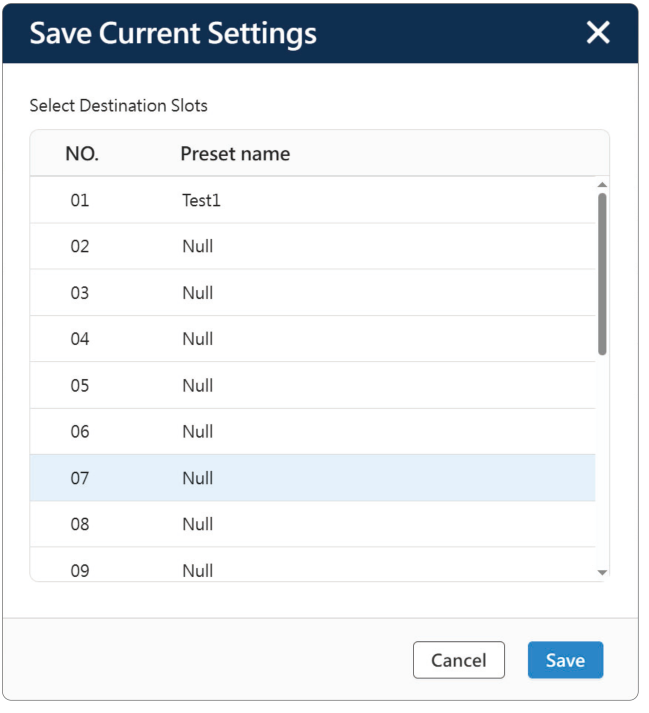


Click **Preset** button to expand the preset configuration menu for the following actions:

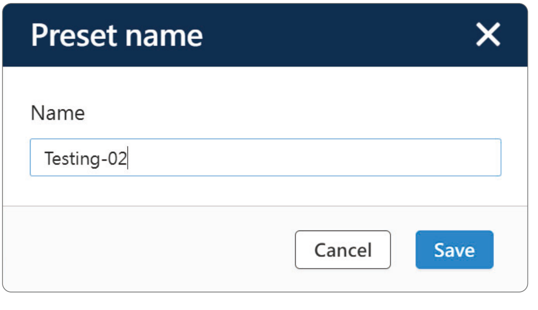
Save a New Preset

To save a new preset:

1. Click on **Save Current Settings** button to open **Save Current Settings** popup window.
2. Click to select a Null item, and click on the save button.

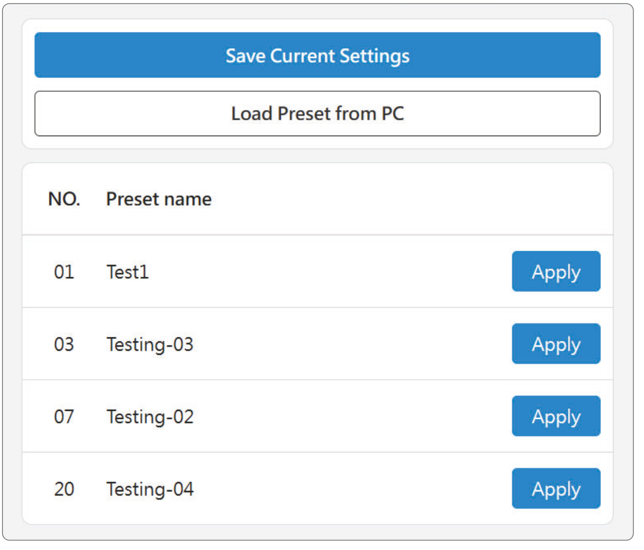


3. Enter the name for this preset to be saved , and click on the save button to complete.



Apply an Existing Preset

Click to open the preset configuration menu, and select the one you need to apply.



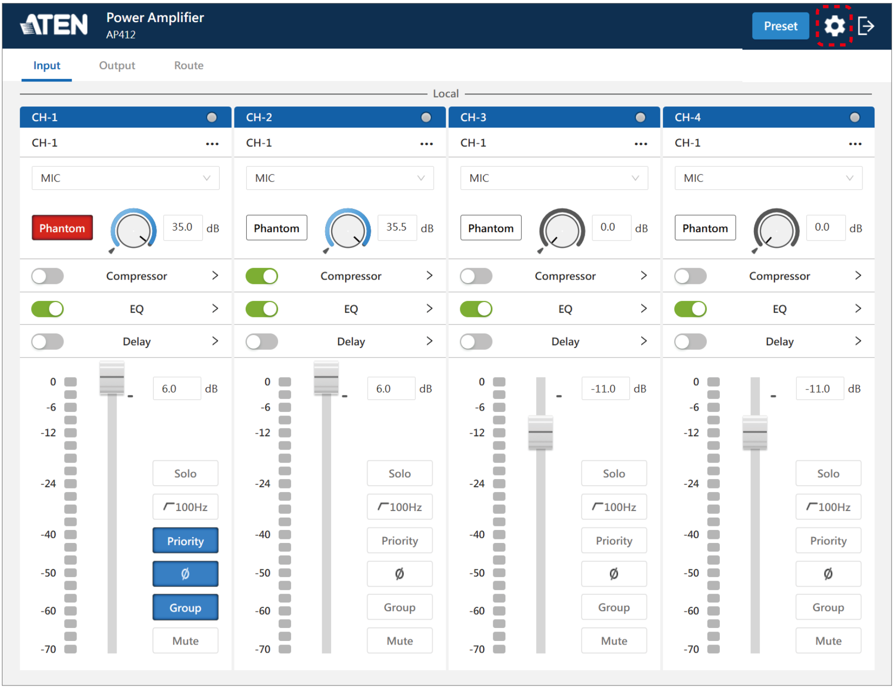
Edit an Existing Preset

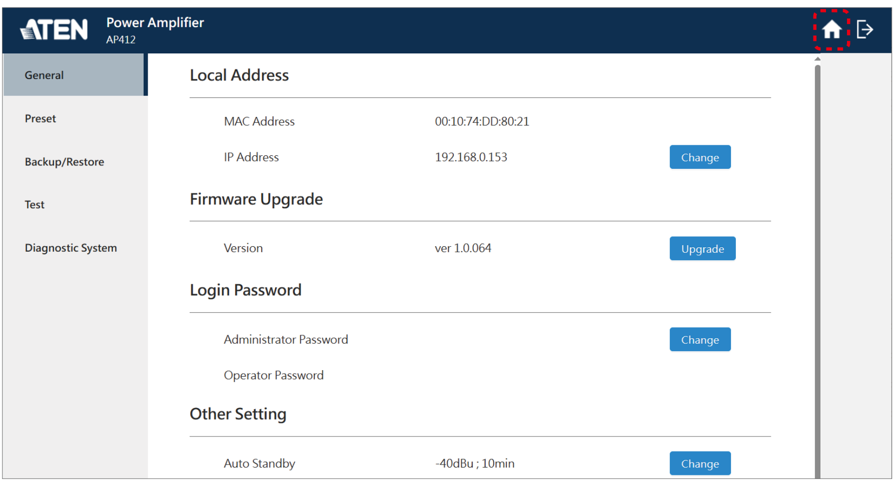
Fellow the steps below to edit an existing preset:

