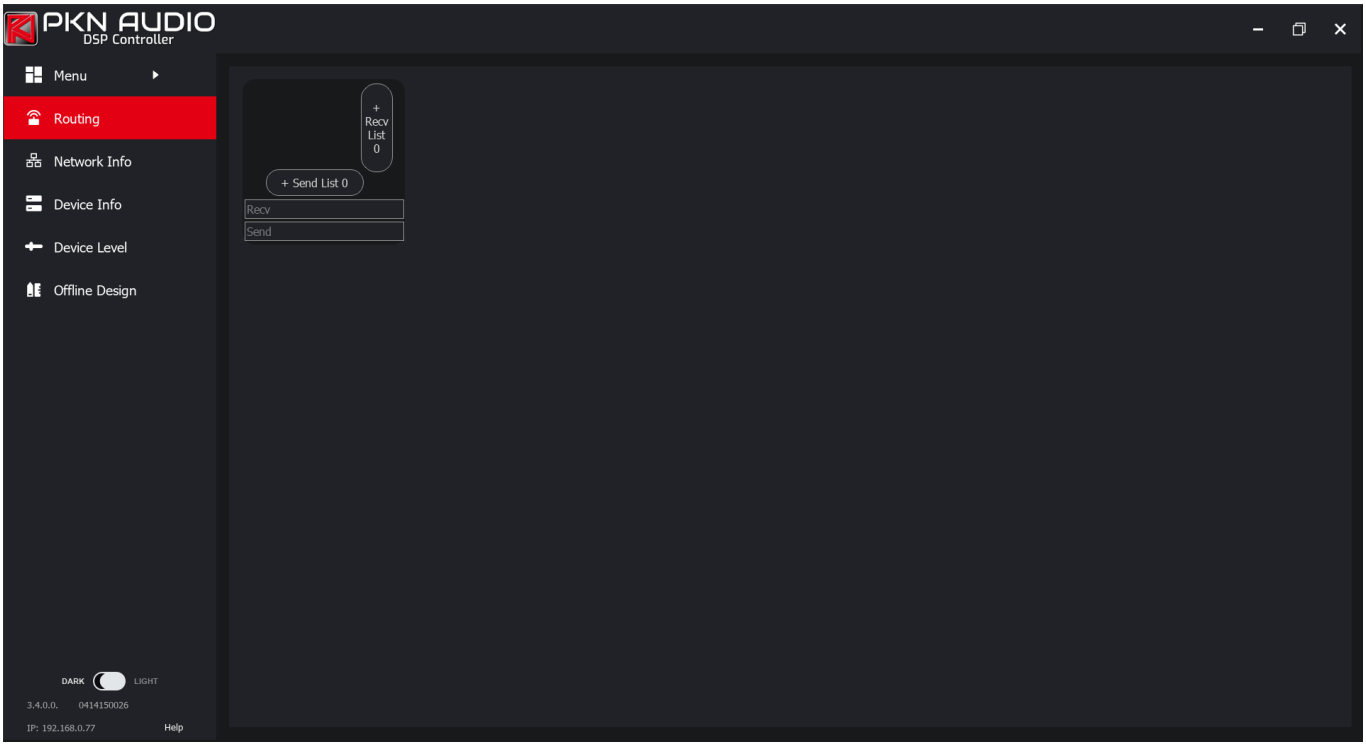
PKN Audio DSP Controller Instruction Manual



The **PKN Audio DSP Controller** software is a comprehensive platform, that covers all Digital Signal Processing and Network Routing functions from the simplest to the most complex audio systems.

The system is capable of routing and distributing both analog and digital audio signals, perform high quality Digital Signal Processing, and handle the controlling and monitoring of all compatible devices in the local area network (LAN), while maintaining lossless audio signal quality and ultra-low latency in accordance with the AES67 standards.

Due to high demand, we implemented Offline Designer features as well, so you can configure any possible scenario without even having our device turned on, on the network.

Another much requested addition is the AmpControl implementation for our XE-DSP devices, which makes remote control more convenient than ever.

Apart from our current hardware, we also support software solutions for digital audio streaming, through an AES67 enabled virtual sound card (VSC). (Optional)

Table of Contents

1. PKN Audio DSP Controller Software installation....3
2. Discover Devices....3
3. Main View – Left Side Menus / Buttons....5
3.1 Routing....5
3.2 Network Info....6
3.3 Device Information....7
3.4 Device Level....7
3.5 Offline Design....8
3.6 Dark ↔ Light Theme Switch....10
3.7 Help Button....11
4. Device Details....11
4.1 Device Config....11
4.2 Stream....13
4.2.1 Unicast Stream....13
4.2.2 Multicast Stream....13
4.3 DSP....14
4.3.1 Diagram....14
4.3.2 Channel Volume....15
4.3.3 Matrix....15
4.3.4 Preset Management....16
4.4 Volume....17
4.5 Latency....17
4.6 Clock Log....18
4.7 Status....18
4.8 AmpControl (XE-DSP Specific)....19
5. Advanced DSP Features....20
5.1 DSP Signal Flow Chart....20
5.2 DSP Editor View....21
5.3 Detailed Explanation of DSP Functions....22
5.3.1 HPF – High Pass Filter....22
5.3.2 LPF - Low Pass Filter....22
5.3.3 EQ - Parametric Equalizer....23
5.3.4 DELAY....24
5.3.5 FBE - Feedback Eliminator....24
5.3.6 COMPRESS / COMPRESS-II – Compressor / Multi-Segment DRC....24
5.3.7 FIR / FIR-II - Finite Impulse Response Filter....26
5.3.8 LIMIT – Peak Limiter....26
5.3.9 EXPAND - Expander....27
5.3.10 NOISEGATE - Noise Gate....27
5.3.11 SIGNALGEN - Signal Generator....27
5.3.12 DUCKER - Sidechain Attenuation....28
6. Firmware Update....29
7. Troubleshooting....30

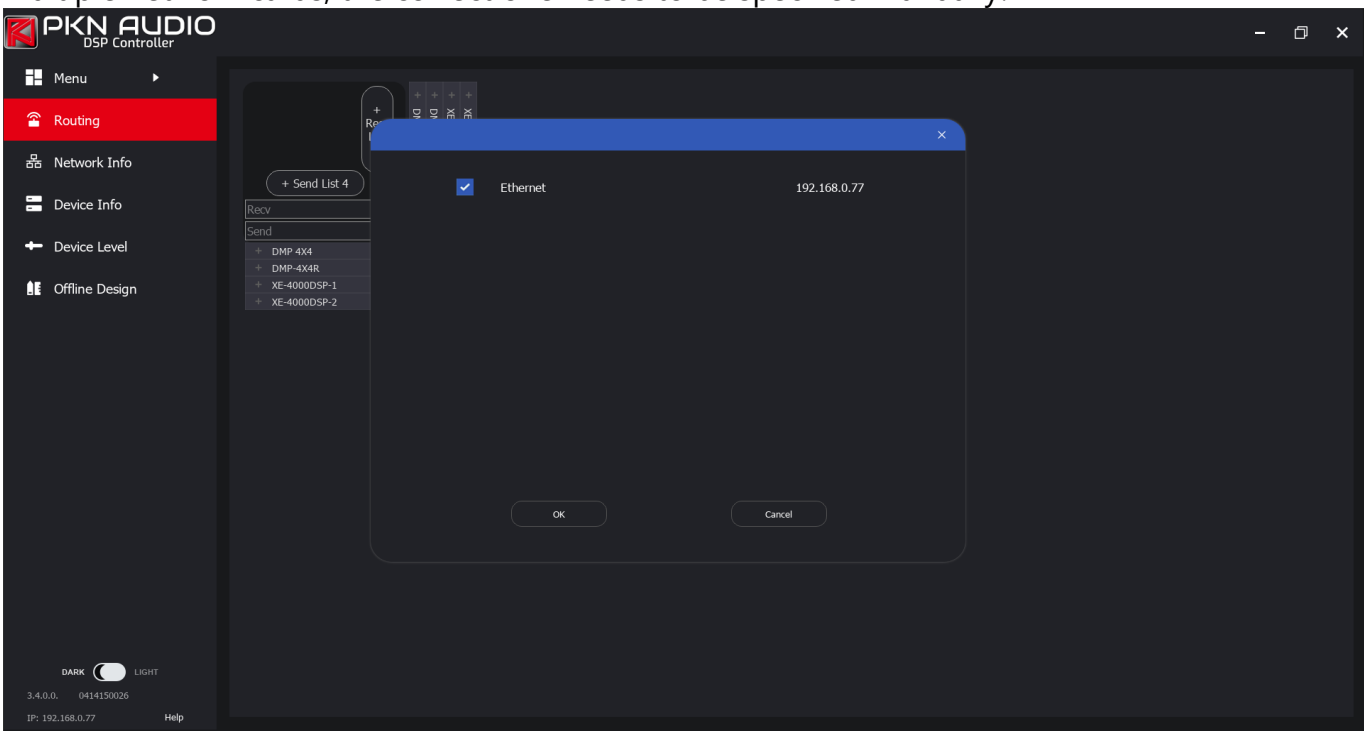
1. PKN Audio DSP Controller Software installation

Make sure you have downloaded our latest software from our official website.

- Double-click the software installation package to install.
- Choose the install location, then click *Next*.
- Choose if you need a desktop shortcut, and/or quick launch icon. After selection, click Next.
- Confirm if your choices are correct and click Install.
- Wait for the installation to be completed.
- Choose whether to launch the application, or exit the installer once you click Finish.
- Make sure to allow the software through the firewall when its prompted, or you can experience abnormalities and issues with device detection and/or correct controlling.

2. Discover Devices

The software discovers and controls devices through LAN (local area network).
Each time, when you open the software, it is necessary to specify which network card of the computer should be used (*different network cards may correspond to different local area networks*).
If there is only one present, the software will select it by default, but if there are multiple network cards, the correct one needs to be specified manually.



If the network cable is unplugged or re-plugged, or there is any network change, the corresponding network card needs to be selected again.

(Note that the Software also works on wireless connection, but only for control. If you consider using digital audio streaming via VSC, a wired connection is absolutely necessary.)

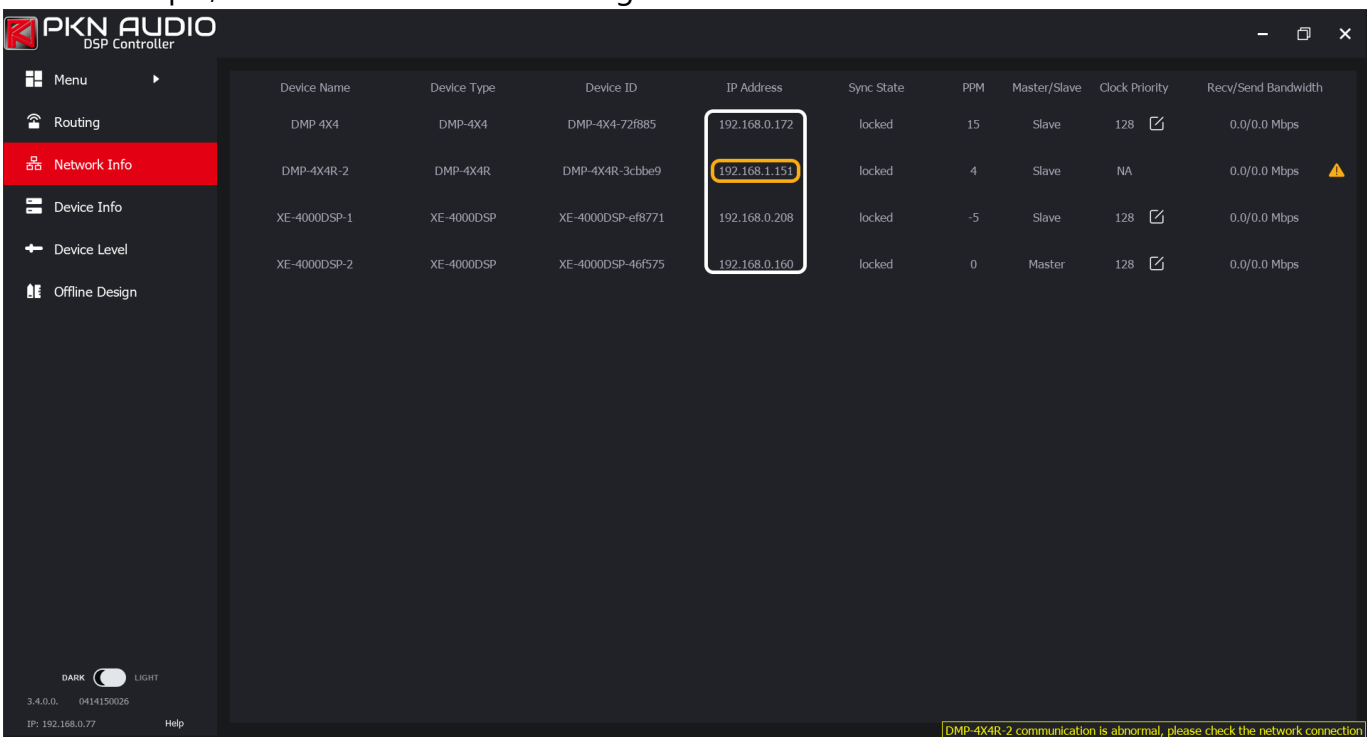
You can select the correct network card anytime by clicking the *Menu* and selecting `Set Network Card` from the drop-down menu:



After selecting the correct network card, the software will automatically discover the devices in the corresponding Local Area Network. This might take up to 5 seconds.

If there are any anomalies, a notification shows up on the bottom right, and if you switch to the `Network Info` view, you can see a warning sign next to the affected device.

In this example, the IP address is in the wrong domain:

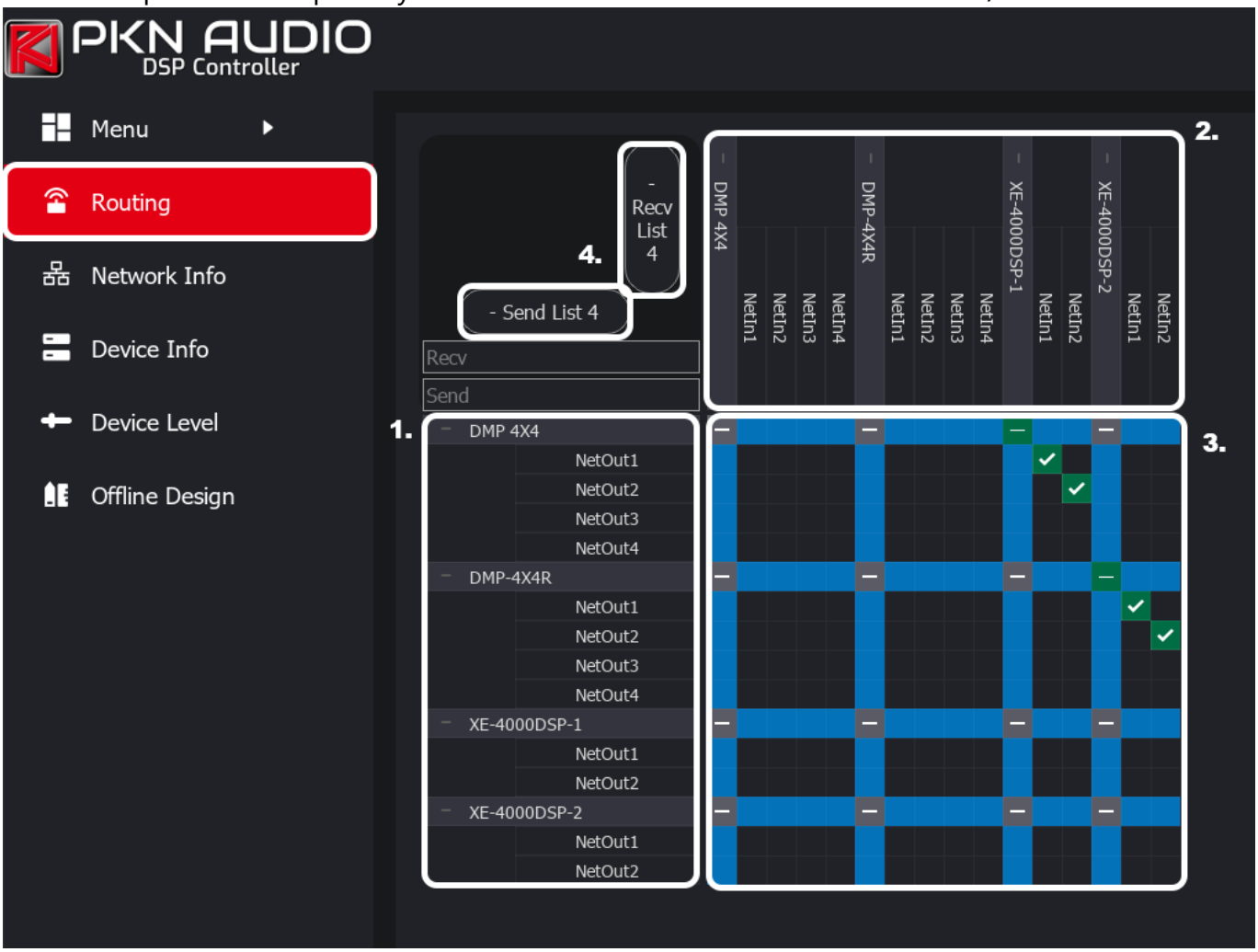


You can correct such issues in the Device Config menu.
(For more details see → *4.1 Device Config -DHCP Settings)*

3. Main View – Left Side Menus / Buttons

3.1 Routing

This view represents all connected device's digital audio in-and-outputs' matrix routing.
You can expand or collapse any row or column with the + and – or the Send/Recv List buttons:

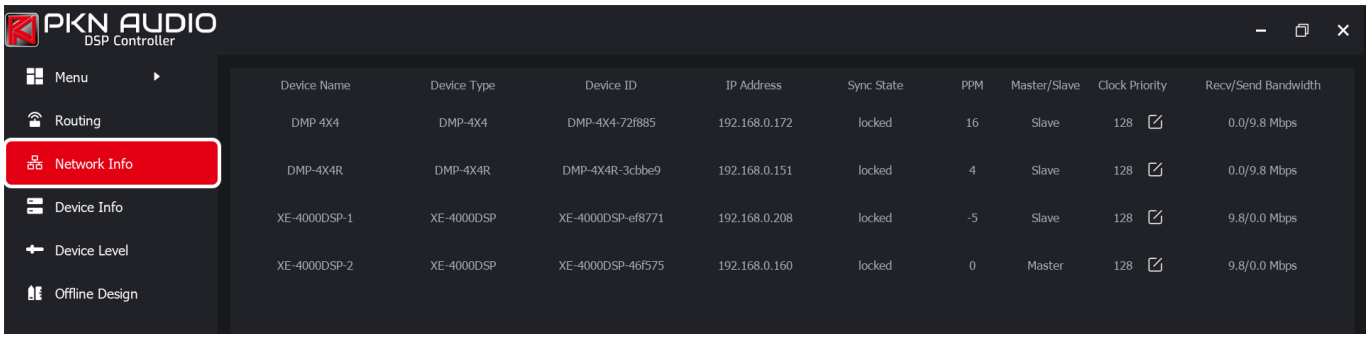


1. This is the sending device list. After expanding it, you can see the network sending channels of each device.
2. This is the receiving device list. After expanding it, you can see the network receiving channels of each device.
3. Here you can set up the digital audio routing. The checked state means that the network sending channel signal of the device corresponding to area 1 is sent to the network receiving channel of the device corresponding to area 2.
4. You can expand or collapse all Sending or Receiving devices list.

Note that there is also an internal routing of each device's digital (and analog) audio in/outputs as well.
(For more details see → 4.3.3 Matrix)

3.2 Network Info

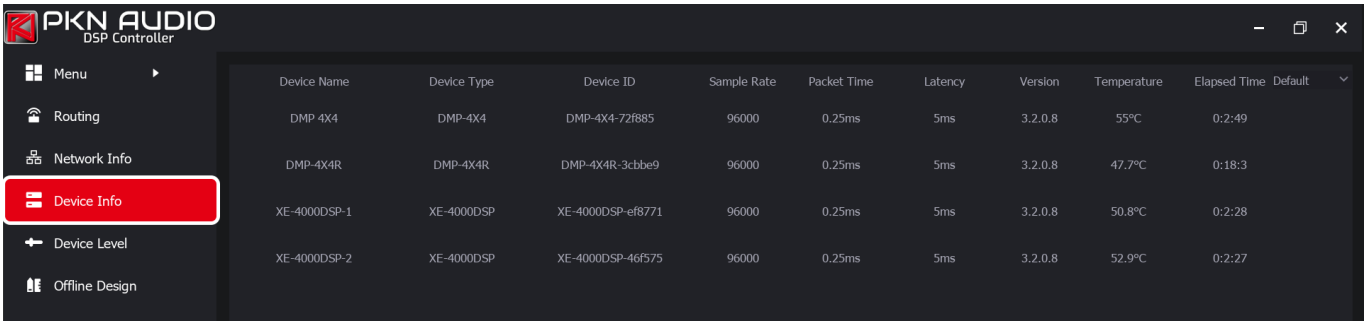
On the left panel, click `Network Info`, to see various network related information about the connected devices.



- `Device Name:` A modifiable display name of each device on the interface.
  You can modify it in the **`Device Details`** → `Device Config` menu: Double-click the device you wish to modify. The **`Device Details`** menu pops up in a new window and under Device Information, you can change the name to your liking.
  (See → 4.1 Device Config -Device Name)
- `Device Type:` Used to distinguish device types. Different types correspond to different products and functions. This is set by the manufacturer, and cannot be changed.
- `Device ID:` It represents the embedded DSP modules' unique ID, also not modifiable.
- `IP Address:` Currently used IP address allocated to the corresponding device. This can be set to either Manual or Automatic (DHCP) acquisition.
  Important to note, that if your computer's IP is set to manual, make sure that the connected devices are also manually set to the same domain, otherwise by default they might not obtain IPs from your computer's DHCP from the manually set IP range.
  This could cause various issues with the correct controlling of your devices!
  (For more details see → 4.1 Device Config -DHCP Settings)
- `Synchronous State:` Status of device synchronization. After any device is connected to the network, they will automatically synchronize with the Master device. The process can take up to 1 minute, the status is refreshed every 5-10 seconds.
- `PPM`: Description parameter for clock alignment of master and slave devices. Generally, it is within ±30.
- `Master/Slave`: Shows whether it is a master or a slave device. There is only one master device in a local area network. If there are other non-PKN Audio devices in the network using PTP clocks, they will also participate in the selection of master and slave devices together.
  (Meaning, if there is a non-PKN Audio device in the same local area network, and its clock priority is higher, all shown devices on the software may act as slaves.)
- `Clock Priority`: A modifiable parameter used to select master and slave devices. The range is 0-255 with the default value of 128. You can modify it with the `Edit` button next to the value.
  The system will automatically select the device with the highest priority (smallest value) in the network as the master device. You can manually specify a certain device to be the master (higher priority).
  (Virtual Sound Cards (VSC) cannot be masters. The clock priority cannot be modified, and is displayed as NA.)
- `Recv/Send Bandwidth:` Current network data transfer rate between devices (Received/Sent) shown in Mbps.

3.3 Device Information

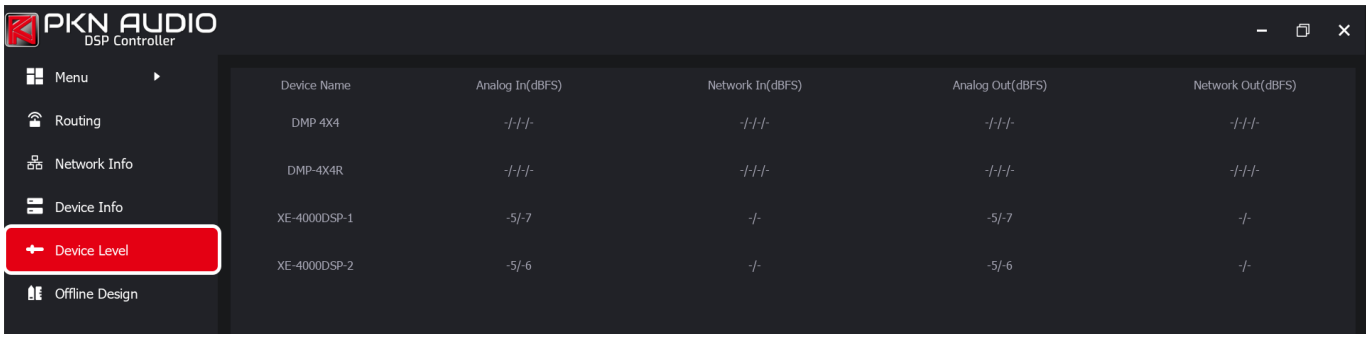
On the left panel, click `Device Info`, to see technical information of the connected devices:



- `Device Name, Device Type, Device ID`: All the same as previously.
  *(You can open the **`Device Details`** pop-up window of any device, from any view by double-clicking the device's name or row you wish to configure.)*
- `Sampling Rate:` The audio sample rate of the device.
- `Packet Time:` A setting parameter for network audio data packets.
  (Can be modified. See → 4.1 Device Config -Packet Time)
- `Latency:` A setting parameter for network audio transmission latency.
  (Can be modified. See → 4.1 Device Config -Latency)
- `Version:` Displays the firmware version of the embedded DSP module of the device.
  (The version of the Virtual Sound Card (VSC) is displayed in the VSC software. This option shows NA.)
- `Temperature:` The embedded DSP module's core processor's current temperature.
  During normal use, the temperature should be below 70°C and should not exceed 80°C.
- `Elapsed time:` The continuous running time of the device since last startup.