1. Apply the preset you'd like to edit.
2. Make changes of the preset. Once the configuration is done, click on the **Preset** button, then **Save Current Settings**.
3. The **Save Current Settings** popup window appears. Click on the preset name of this preset, and then click on the save button to save it.
4. Confirm to overwrite the preset to save your changed settings.

System Settings

Click on the setting button  on the DSP configuration screen to switch to the system settings screen, and here you can configure the followings:





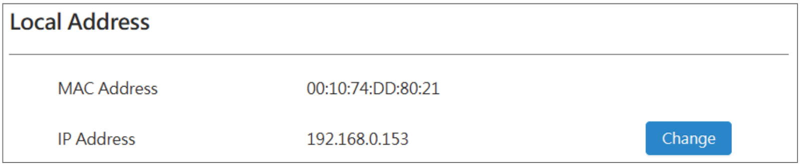
Tab	Supported Functions	Detailed Information
General	◆ Configure the network settings. ◆ Upgrade the unit's firmware. ◆ Set the inactivity duration that triggers the unit enters a low-power mode. ◆ Select the interface language. ◆ Configure the passwords for **Administrator** and **Operator**. ◆ Configure the account security settings.	For more information, see *General Tab*, page 68.
Preset	◆ Edit the existing preset(s). ◆ Delete the existing preset(s). ◆ Import and export the preset(s).	For more information, see *Preset Tab*, page 75.
Backup / Restore	◆ Create a backup file containing system presets and selected general settings. ◆ Restore presets and supported system configurations from a backup file. ◆ Maintain preset order consistency during backup and restore operations. ◆ Facilitate system recovery and configuration duplication.	See *Backup / Restore Tab*, page 78.
Test	◆ Calibrate the connected audio equipments.	For more information, see *Test Tab*, page 80.
Diagnostic System	◆ Check the status of the unit. ◆ Export the status report of the unit.	For more information, see *Diagnostic System Tab*, page 82.

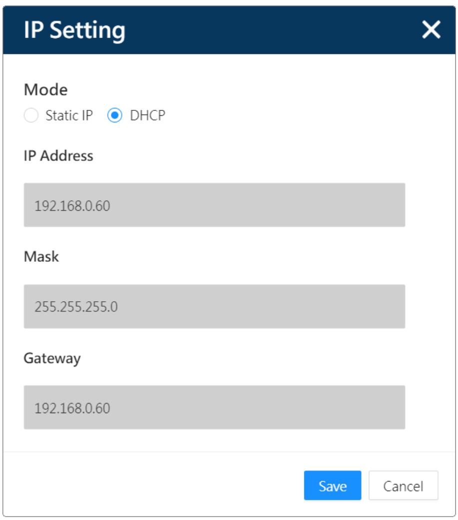
Note: Only **Administrator** can access to the system settings page.

General Tab

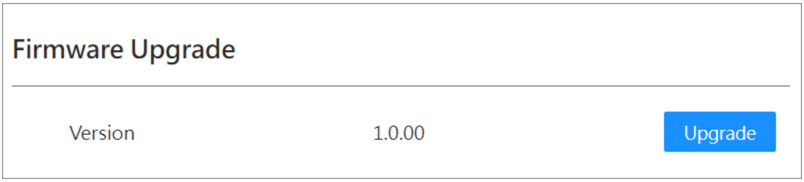
The **General** tab allows the **Administrator** to control or customize various settings that are used throughout the unit's configuration system.

Local Address



Item	Description
MAC Address	Shows the MAC address (media access control address) that is assigned to the unit connected to the network.
IP Address	Shows the IP address of this unit. Click on the change button next to the IP addresss to open the popup window to select the mode using the radio button:  ◆ **Static IP**: By selecting Static IP, the fields IP Address, Mask, and Gateway in this popup are available for you to define a fixed IP address for this unit. ◆ **DHCP**: Get the IP address that is assigned dynamically.

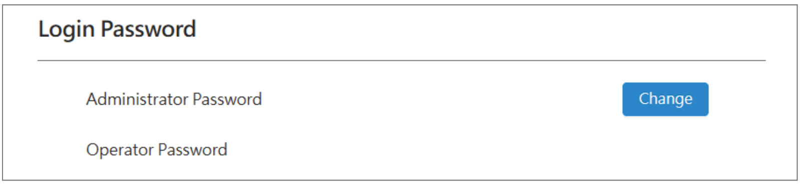
Firmware Upgrade

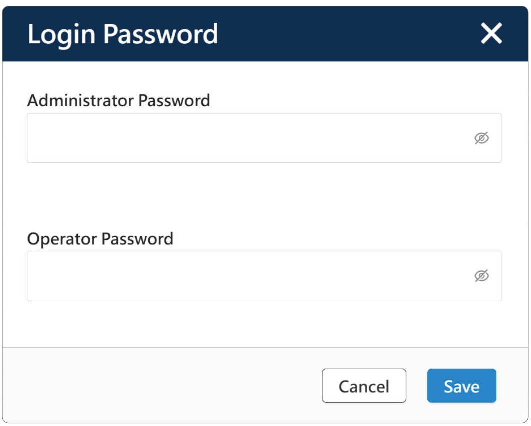


To upgrade the firmware of this unit, click the upgrade button to browse and select the firmware file from your PC.

Login Password

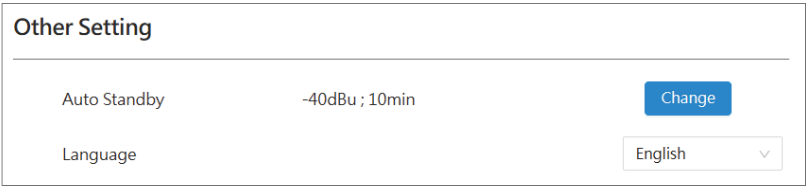
Allows users to change the login passwords for both **Administrator** and **Operator**.