3.4 Device Level

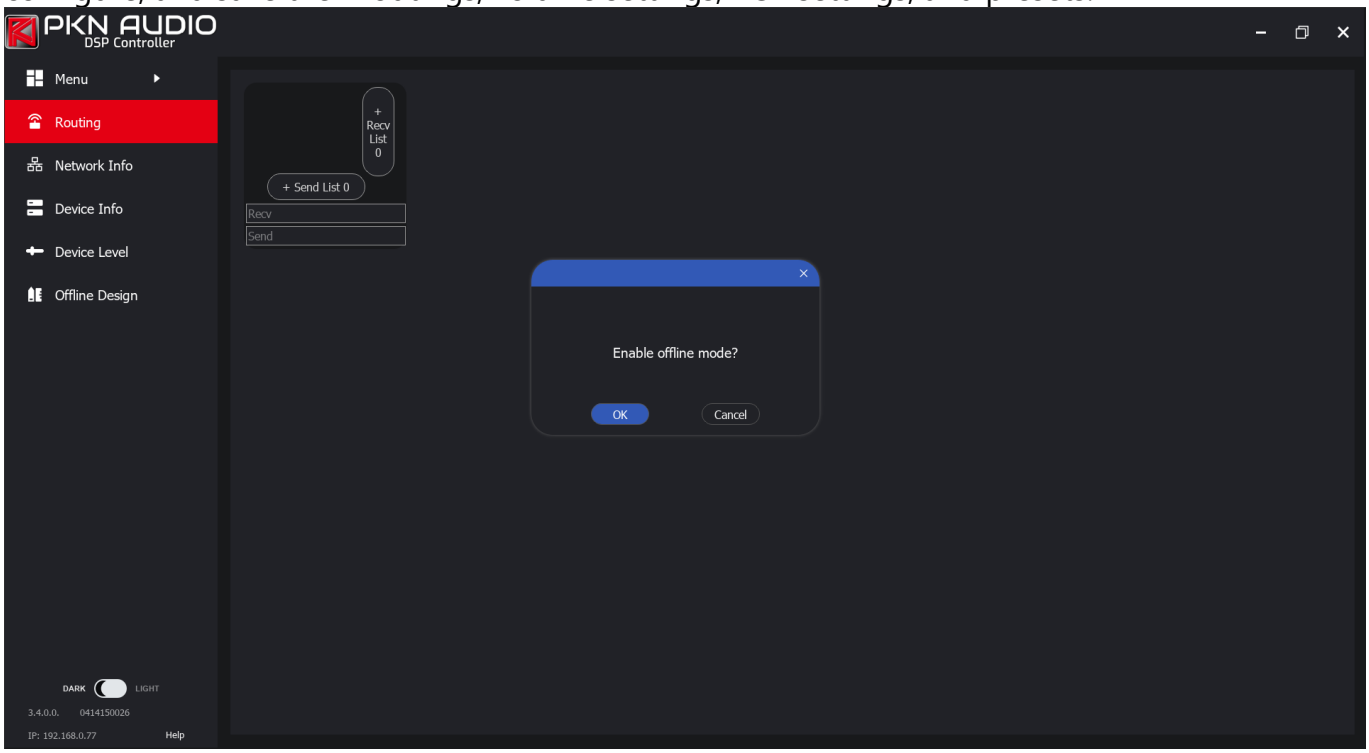
On the left panel, click `Device level` to quickly see the signal levels (in dBFS) of every connected device:



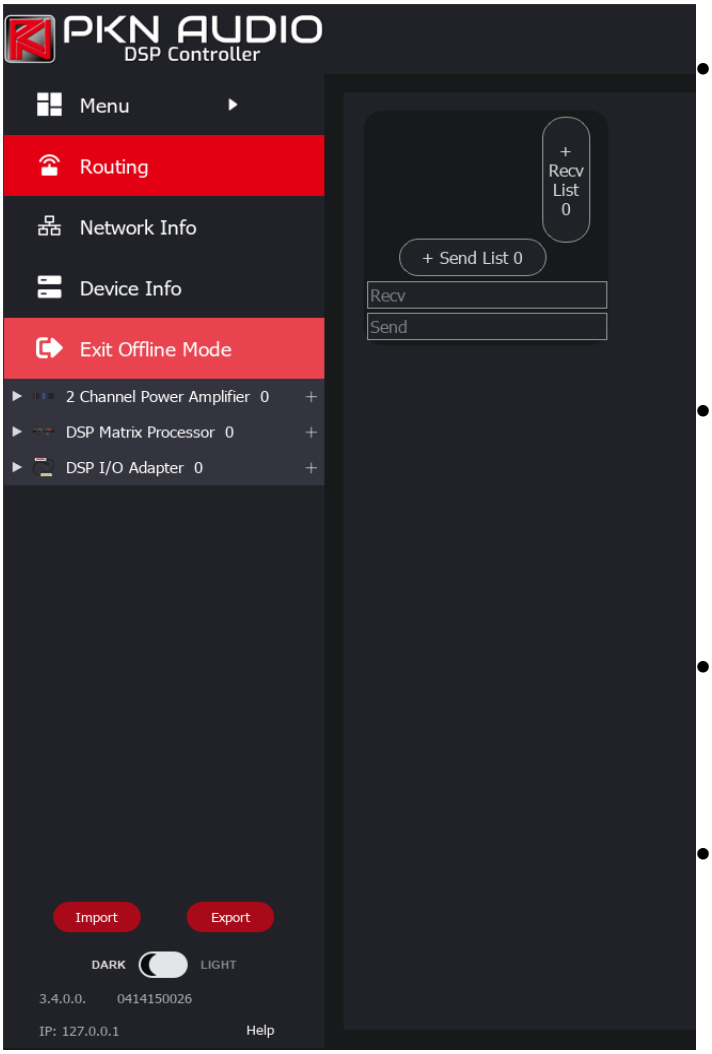
Each row contains the corresponding input and output levels of each connected device's respective channels divided by a "/". If there is no input/output "-/-" can be seen for each pair of channels.

3.5 Offline Design

The Offline Designer is a new feature, that enables a simulated local environment, in which you can add as many devices from our current DSP enabled portfolio as you'd like, and can pre-configure, and save their routings, volume settings, DSP settings, and presets.

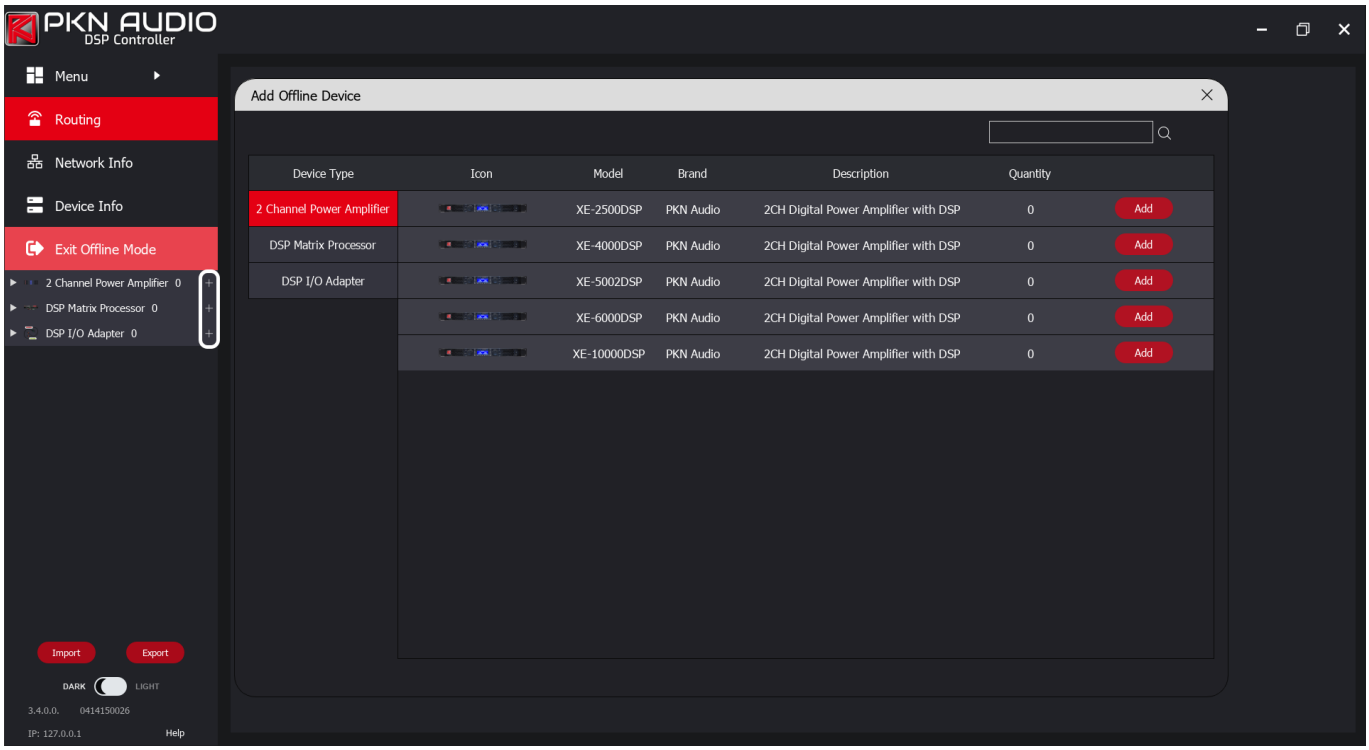


Clicking OK will open the Offline Designer view. If you opened it before, it will ask if you would like to recall the last saved session (OK), or start over from scratch (Cancel).



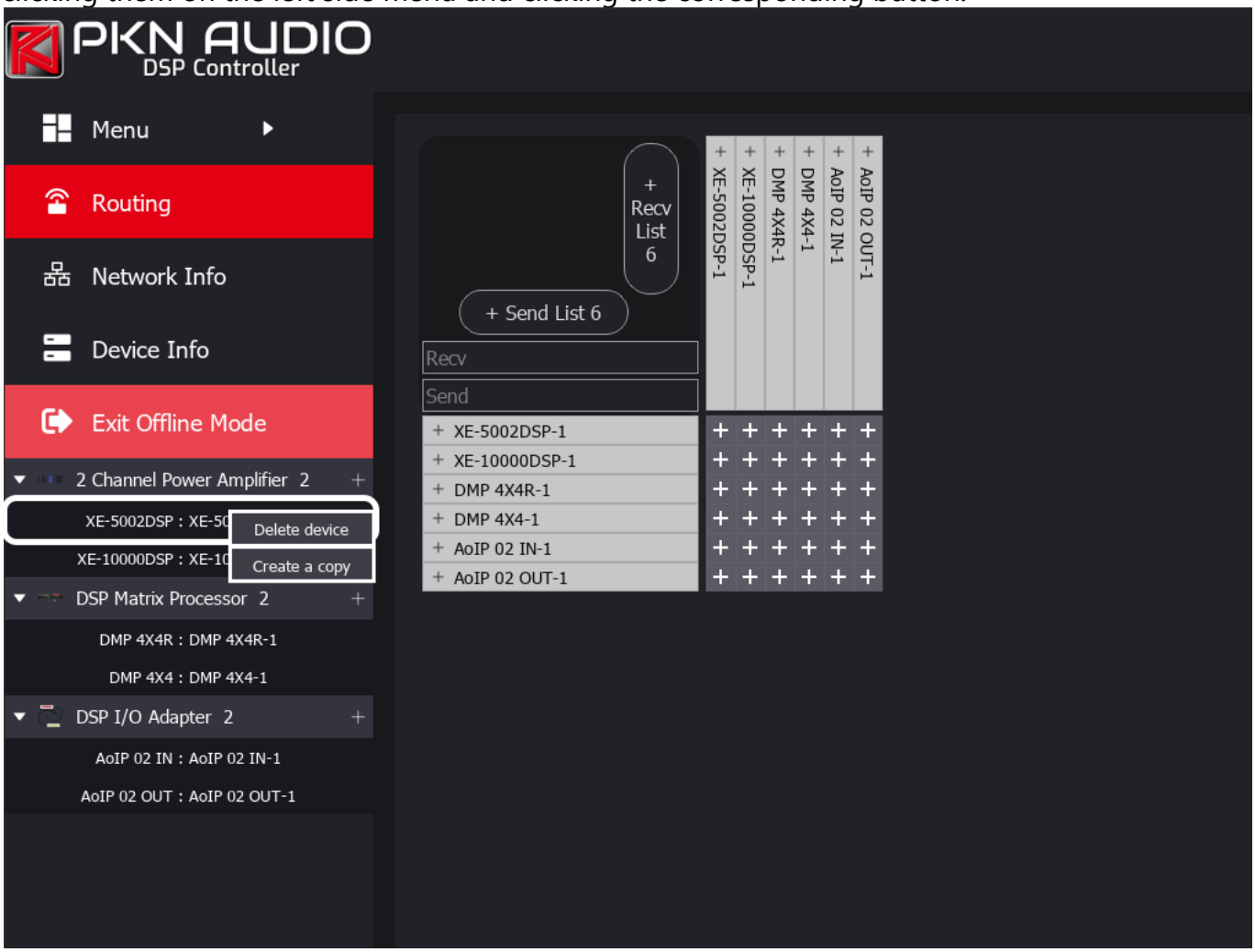
- Clicking the light red `Exit Offline Mode` button will go back to the default, online view. It will always ask if you would like to save the current session, so you can return to it later.
- All our DSP enabled product families will show up on the left side menu, from where you can manage the pool of offline devices you can configure.
- `Import` and `Export` buttons also show up on the bottom.
- Also note that your computer's IP shows as localhost in Offline Mode.

By clicking + next to the product types, the `Add Offline Device` window will pop up.

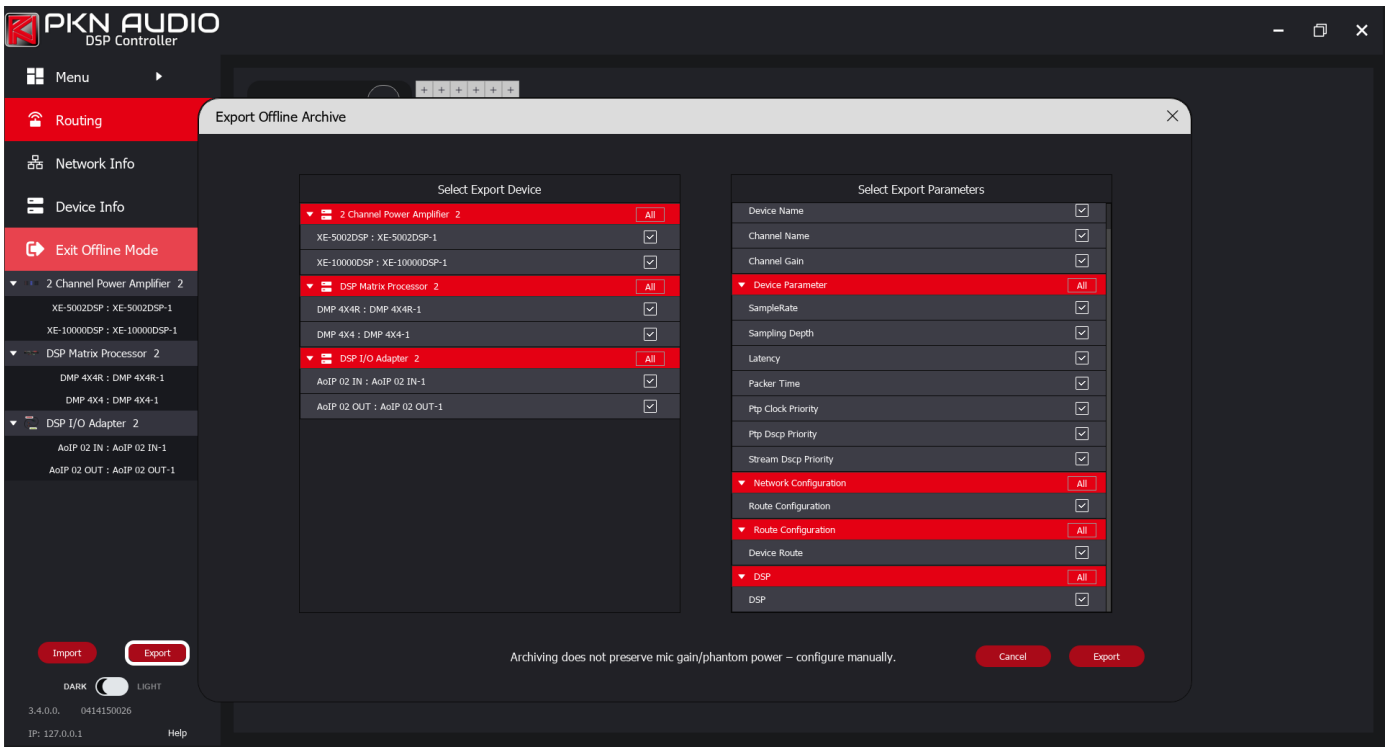


Here you can add any quantities of the devices you wish to simulate. Once you're done, you can close up the window and route/configure them as needed.

You can also quick delete or create a copy of any of the already added devices, by simply right clicking them on the left side menu and clicking the corresponding button:



You can also selectively export your modifications (by clicking `Export`), for backups, sharing them to another computer, to create templates before an install, or even share them with your fellow PKN Audio DSP Controller users:

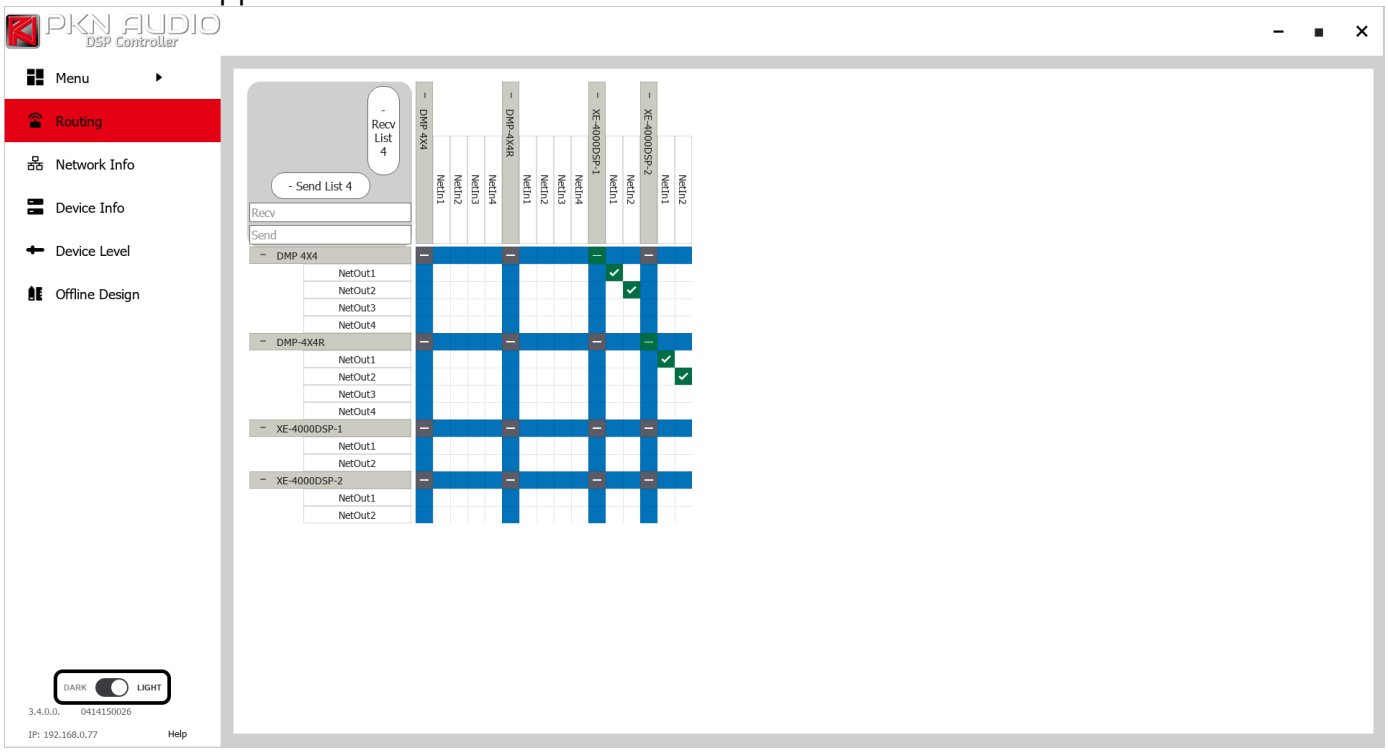


The `Import` button is of course to load the already exported data from a directory on your PC.

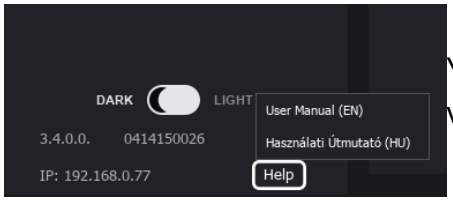
The following topic on the ***`Device Details`*** menu will explain the configuration of the devices also found in the **`Offline Design`** menu.
Please note that some features might be unavailable, compared to a live device.

3.6 Dark ↔ Light Theme Switch

For better indoor/outdoor visibility options, we added a Light Theme to the software. You can toggle between Default (Dark) and Light at any time from this button on the Main View, and the theme will be applied across all other views as well.



3.7 Help Button

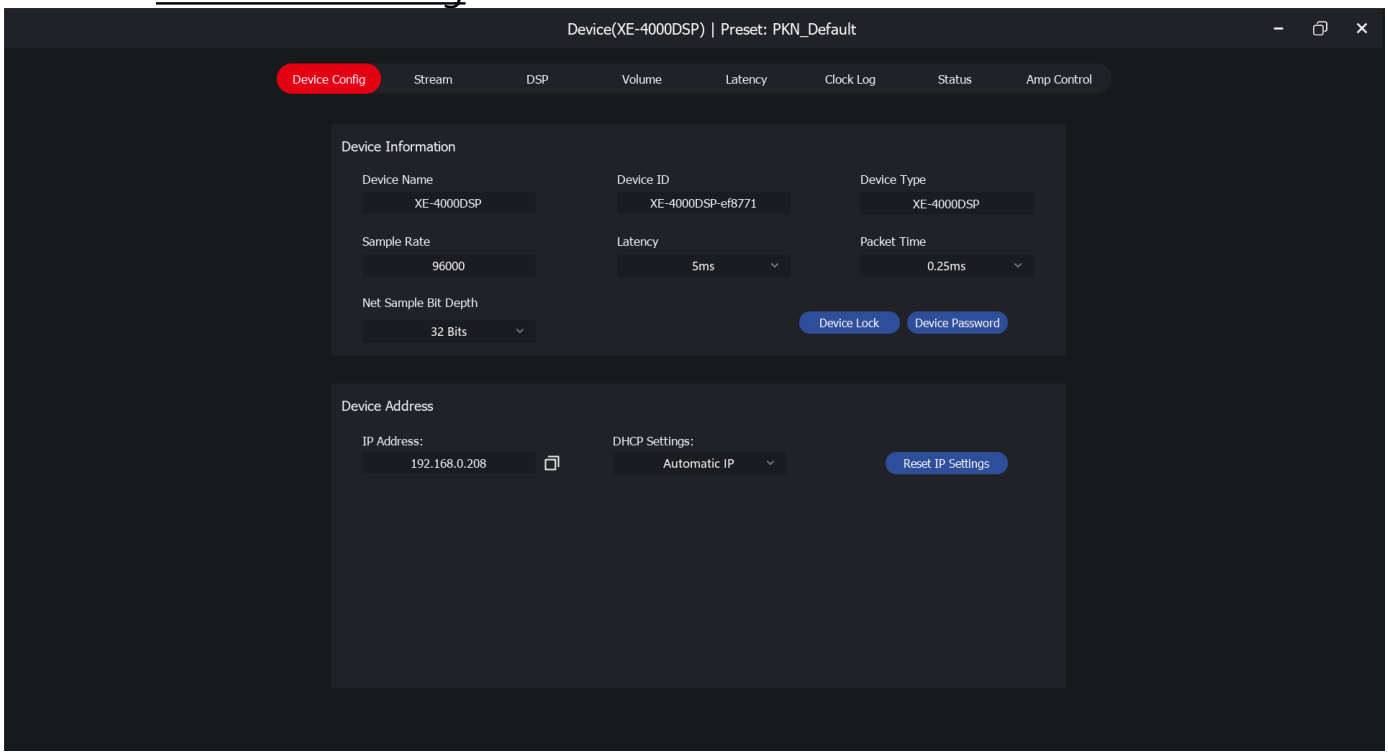


You can open the User Manual suitable for your current software version in English or Hungarian.