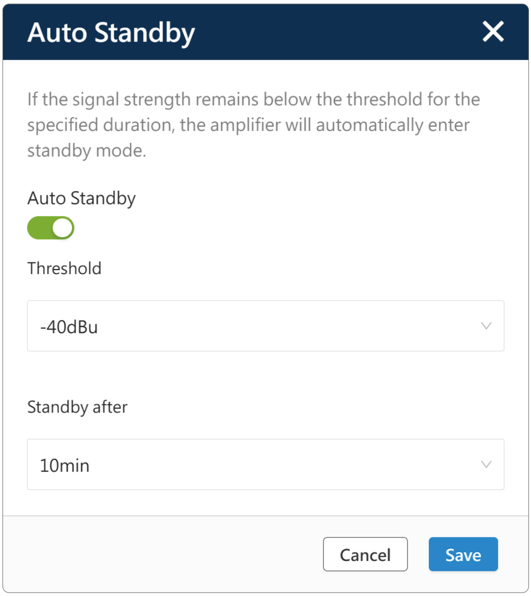




Other Settings

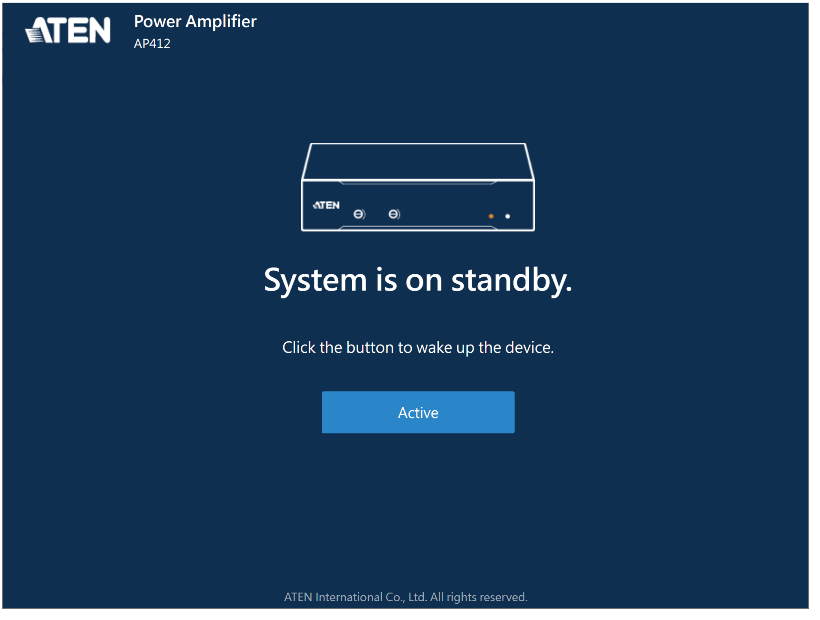
Click on the change button next to the function you'd like to configure to open its popup window for further settings:



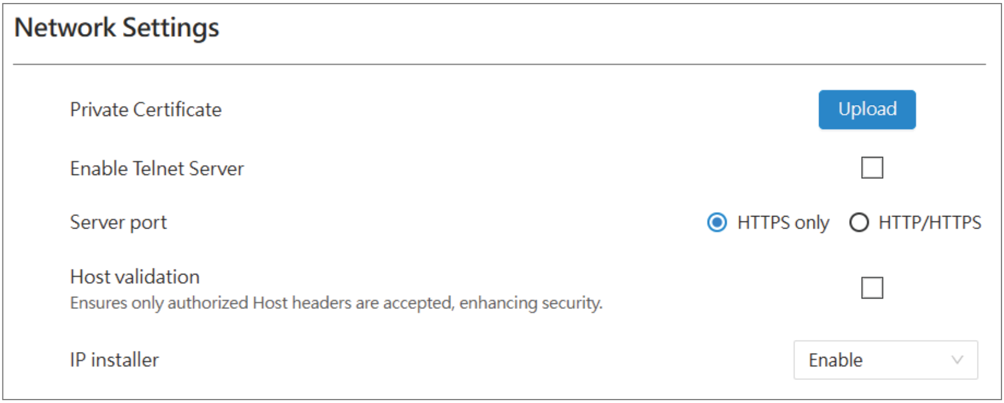
Item	Description	Default
Auto Standby	Define the followings by selecting the options from the drop-down menu:  ◆ **Auto Standby**: Enable or disable Auto Standby ◆ **Threshold**: Select the signal strength that trigger the wakeup of the unit. ◆ **Standby after**: Set the duration that the unit automatically enters standby mode after an elapsed time of inactivity.	◆ **Auto Standy switch**: On ◆ **Thresh-old**: -40 dbu ◆ **Standby after**: 15 minutes

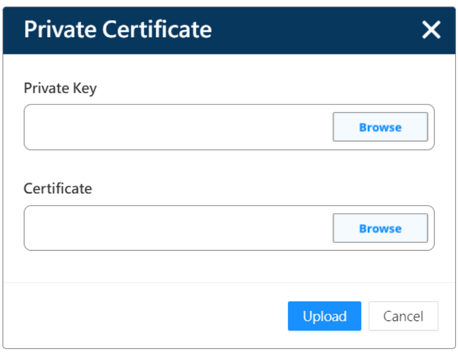
Item	Description	Default
Language	Use the drop-down menu to select the web GUI interface language.	PC OS language

Note: By enabling the auto standby settings, you have to wake up the unit once the it enters the standby mode. Follow the on-screen instructions to complete.