4. Device Details

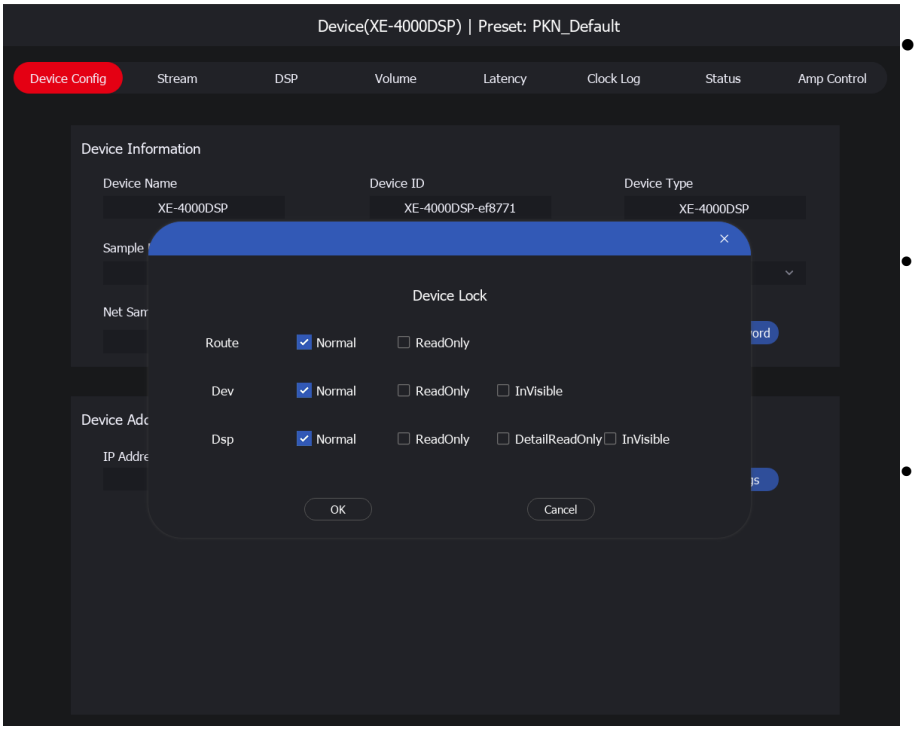
On any `Routing`, `Network Info`, `Device Info`, or `Device Level` menu **double-click** the name or row of the device you wish to configure.
This will pop up the **`Device Details`** in a new window, by default on `Device Config` tab.

4.1 Device Config



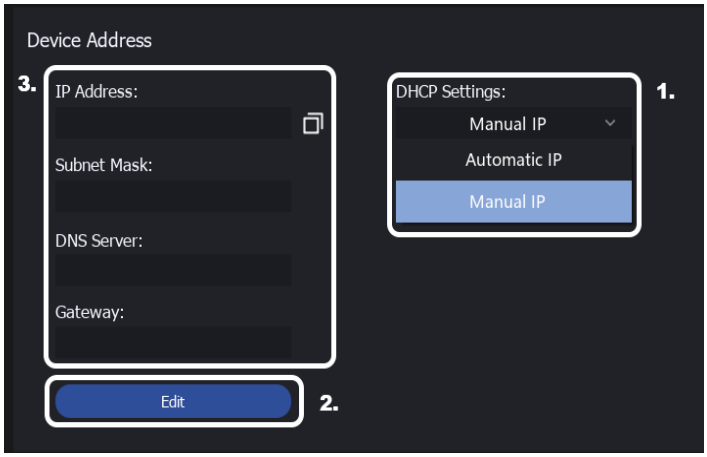
- `Device Name`: By default, it is always the device type, but you can rename it to your liking by clicking in the text box.
  Please note that the name cannot exceed 30 characters. Once done, you can either hit enter, or click anywhere out of the box to apply.
- `Device ID`: The unique ID for identifying the device (cannot be modified).
- `Device Type`: Used to distinguish the device types and series (cannot be modified).
- `Sample Rate`: Currently set audio sampling rate. Some devices support quick switching between 96000 and 48000kHz.
  (For XE-DSP Amplifiers quick switching is currently unsupported. Factory default is 96000. Please contact us if you require 48000 exclusively.)
- `Latency`: Network latency parameter set in ms.

- `Packet Time:` A parameter for the network data packets' transfer rate. The network latency can not be less than 4 times the packet time.
  (Meaning, if the packet time is set to 0.25ms, the network latency can be set to a minimum of 1ms. If the packet time is set to 1ms, the network latency can be set to a minimum of 5ms. If you want to set the network latency to 1ms, you must first change the packet time to 0.25ms. Changing the packet time requires rebooting of the DSP module.)
- `Net Sample Bit Depth:` Select sampling bit depth between 16/24/32.
- `Device Lock / Device Password:` When clicking either buttons first, you'll have to set a password, confirm and save it. You can change this password afterwards with the `Device Password` button, given you provide the previously set password.
  After clicking Device Lock you can select various properties of the device to be inaccessible for unauthorized users:



- **Route:** Set whether the Main View's Routing feature is modifiable (`Normal`) or `ReadOnly` for the device's digital inputs.
- **Dev:** Set the Device Details main tabs' contents to be modifiable (`Normal`), `ReadOnly` or completely `InVisible`.
- **Dsp:** Set the DSP and its submenus to be modifiable (`Normal`), the DSP Chain visible without function details (`ReadOnly`), with function details (`DetailReadOnly`) or completely `InVisible`.

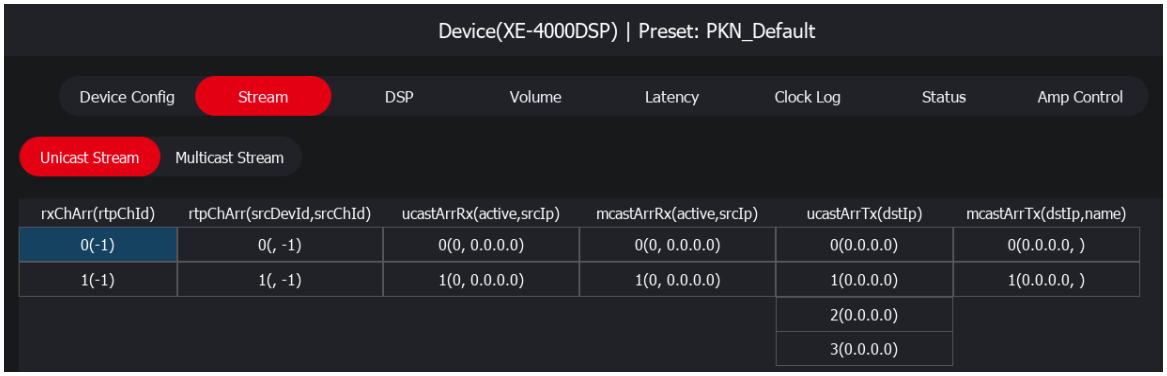
- `IP Address:` Displays the current IP address of the device.
- `Reset IP Settings:` If the IP address is set manually, and incorrect parameters are given, it could cause instability in the software being unable to obtain/view some device information. In this case, the IP setting can be reset to the default automatic acquisition with this button.



- `DHCP Settings:`
  Automatic or Manual acquisition method selection of IP addresses. For manual setting, select `Manual IP` ***(1.)***, then click `Edit` ***(2.)***, and enter the specified values in the corresponding input boxes ***(3.)*** and hit enter. Ensure that the IP address, subnet mask, DNS server address and gateway values are correct, otherwise, it may not be possible to view/obtain device Information correctly.

4.2 Stream

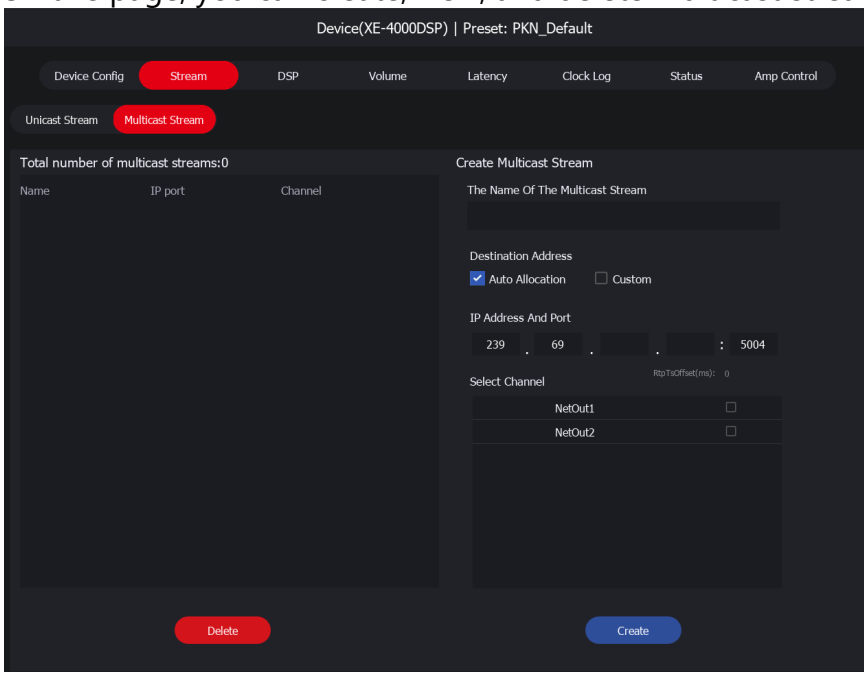
4.2.1 Unicast Stream



- `rxChArr(rtpChId)`: The routing records of the receiving channels. The first number represents the channel ID (0 represents channel 1). The content in parentheses represents the subscript of rtpChArr, corresponding to the index of the rtcChArr(srcDevId, srcChId) list. -1 means there is no routing.
- `rtpChArr(srcDevId, srcChId)`:srcDevId corresponds to the ID of the sending device. srcChId corresponds to the channel index of the sending device. (0 represents channel 1)
- `ucastArrRx(active, srcIp)`:It represents the array of unicast received data streams. 0 means inactive, and 1 means active. Only in the active state can audio streams be transmitted. "`srcIp`" represents the IP address of the sending device.
- `mcastArrRx(active, srcIp)`: Represents the array of multicast received data streams. 0 means inactive, and 1 means active. Only in the active state can audio streams be transmitted. "`srcIp`" represents the IP address of the sending device.
- `ucastArrTx(dstIp)`: Represents the array of sending channels that have currently established connections (the array index has nothing to do with the channel). "`dstIp`" represents the IP address of the receiving device.
- `mcastArrTx(dstIp, name)`: The array of multicast sending streams that have been created. "`dstIp`" represents the multicast stream address. "name" represents the multicast stream name.

4.2.2 Multicast Stream

On this page, you can create, view, and delete multicast streams.