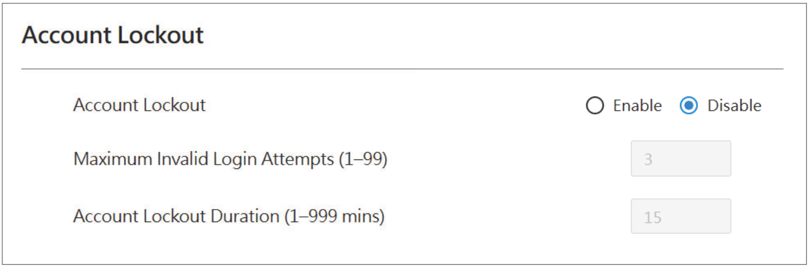
Network Settings



Item	Description	Default
Private Certifcate	When logging in over a secure (SSL) connection, a signed certificate is used to verify that the user is logging in to the intended site. For enhanced security, the Private Certificate section allows you to use your own private encryption key and signed certificate, rather than the default ATEN certificate. 	N/A
Enable Telnet Server	Check the checkbox to enable the Telnet login function that connects to the unit over a network to provide text based management and control.	Disabled

Item	Description	Default
Server Port	Use the radio button to select only use HTTP for a browser login or both HTTP and HTTPS which is for a secure browser login.	HTTPS only
Host validation	Enables or disables protection against forged HTTP Host headers. When enabled, the system validates the Host field in incoming HTTP requests to prevent attacks that exploit modified or fake Host headers, which could otherwise cause unauthorized redirects or security bypasses. ◆ **Enable**: Verifies the Host value in HTTP requests to enhance security. ◆ **Disable**: Turns off validation for compatibility or testing purposes.	Disable
IP Installater	**IP Installer** is an ATEN utility that allows users to discover devices on the network and view or configure their IP address settings. For details on viewing DHCP-assigned IP addresses, see *DHCP-assigned IP Address*, page 21. ◆ **Enable**: The unit's IP address can be found by IP Installer and configured through the Set IP function of IP Installer. ◆ **View Only**: The unit’s IP address can be found by IP Installer, but it cannot configured through IP Installer. ◆ **Disable**: The unit's IP address cannot be found by IP Installer.	Enable

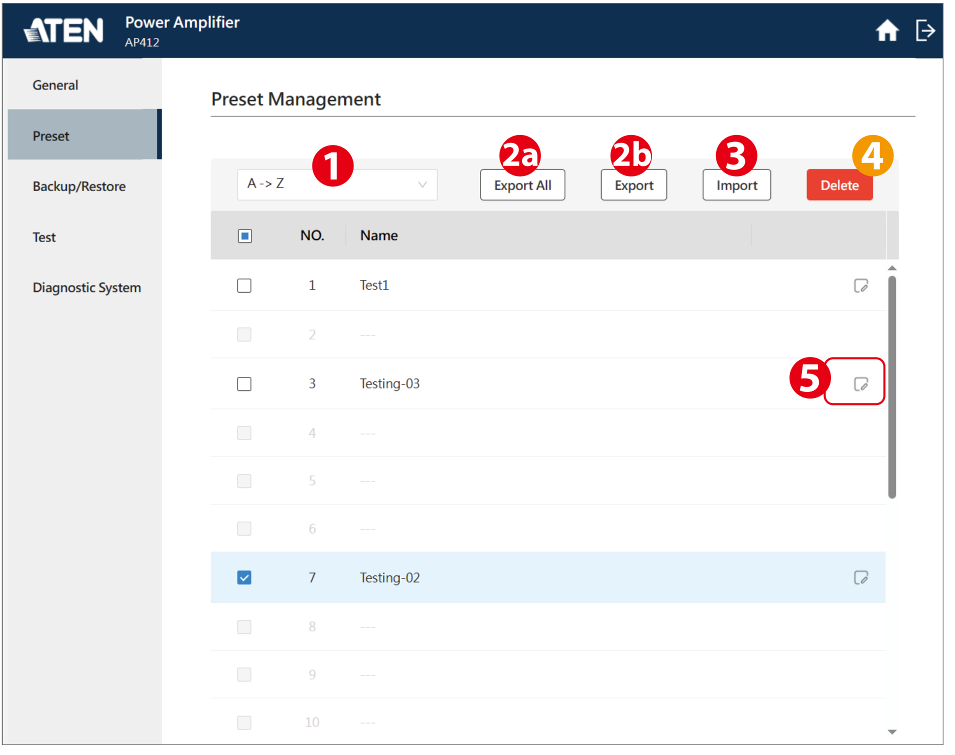
Account Lockout



Account Lockout helps to protect the accounts by limiting the number of failed login attempts.

Item	Description	Default
Account Lockout	To enable or disable the function that locks the login account after a certain number of failed sign-in attempts.	Enable
Maximum Invalid Login Attempts (1–99)	Enter the maximum number of failed sign-in attempts.	3 (attempts)
Account Lockout Duration (1–999 mins)	Define the number of minutes that a locked-out account remains locked out before it gets unlocked.	15 (minutes)

Preset Tab



A preset is a set of settings that defines how the DSP manipulate the audio signal and configured by users in the AP412 web GUI. **Preset** tab lists all the saved presets, and on this tab page, **Administrator** can perform the following actions:

1. **Sort and Display the Presets Alphabetically**:
   To sort the saved preset(s) in ascending or descending alphabetical order, select **A -> Z** or **Z -> A** from the drop-down menu.

2. **Export the Preset(s)**:
   To export existing presets as a single .bin file that may contain one or multiple presets for backup or reuse.

   a) Click **Export All** to export all presets into one .bin file to your PC.

   b) Select the checkbox(es) of the preset(s) you'd like to export, click **Export**, and save the generated .bin to your PC.

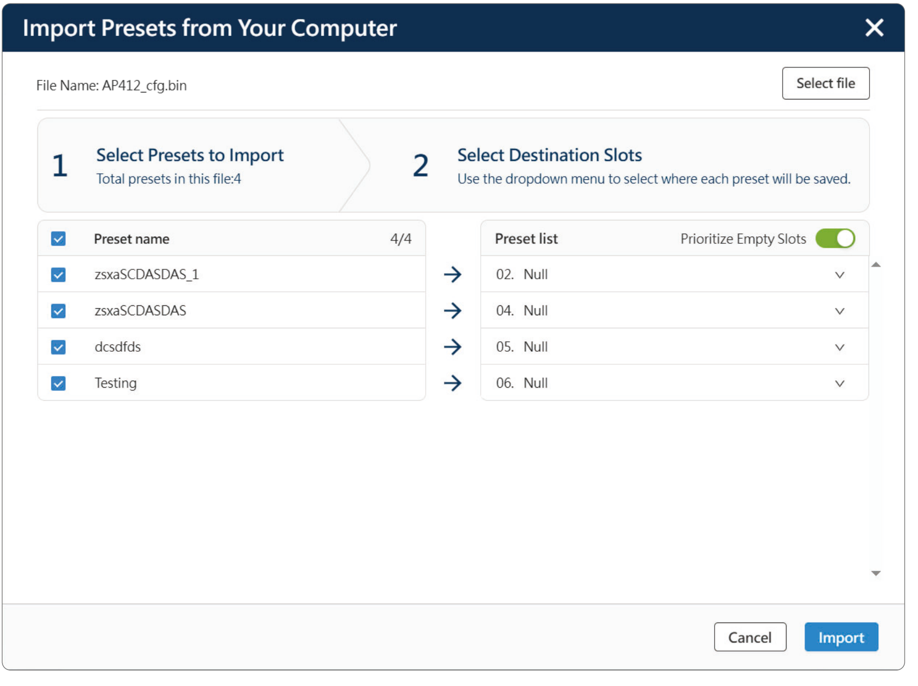
   Exported preset files can be imported again using the **Import** function.

3. **Import the Preset(s)**:
   To streamlining setup across units via configuration files, or to import your previously exported preset(s), do the following:

   a) Click the **Import** button to browse and select the .bin file saved on your PC.

   b) The **Import Presets from Your Computer** popup window appears.

   - Step 1 – **Select Presets to Import**:
     Select the preset(s) to be imported by checking the checkbox(es).
   - Step 2 – **Select Destination Slots**:
     Use the drop-down menus to assign each selected preset to a destination slot and arrange their order in the **Preset List**.

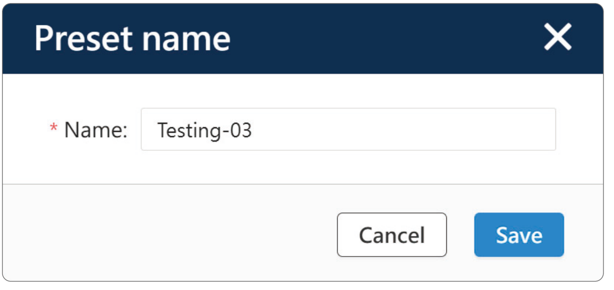


Note: By enabling **Prioritize Empty Slots** switch, the blank preset field will be preselected. To overwrite the existed preset field, manually select the field or disable **Prioritize Empty Slots** to preselect the filed from item no. 1.

   c) Click on the **Import** button to complete the action.

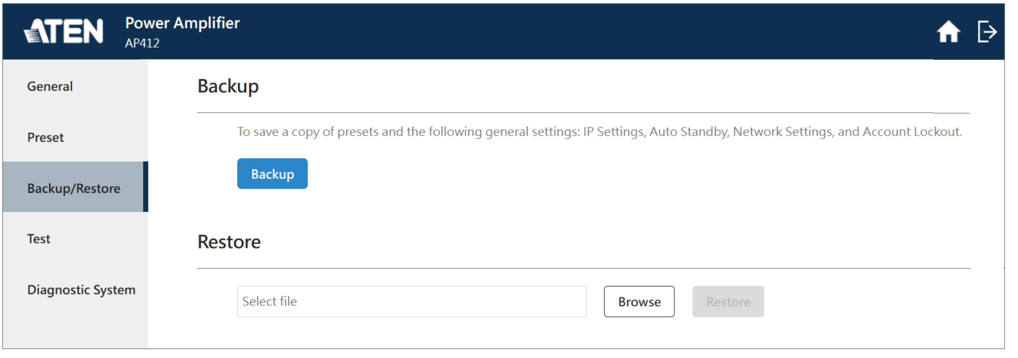
4. **Delete the Existing Preset(s):**
   Select the preset(s) to be removed, and then click on the **Delete** button.

5. **Change the Preset Name:**
   To change the name of the existed preset, click the edit button of the preset to be changed, enter the new name for this preset, and save the change. The preset name changes immediately.



Backup / Restore Tab

The **Backup** / **Restore** function allows users to save and recover system presets and selected general settings. This function is intended for system duplication, configuration recovery, and maintenance scenarios.



Backup

Click **Backup** to export a configuration file that includes:

- Preset data
    - Preset values
    - Preset names
    - Preset order (index)
- General settings
    - IP settings
    - Network settings
    - Auto Standby settings
    - Account Lockout settings

Note: User account information (administrator/operator passwords) is not included in the backup file.

The backup file preserves both the content and the original order of presets at the time of backup.

Restore

Use **Browse** to select a previously backed-up file, then click **Restore** to apply the settings to the unit.

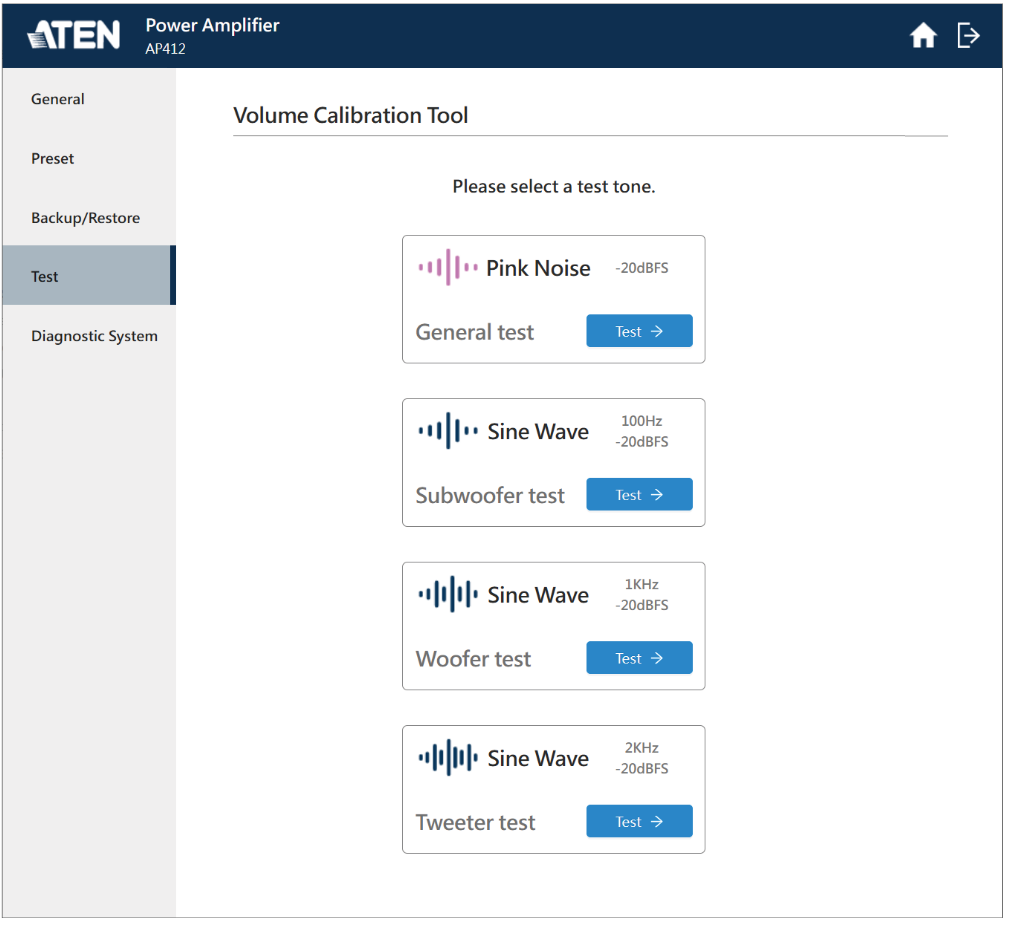
When restoring a backup file:

- All included presets are restored with their original preset order
- Existing presets on the unit are overwritten by the restored data
- Restored presets return to their original positions as defined in the backup file

The restore operation does not reassign presets based on current preset slots. Presets are restored strictly according to the order stored in the backup file.