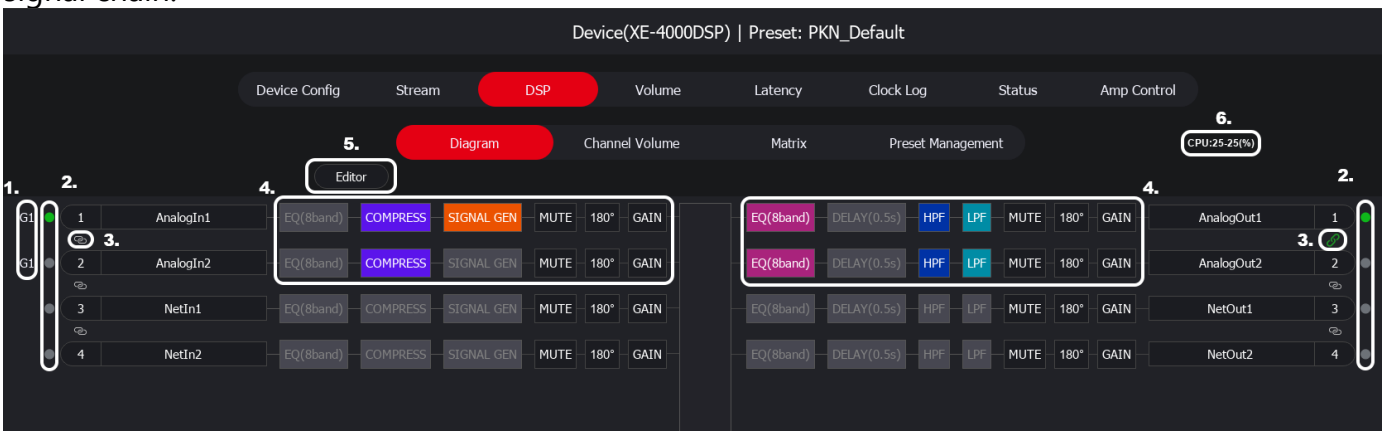
- `The Name Of The Multicast Stream`: The name displayed after creation. It should not exceed 32 characters.
- `Destination Address`: The address of the multicast stream. If there are no special requirements, automatic allocation can be selected. You can also customize the address.
- `Select Channel`: Selection of digital channels that can be sent out using the multicast stream.
- `Create`: Create the multicast stream with the parameters you set above.
- `Delete`: Delete a previously created multicast stream.

4.3 DSP

This tab is dedicated to all digital signal processing features on every analog and digital channels of your device. There are 4 sub menus regarding the DSP:

4.3.1 Diagram

On this page, you can easily enable, bypass or modify the functions of the DSP input and output signal chain.



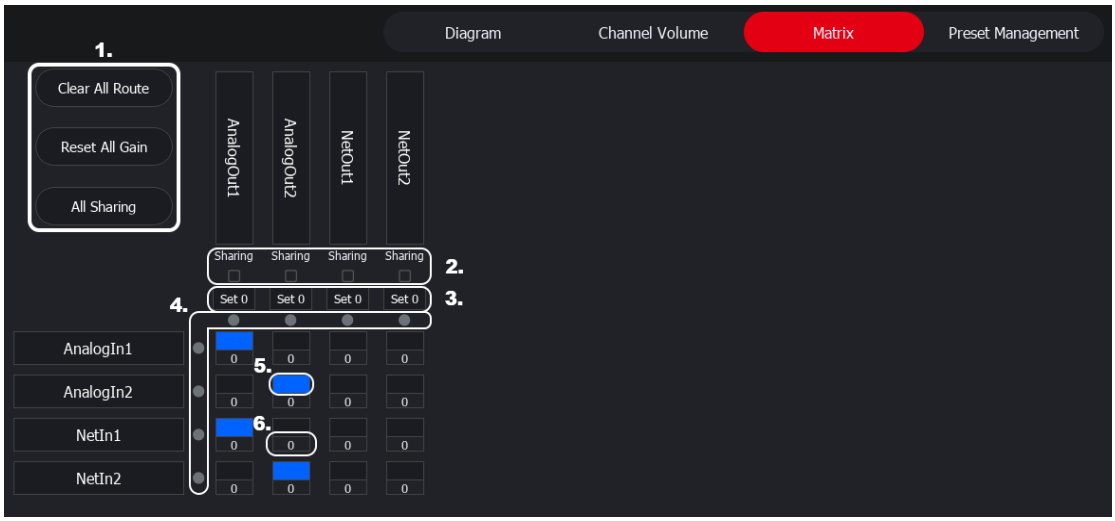
1. Channels that are grouped together are displayed here. You can right-click the names of any channel and choose which groups to allocate them to. Also by left-clicking the channel names, you can modify them.
   (Currently this feature is only implemented within one device, meaning you cannot group channels from different devices together yet.)
2. A basic signal presence indicator, where green means there is signal on that channel, gray means there's none, or it's below -60dB.
3. You can link channels with this button together. This way if you enable/disable, or modify a parameter on a DSP function, it will automatically change on the linked channel as well. This is only possible if the DSP functions are the same on both channels, above and below this icon.
4. This area represents what DSP functions are present on a channel.
   Bright color indicates that the function is enabled (not bypassed), and gray indicates that the function is disabled (bypassed).
   You can quick-toggle between enabled and disabled by right-clicking and selecting `Set/Cancel Bypass` on the corresponding function box. Also, on the right-click context menu, you can copy and paste function parameters between the same type of DSP functions. The `MUTE`, `180°` (Invert Phase), and `GAIN` functions are fixed for every channel and are always found on the end of the chain.
   Clicking mute and 180° toggles the feature, clicking the gain box shows its adjustment view.
   Double-clicking any other function box will pop up the detailed parameter settings view.
   (Multiple detailed parameter setting pages can be opened at the same time, and they stay always on top between the DSP submenus (`Diagram`, `Channel Volume`, `Matrix` and `Preset Management`). You can also freely move the windows around, or enable/disable the functions, with the `Bypass` buttons.)
   (For each module's Detailed view see → 5.3 Detailed Explanation of DSP Functions)
5. You can modify the DSP chain with the `Editor` button.
   (For more details see → 5.2 DSP Editor View)
6. Current CPU usage is shown here. Some DSP functions use more resources then others, but generally using more features on more channels, means higher CPU usage and higher temperatures. Please make sure CPU usage does not exceed 85% for stable use.

4.3.2 Channel Volume



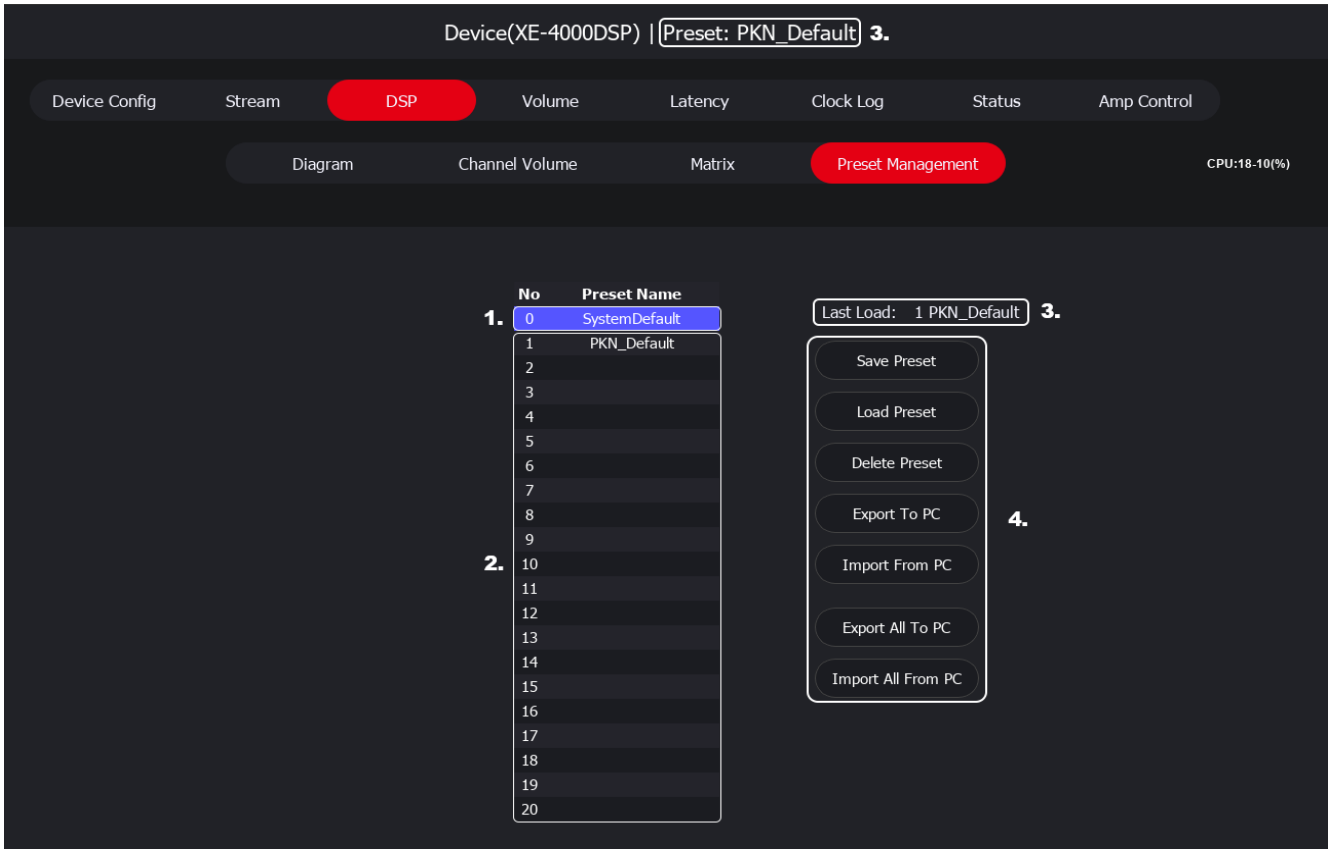
1. Input, output, and group view switch.
2. Quick toggle mute on/off for all input and output channels.
3. Channel identificators and custom names.
4. Real time level indicator.
5. Channel gain, mute, and phase inverse controls. You can set level values by dragging the level meter or typing in the blue box. The blue 0dB button instantly resets the levels to 0dB.

4.3.3 Matrix



1. Shortcut functions for one-click setting parameters for all channels.
2. If the `Sharing` box is checked, the gain sharing method will be used when multiple input channels are mixed to a single output channel.
3. Gain reset button, that sets the gain of all input channels for the corresponding output channel to 0dB.
4. A basic signal presence indicator, where green means there is signal on that channel, gray means there's none, or it's below -60dB.
5. Indicating if the corresponding input channel is mixed to the corresponding output channel. (in this case, and by default AnalogIn1 is mixed to AnalogOut1, AnalogIn2 to AnalogOut2 etc.)
6. Shows the gain of the corresponding input channel mixed to the corresponding output channel. You can click it to open an adjustment window.

4.3.4 Preset Management

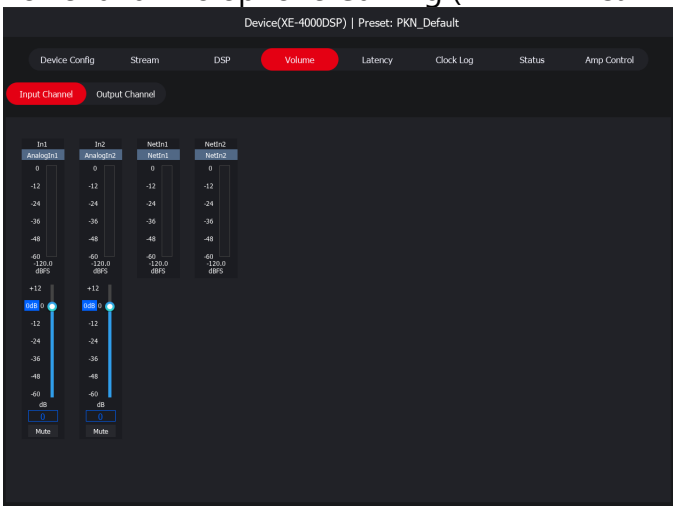


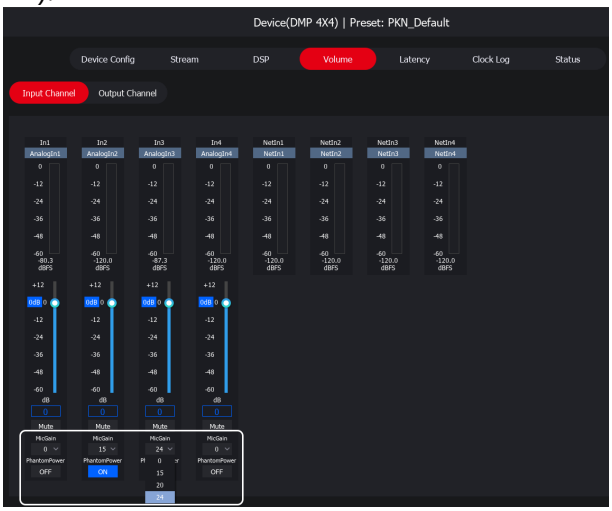
1. The No. 0 preset contains the system default parameters, which clears every user setting, including the DSP chain, volume and matrix settings. This preset cannot be modified, overwritten, or deleted.
2. From No. 1 to 20 are the user customizable preset slots. No.1 contains our factory PKN_Default preset, which has the matrix set up for plug-n-play operation.
3. The last loaded preset is always shown on the window header and in Preset Management.
4. This is the management function area. Here you can:
    - `Save Preset`: Select the corresponding row in the list to save current parameters to. You will have to name your preset at this point. If you selected an already existing preset, clicking OK after naming your preset will overwrite it.
    - `Load Preset`: Select the corresponding preset from the list to load it's saved parameters.
    - `Delete Preset`: Delete the selected preset from the list.
    - `ExportToPC`: Export the selected preset to a file, stored on your PC.
    - `ImportFromPC`: Select the corresponding row in the list to import a previously exported file into the device. If you select an already existing slot, there is a popup to confirm if you would like to overwrite it with the imported settings, or not.
    - `ExportAllToPC`: Bulk export all 20 custom presets to a file, stored on your PC.
    - `ImportAllFromPC`: Bulk import all 20 custom presets from a previously exported file into the device. Note that this will overwrite every preset that is present in device memory, after confirmation!