Test Tab

Test tab offers the methods that helps to calibrate the volume levels:

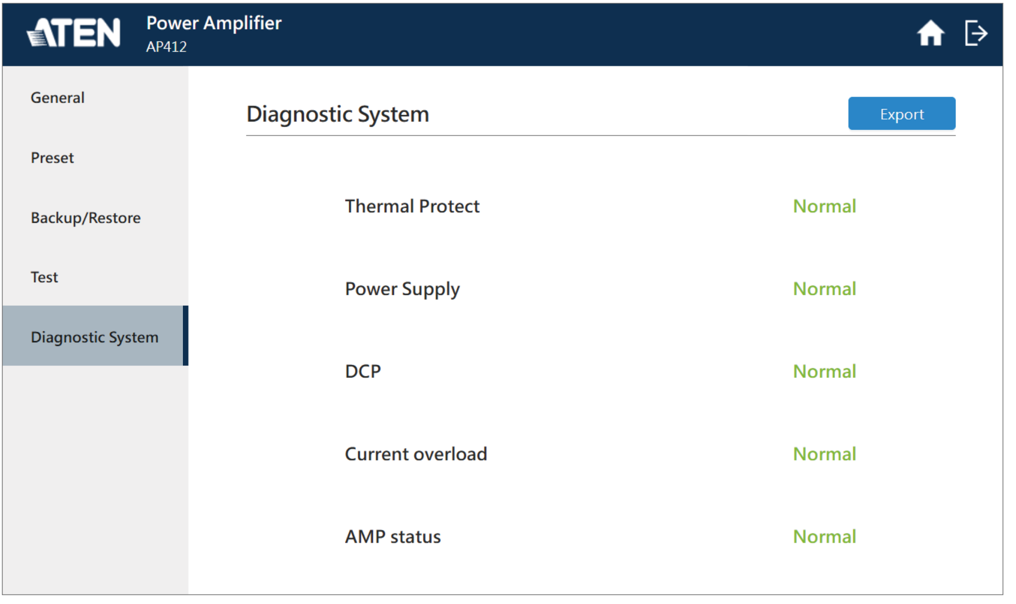


Item	Testing Wave-form	Description
General Test	Pink Noise	Pink noise is ideal for the output volume level calibration beacuse it consists of every frequency band at exactly the same level.

Item	Testing Waveform	Description
Subwoofer Test	Sine Wave	A sine wave represents a pure tone with a single frequency. Use the pure and consistent signal to measure and calibrate the performance of the audio equipments.
Woofer Test		
Tweeter Test		

Follow the on-screen instructions to complete the test procedure and take the suggested action to adjust the gain settings if needed.

Diagnostic System Tab



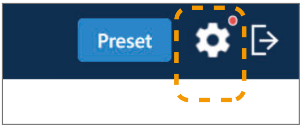
Diagnostic System tab identifies the following operational status and the problems on the AP412 unit.

- Thermal Protect
- Power Supply
- DCP
- Current Overload
- AMP Status

Through the **Export** button, a report in .txt file format is generated. Use the generated report that contains the unit's event logs for troubleshooting if needed.

Critical Notifications

When a non-normal condition is detected on the amplifier, a red badge appears next to the **Settings** button on the DSP configuration screen. Go to the **Diagnostic System** tab to view the detailed notification.



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Chapter 4
DSP Effects on Audio Output

Overview

This chapter explains how DSP processing shapes the final audio output of the AP412.

Instead of describing how to operate individual controls, this section focuses on how DSP functions such as equalization, compression, delay, and limiting affect the signal after configuration. It explains how different parameter settings influence tonal balance, dynamics, timing, and speaker behavior.

By linking DSP parameters to their audible results, this chapter helps users understand the relationship between on-screen settings and actual sound output. It is intended as a conceptual reference to support system tuning, environment-specific adjustment, and more predictable control over the resulting audio performance.

Compressor

A compressor controls the dynamic range of an audio signal by reducing the level of signals that exceed a defined threshold. By managing sudden level changes and limiting excessive peaks, compression helps maintain a more consistent and controlled output level.

In audio output processing, a compressor is commonly used to improve signal stability, prevent overload, and ensure balanced loudness across different program materials.

Controls

The compressor parameters define how level reduction is applied once the input signal level exceeds the threshold.

- **Threshold**
  Determines the signal level at which compression begins
- **Ratio**
  Defines how much the signal is reduced once it exceeds the threshold
- **Attack**
  Controls how quickly level reduction is applied after the threshold is crossed
- **Release**
  Controls how quickly the signal returns to its normal level after it falls below the threshold
- **Hold**
  Determines how long full level reduction is maintained before the release phase begins

Compression Behavior

- Increasing the ratio results in stronger compression and greater control over peaks.
- A short attack time applies stronger control to sudden increases in signal level.
- A longer attack time allows initial level increase to pass through, preserving impact.
- A short release time restores gain quickly but may cause audible pumping.
- A longer release time produces smoother and more natural level recovery.

Soft Knee vs. Hard Knee

The knee setting determines how compression is applied around the threshold:

- **Hard knee**

  Hard knee compression applies full level reduction immediately once the signal exceeds the threshold, resulting in a more abrupt compression response.

- **Soft knee**

  Soft knee compression starts level reduction earlier, before the signal reaches the threshold.

  This allows the compression effect to build up more smoothly and results in more transparent dynamic control.