(Note that after loading a preset and manually adjusting any parameters, the changes are kept, even after a reboot, but are NOT saved automatically to the last loaded preset.
Meaning, if preset No. 2 is manually loaded and any parameters are modified, the device is shutdown and restarted, the modified parameters will still be in effect, but if preset No. 2 is loaded again, the modified parameters will be lost.
If you want to save the modified parameters to a preset, be sure to manually click `Save Preset` overwriting it to save your changes that can be loaded again.)

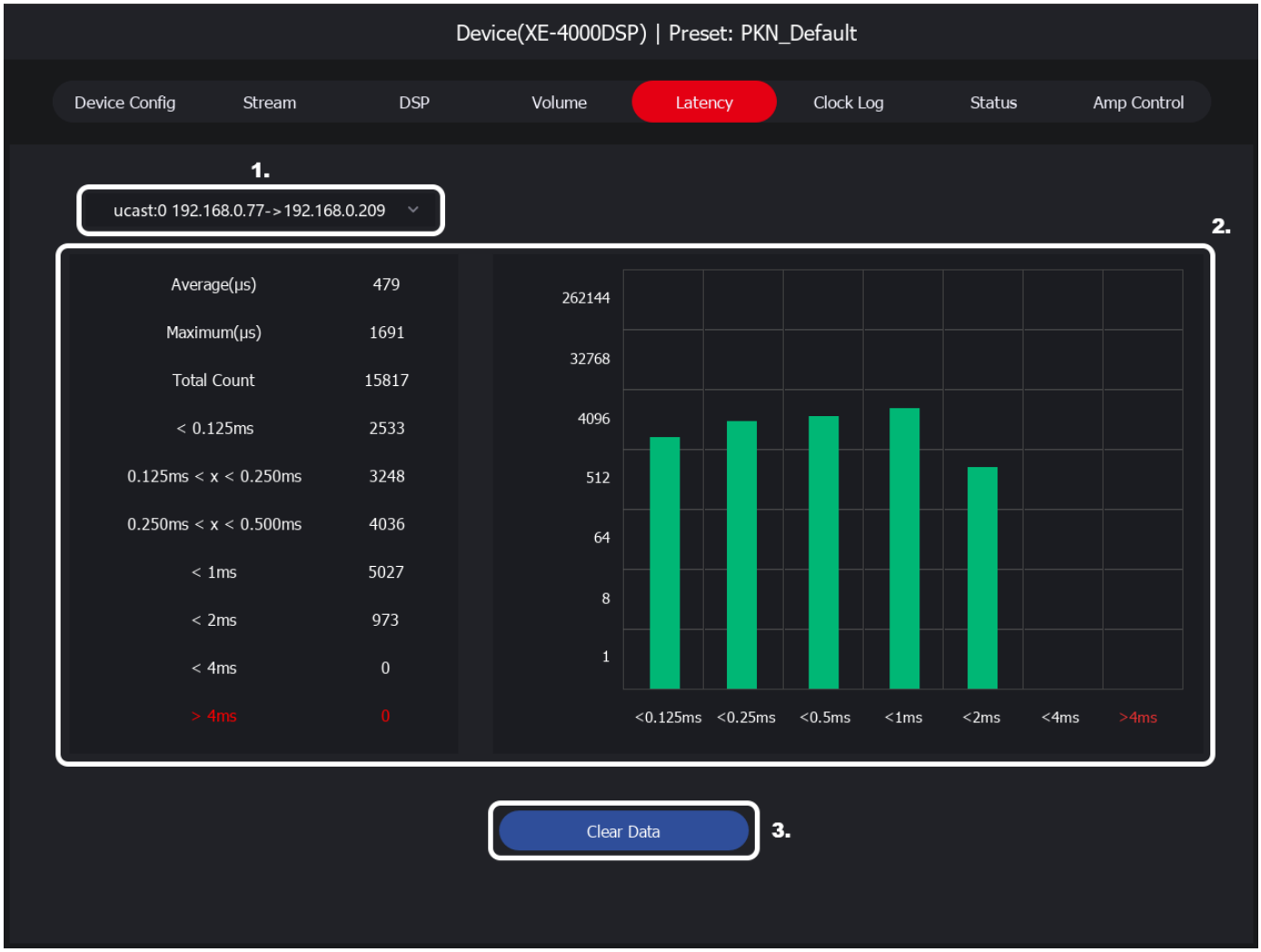
4.4 Volume

Not to be confused with 4.3.2 ChannelVolume, this is a simple, independent volume controlling and monitoring surface, with some additional features for devices that support +48V Phantom Power and Microphone Gaining (DMP 4X4 & DMP 4X4R).





4.5 Latency



This view shows packet times and latency statistics of the corresponding received streams.

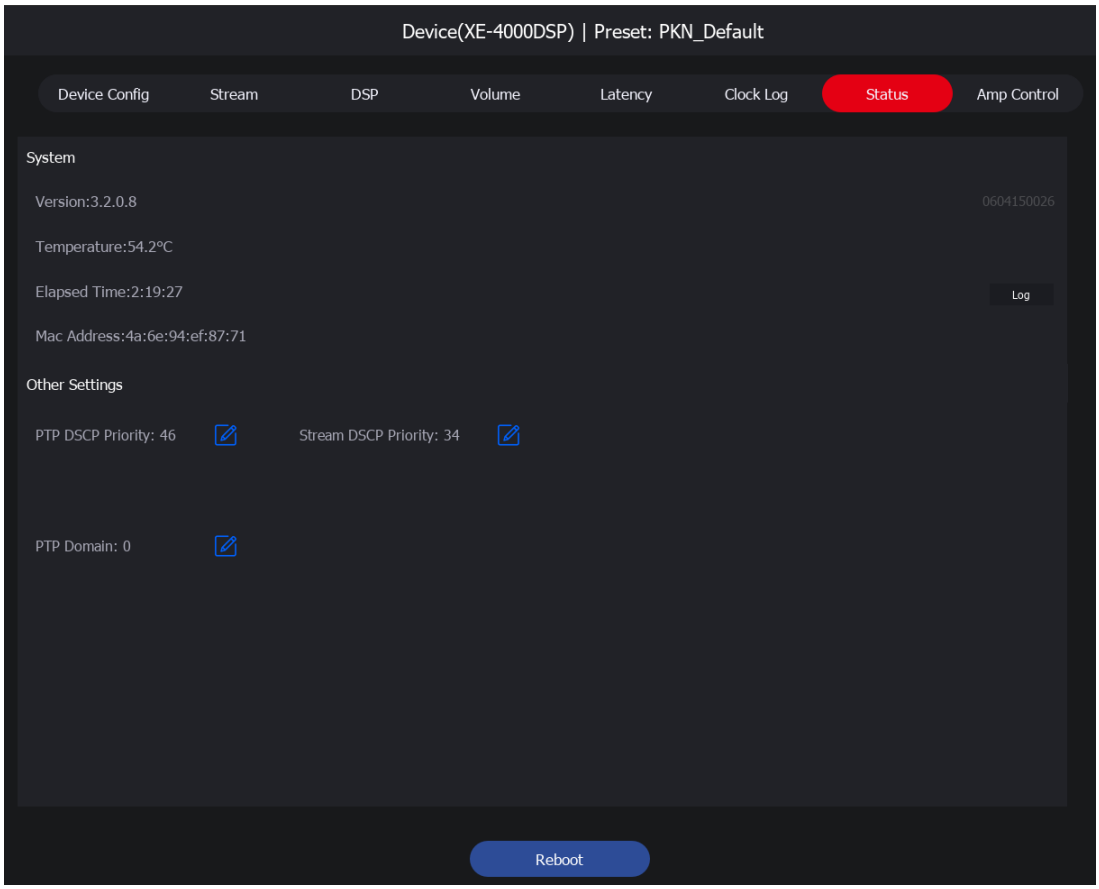
1. If audio data is received from multiple sources, you can select the stream you wish to monitor here. If there is only one, it will be selected automatically.
2. This area represents statistical packet latency information.
   Please note that before the device is powered on and synchronized, the received data can appear as >4ms (red bar). This is normal behavior.
   Right after powering-on, the clock is not yet synchronized. After the power-on state is complete, and the clocks are synchronized, the red bar will no longer increase.
3. This button can clear all currently logged data, so you can perform statistics measurement again from zero.
   Generally, if there are occasional over 4ms late data packets, and the red bar appears, it is due to fluctuations in the network environment, and does not affect sound transmission.
   If there is a continuous increase in red data at intervals, or audible errors in the sound transmission, it indicates abnormal network or device data transmission, and troubleshooting is required. It's recommended to check the `Packet Time` and `Latency` values in *4.1 Device Config*.

4.6 Clock Log

This panel is used for device debugging. If you encounter issues with digital audio streaming, such as audio stuttering, a device abruptly going offline, or the clock is unable to synchronize, please export the complete Clock Log and provide it to us for analysis.

4.7 Status

This page displays the DSP core's firmware version information, temperature, running time, and MAC address.



Under `Other Settings`, with the corresponding edit button , you can set PTP DSCP Priority, Stream DSCP Priority and PTP Domain values.

Also you can `Reboot` the embedded DSP module, without powering the device

4.8 AmpControl (XE-DSP Specific)

AmpControl is a major addition in this software version, intended to give better remote control over our XE-DSP series amplifiers, similar to the original XE series' webserver and Amp Control Software solutions.