Effect on Audio Output

Properly applied compression helps:

- Prevent sound distortion
- Maintain consistent loudness levels
- Improve clarity in dynamic audio content

Equalizer

The equalizer (**EQ**) allows users to adjust the level of an audio signal at specific frequency ranges. By boosting or attenuating selected frequencies, EQ can be used to shape the tonal balance of the audio signal and adapt it to different input sources or listening environments.

In audio output processing, an equalizer is commonly used to enhance clarity, compensate for acoustic characteristics, and fine-tune the overall sound profile.

Controls

The EQ parameters define how gain is applied across different frequency bands of the audio signal.

- **HPF (High-Pass Filter)**
  Removes low-frequency content below a specified cutoff frequency to reduce unwanted low-end noise or hum.
- **Band 1–4**
  Each band represents an adjustable frequency range within the EQ.

  The following parameters are available for each band:
  - **Frequency**
    Determines the center frequency of the selected band.
  - **Gain**
    Defines the amount of boost or attenuation applied to the selected frequency range.
  - **Q**
    Controls the width of the affected frequency range around the center frequency.

Equalization Behavior

- Increasing gain emphasizes the selected frequency range, making it more prominent.
- Decreasing gain attenuates the selected frequency range, reducing its presence.
- A higher Q value affects a narrower range of frequencies, allowing for more precise tonal adjustments.
- A lower Q value affects a wider range of frequencies, resulting in broader tonal shaping.

Effect on Audio Output

Properly applied equalization helps to:

- Improve overall tonal balance and clarity
- Reduce unwanted resonances or frequency masking to prevent system feedback
- Adapt the audio output to different acoustic environments
- Optimize sound characteristics for various program materials

Delay

Delay controls the timing of an audio signal by introducing a precise time offset before output. By aligning the arrival time of sound from different speakers, delay processing ensures that audio reaches the listening position simultaneously, improving coherence and spatial accuracy.

In audio output processing, delay is commonly used to compensate for differences in speaker distance and environmental conditions, helping to maintain clarity, localization, and a consistent listening experience across the coverage area.

Controls

The delay parameters define how time alignment is applied to the audio output signal.

- **Delay**
  Specifies the amount of time the audio signal is postponed before output. Delay is measured in milliseconds and directly determines the temporal offset applied to the signal.
- **Distance**
  Defines the physical distance between the speaker and the listening position. The system automatically converts the specified distance into an appropriate delay time based on the speed of sound.
- **Temperature**
  Accounts for ambient temperature variations that affect the speed of sound. Temperature compensation enables more accurate delay calculation, particularly in large or acoustically sensitive environments.

Delay Behavior

- Increasing the delay time postpones audio output, allowing farther speakers to synchronize with closer ones.
- Distance-based delay simplifies time alignment by translating physical speaker placement into timing compensation.
- Temperature compensation refines delay accuracy by adapting calculations to real-world environmental conditions.

Effect on Audio Output

Properly applied delay processing helps to:

- Align sound arrival across multiple speakers
- Reduce phase-related interference caused by uneven speaker distances
- Create a more focused and coherent listening experience

Limiter

A limiter controls the maximum output level of an audio signal by enforcing a defined ceiling. When the signal approaches or exceeds this limit, level reduction is applied to prevent further level increase.

In audio output processing, a limiter is primarily used as a protection mechanism. By controlling extreme peaks before amplification and playback, limiting helps prevent speaker overload, distortion, and potential damage caused by unexpected signal spikes.

Controls

The limiter parameters define how peak protection is applied to the audio output signal.

- **Threshold**
  Defines the maximum output level allowed. When the signal exceeds this threshold, the limiter applies level reduction to prevent further level increase.
- **Soft Knee**
  Determines how gradually limiting is applied as the signal approaches the threshold. Soft knee limiting introduces level reduction progressively, resulting in smoother and less perceptible control compared to a hard knee response.
- **Attack**
  Controls how quickly the limiter responds once the signal exceeds the threshold. Shorter attack times clamp sudden increases in signal level more aggressively, while longer attack times allow brief increases in level to pass through.
- **Hold**
  Defines how long full level reduction is maintained after the signal falls below the threshold. The release phase does not begin until the hold time has elapsed.
- **Release**

Controls how quickly level reduction is removed after the signal drops below the threshold. Shorter release times restore signal level faster, while longer release times produce smoother level recovery.

Limiting Behavior

- When the signal exceeds the threshold, level reduction is applied immediately to cap output level.
- Short attack times improve peak containment, while longer attack times preserve transient detail.
- Hold time stabilizes level reduction and prevents rapid level fluctuations.
- Release time determines how smoothly the signal returns to its normal level after limiting.

Effect on Audio Output

Properly applied limiting helps to:

- Prevent sudden signal peaks from exceeding safe output levels
- Protect speakers from distortion, overload, or damage
- Maintain consistent maximum loudness
- Provide transparent peak control in music and speech applications

Low-pass Filter (LPF)

A low-pass filter (LPF) limits the frequency range of an audio signal by attenuating content above a specified cutoff frequency. Frequencies below the cutoff are allowed to pass through, while higher-frequency components are progressively reduced.

In audio output processing, LPF is typically applied to line-level outputs to control spectral content before the signal is delivered to downstream audio devices. This allows the output signal to better match the frequency characteristics of connected equipment.

Controls

The LPF defines how high-frequency content is managed in the output signal:

- **Cutoff Frequency**
  Determines the frequency above which audio content is progressively attenuated. Frequencies below the cutoff pass through unaffected, while higher frequencies are reduced according to the filter slope.
- **Filter Slope**
  Defines how steeply frequencies above the cutoff are attenuated. A steeper slope results in more aggressive high-frequency reduction, while a gentler slope provides a smoother transition.

Filtering Behavior

Applying an LPF alters the spectral balance of the output signal:

- High-frequency content above the cutoff frequency is reduced, focusing the output on low-frequency energy.
- The transition between passband and attenuated frequencies is determined by the selected slope, affecting how natural or controlled the filtering sounds.
- LPF operates continuously on the signal path once applied, shaping the output before it reaches connected devices.

Effect on Audio Output

Proper use of LPF helps:

- Optimize line-level outputs for devices such as subwoofers or low-frequency speaker systems
- Prevent frequency overlap between different audio paths or speaker zones
- Ensure better integration with external processors or crossover systems

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Appendix

Safety Instructions

General

- This product is for indoor use only.
- Read all of these instructions. Save them for future reference.
- Follow all warnings and instructions marked on the device.
- Do not place the device on any unstable surface (cart, stand, table, etc.). If the device falls, serious damage will result.
- Do not use the device near water.
- Do not place the device near, or over, radiators or heat registers.
- The device cabinet is provided with slots and openings to allow for adequate ventilation. To ensure reliable operation, and to protect against overheating, these openings must never be blocked or covered.
- The device should never be placed on a soft surface (bed, sofa, rug, etc.) as this will block its ventilation openings. Likewise, the device should not be placed in a built in enclosure unless adequate ventilation has been provided.
- Never spill liquid of any kind on the device.
- Unplug the device from the wall outlet before cleaning. Do not use liquid or aerosol cleaners. Use a damp cloth for cleaning.
- The device should be operated from the type of power source indicated on the marking label. If you are not sure of the type of power available, consult your dealer or local power company.
- To prevent damage to your installation it is important that all devices are properly grounded.
- Do not allow anything to rest on the power cord or cables. Route the power cord and cables so that they cannot be stepped on or tripped over.
- Position system cables and power cables carefully; Be sure that nothing rests on any cables.
- Never push objects of any kind into or through cabinet slots. They may touch dangerous voltage points or short out parts resulting in a risk of fire or electrical shock.

- Do not attempt to service the device yourself. Refer all servicing to qualified service personnel.
- If the following conditions occur, unplug the device from the wall outlet and bring it to qualified service personnel for repair.
  - The power cord or plug has become damaged or frayed.
  - Liquid has been spilled into the device.
  - The device has been exposed to rain or water.
  - The device has been dropped, or the cabinet has been damaged.
  - The device exhibits a distinct change in performance, indicating a need for service.
  - The device does not operate normally when the operating instructions are followed.
- Only adjust those controls that are covered in the operating instructions. Improper adjustment of other controls may result in damage that will require extensive work by a qualified technician to repair.
- To prevent electric shock, please do not remove the top cover as there are no user serviceable parts inside. Please refer to qualified service personnel for servicing.
- To completely disconnect this apparatus from the AC mains, disconnect the power supply cord plug from the AC receptacle.
- Exposed high voltage on the speaker out. Touching uninsulated terminals of wiring may result in an unpleasant sensation.

- The lightning symbol in the triangle indicates insinuated components inside the housing with hazardous voltages that can cause injury to persons.

Rack Mounting

- Before working on the rack, make sure that the stabilizers are secured to the rack, extended to the floor, and that the full weight of the rack rests on the floor. Install front and side stabilizers on a single rack or front stabilizers for joined multiple racks before working on the rack.
- Always load the rack from the bottom up, and load the heaviest item in the rack first.
- Make sure that the rack is level and stable before extending a device from the rack.
- Use caution when pressing the device rail release latches and sliding a device into or out of a rack; the slide rails can pinch your fingers.
- After a device is inserted into the rack, carefully extend the rail into a locking position, and then slide the device into the rack.
- Do not overload the AC supply branch circuit that provides power to the rack. The total rack load should not exceed 80 percent of the branch circuit rating.
- Make sure that all equipment used on the rack – including power strips and other electrical connectors – is properly grounded.
- Ensure that proper airflow is provided to devices in the rack.
- Ensure that the operating ambient temperature of the rack environment does not exceed the maximum ambient temperature specified for the equipment by the manufacturer.
- Do not step on or stand on any device when servicing other devices in a rack.
- For desktop mounting, install the unit in an open and unobstructed area.

Technical Support

International

- For online technical support – including troubleshooting, documentation, and software updates: http://support.aten.com
- For telephone support, see *Telephone Support*, page iv:

North America

Email Support		support@aten-usa.com
Online Technical Support	Troubleshooting Documentation Software Updates	http://www.aten-usa.com/support
Telephone Support		1-888-999-ATEN ext 4988

When you contact us, please have the following information ready beforehand:

- Product model number, serial number, and date of purchase
- Your computer configuration, including operating system, revision level, expansion cards, and software
- Any error messages displayed at the time the error occurred
- The sequence of operations that led up to the error
- Any other information you feel may be of help

Specifications

AP412

Function	AP412
System Specification	
Amplifier Type	Class D
Rated Power	120W per Channel @4Ω or 8Ω 240W per Channel @8Ω Bridge or 70/100V Hi-Z Bridge
Sensitivity	Balanced: +4 dBu Unbalanced: -10 dBV (0 dBu=0.775 Vrms, 0 dBV=1 Vrms)
Distortion	THD+N: <0.1%, 1 kHz, 3 dB before clipping
Frequency Response	50–20 kHz, +/-2.5 dB @4Ω, 1W, MIC/Line Input to SPK Output
Microphone Inputs	
Gain Range	0 dB to +36 dB
Impedance	3kΩ
Phantom power	+48V
Line Inputs	
Gain Range	Bal: -6 dB to +4 dB UnBal: -6 dB to +10 dB
Impedance	Bal: 5kΩ UnBal: 2.5kΩ
Maximum Input Level	+19 dBu / Nominal: +4 dBu
Line Outputs	
Impedance	200Ω
Maximum output Level	+10 dBu / Nominal: +4 dBu
Headphone Outputs	
Frequency Response	50–20 kHz, -3 dB @10 mW
Maximum output Level	>30 mW @32Ω

Function	AP412
Audio Effects	Built-In DSP (COMP, EQ, Delay, Priority, Limiter)
Protection	Output Shortcut AMP Output over/under voltage High Frequency Overload Thermal Protect
Communication	
RS-232	Connector: 3-pin, 3.5 mm Terminal Block Baud rate and protocol: ◆ Baud Rate: 19200 ◆ Data Bits: 8 ◆ Stop Bits: 1 ◆ Parity: No ◆ Flow Control: No
Compliance	
Certification	FCC, CE, UKCA, IEC62368-1:2018
Power	
Input Power Rating	100–240 V~, 50/60 Hz, 1.5 A with PFC
Power Consumption	AC110V; 81W; 380BTU/h AC220V; 76.7W; 360BTU/h **Note:** 1/8 Max. Power Output
Environmental	
Operating Temperature	0 °C to 50 °C
Humidity	0% to 80% RH, Non-Condensing
Storage Temperature	-20 °C to 60 °C
Physical Properties	
Weight	5.21 kg (11.48 lb)
Housing	Metal
Dimensions (L × W × H)	43.24 × 33.97 × 4.40 cm (17.02 × 13.37 × 1.73 in.)

ATEN Warranty Policy

The warranty policy may vary by product category and region of purchase. For details, please visit ATEN's official website, select your purchase counties/ regions and then go to the Support Center, or contact your local ATEN sales representative for further assistance.

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