From here you can monitor and control various amplifier specific settings remotely:



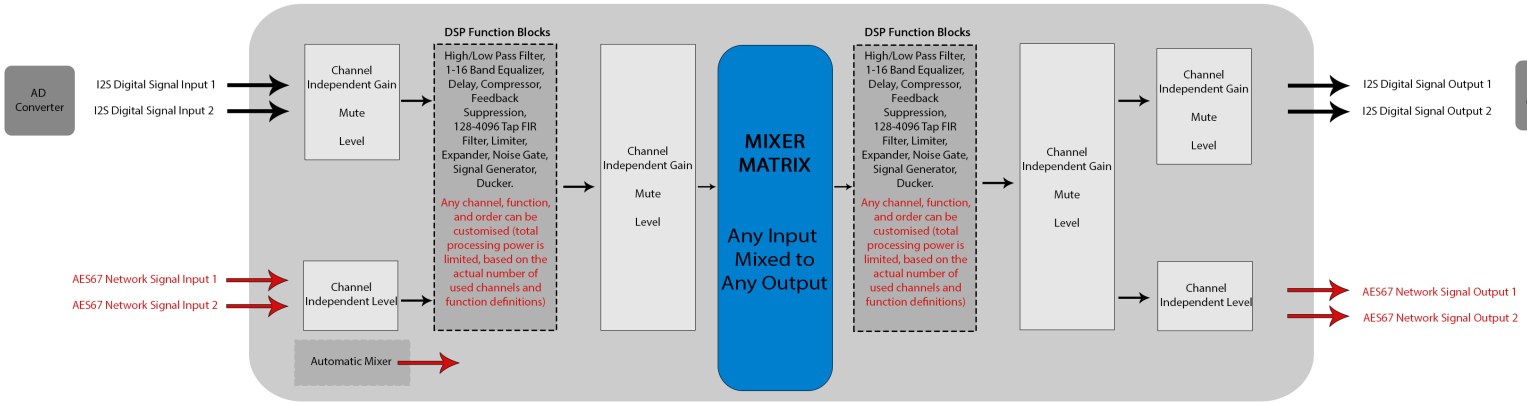
1. **Temperature:** Taking measurements from the amplifier's endstage and power supply heatsinks, always displaying the highest value.
   The indicator can have 4 statuses: Normal – White | Below 75°C
   Stage I – Light-Orange | 75-80°C
   Stage II – Dark-Orange | 80-90°C
   OVERHEAT – Red | Above 90°C
   (For more information about temp stages, see the XE-DSP Series User Manual)
2. **Volume and Mute control:** You can use the slider or the blue box below it to set the amplifier's volume (in 0.5 increments), and toggle channel muting. These features are linkable **(4.)**
3. **Output Voltage and Current:** Live measurements taken from the endstage (0.5s intervals).
4. **Link (Ganging) Channels:** If enabled, the link button turns green and both channels outlines are highlighted in light blue. When channels are ganged Volume, Muting **(2.)** and Limiter **(8.)** values are synced for both channels.
5. **Status indicators:** ***Ready***: The amplifier is operational
   MUTED: Channel Mute is engaged
   SHORT: Output Short Protection is active
   DC FAULT: DC Fault Protection is active, endstage is disabled
   OVERHEAT: Temperature levels above 90°C, endstage is disabled
   (For more information about fault states, see the XE-DSP Series User Manual)

6. **Actual Impedance & Power:** Reference calculations from the measured peak voltage and current values averaged from a 10 second interval.
7. **Input/Output AGR:** Indicators of Automatic Gain Reduction in effect on the amplifier's analog in and outputs. Output AGR can be engaged by different temp states, and limiter settings.
   (For more information about AGR, see the XE-DSP Series User Manual)
8. **Limiter Settings:** You can switch between two setting methods:
   - *Limiter Wizard:* You can set the impedance in Ohms and desired maximum power in watts for a calculated maximum voltage. Modifying either values will make the "Set" button blink in orange for 30 seconds. If it is clicked under this time, the desired limiter value will be set on the corresponding channel, or if 30 seconds pass, the blinking stops, and the values will reset to their previous state.

   - *Advanced Limiter:* You can manually set the limiter voltage.
   (Note, that Advanced Limiter instantly sets the limiter once the value is given, but if you switch to Limiter Wizard, you still have to click "Set". Until then the Advanced Limiter value is in effect.)

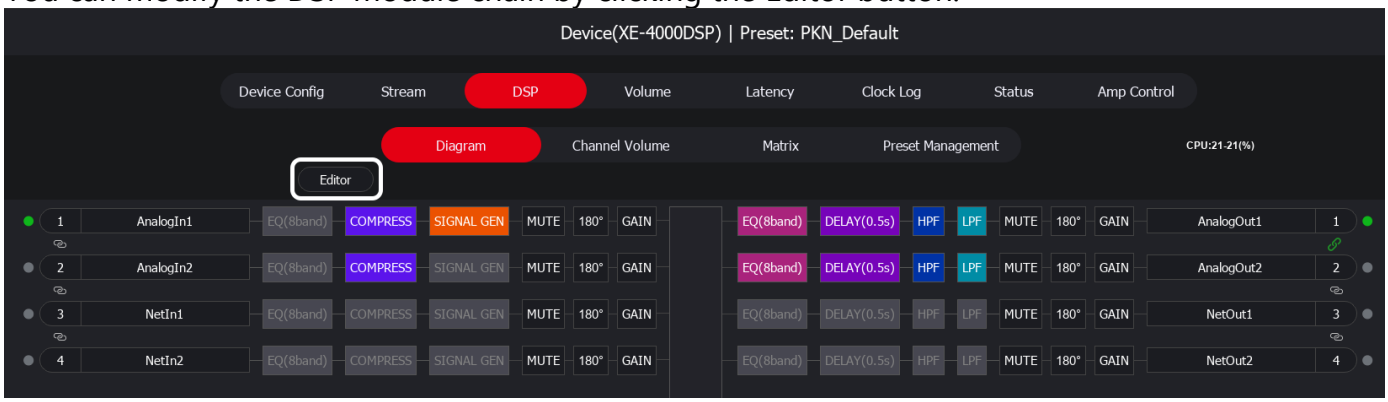
5. Advanced DSP Features

5.1 DSP Signal Flow Chart



5.2 DSP Editor View

You can modify the DSP module chain by clicking the Editor button:



This will open the chain modification view:



1. You can add an extra Auto Mixer channel to the input channels with this button.
2. Here you can save or cancel the changes made since you clicked the Editor.
3. This shows the predicted maximum CPU usage assessment. The combination of DSP functions should not exceed the maximum CPU occupancy rate.

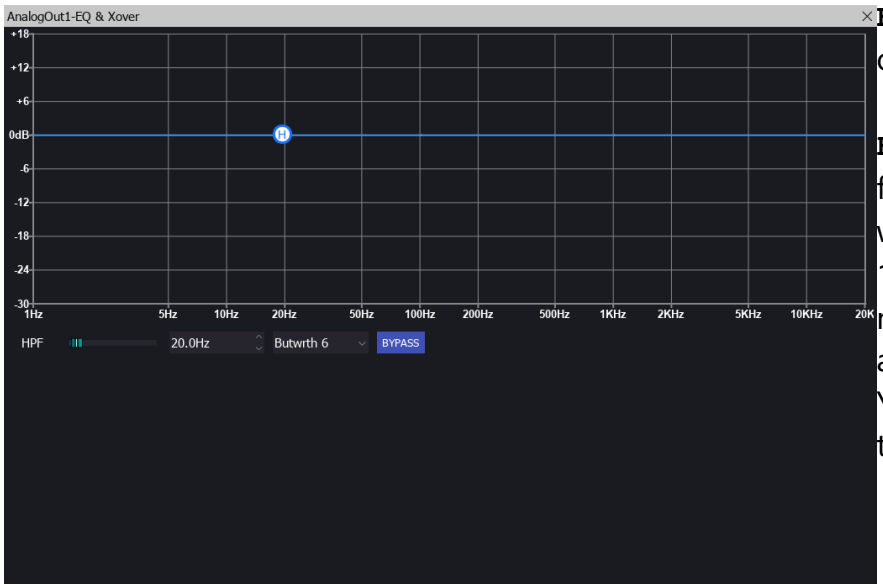
   However, you can disable the max CPU rate limitation with the `ForceAdd` checkbox. If it's absolutely necessary, you can try adding more than the recommended amount of features to the signal chain, but this requires constant monitoring of the current CPU usage **(4.)** and temperature levels (under Status or Device Info).
   (CPU usage should not exceed 85%, and DSP core temperature should never be above 80°C)

   WE DO NOT TAKE ANY RESPONSIBILITY IF THERE IS ANY DAMAGE OR THE DEVICE BECOMES FAULTY DUE TO OVERLOADING THE CPU!
4. Current CPU usage.
5. Click the + button to add a DSP function to the corresponding place in the chain. You can choose any from the list **(6.)** with limitations for equalization, high pass, low pass and delays. From these only one can be set for each channel, and they share the same edit window.
   As for the delay function, all channels share a delay upper limit, depending on your latency settings.
6. The list of DSP Functions that can be added to the chain.

5.3 Detailed Explanation of DSP Functions

These are the modules you can add to any "+" marked places of the signal chain from the Editor menu:

5.3.1 ***HPF*** *– High Pass Filter*



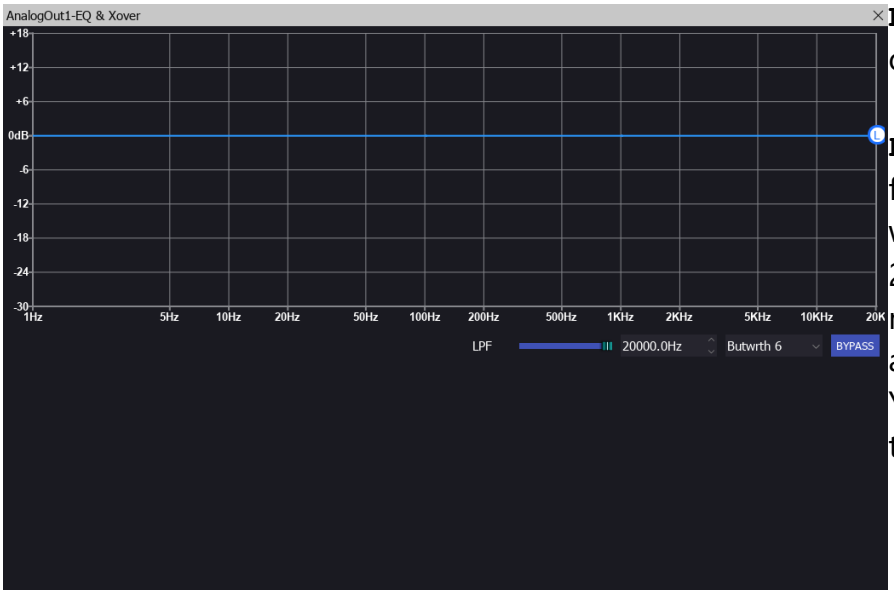
`BYPASS:` Toggle Bypass / Enable of the function.

`HPF:` You can adjust the cutting frequency of the high pass filter with the **`slider`** between 19.7-20000Hz or typing the value manually up to one decimal accuracy.
You can also drag the Ⓗ to adjust the frequency by hand.

`Butwrth 6:` Filter characteristics selector:

The types of available filters are the following:
Butterworth (Butwrth): 6/12/24/36/48 dB/oct
Bessel: 12/24/36/48 dB/oct
Linkwitz-Riley (Lnk-Ril): 12/24/36/48 dB/oct

5.3.2 ***LPF*** *- Low Pass Filter*



`BYPASS:` Toggle Bypass / Enable of the function.

`HPF:` You can adjust the cutting frequency of the low pass filter with the **`slider`** between 19.7-20000Hz or typing the value manually up to one decimal accuracy.
You can also drag the Ⓛ to adjust the frequency by hand.

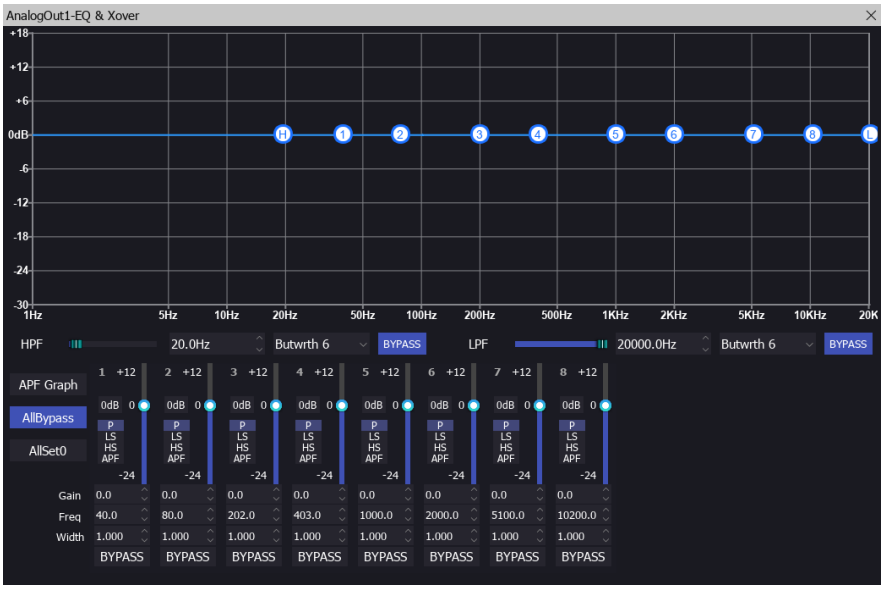
`Butwrth 6:` Filter characteristics selector:

The types of available filters are the following:
Butterworth (Butwrth): 6/12/24/36/48 dB/oct
Bessel: 12/24/36/48 dB/oct
Linkwitz-Riley (Lnk-Ril): 12/24/36/48 dB/oct

Note that if both High and Low Pass Filter is present on a channel, they share the same view,

along with the Equalizer. Also you can't add more than one LPF, HPF, or EQ to each channel.

*5.3.3 **EQ** - Parametric Equalizer*

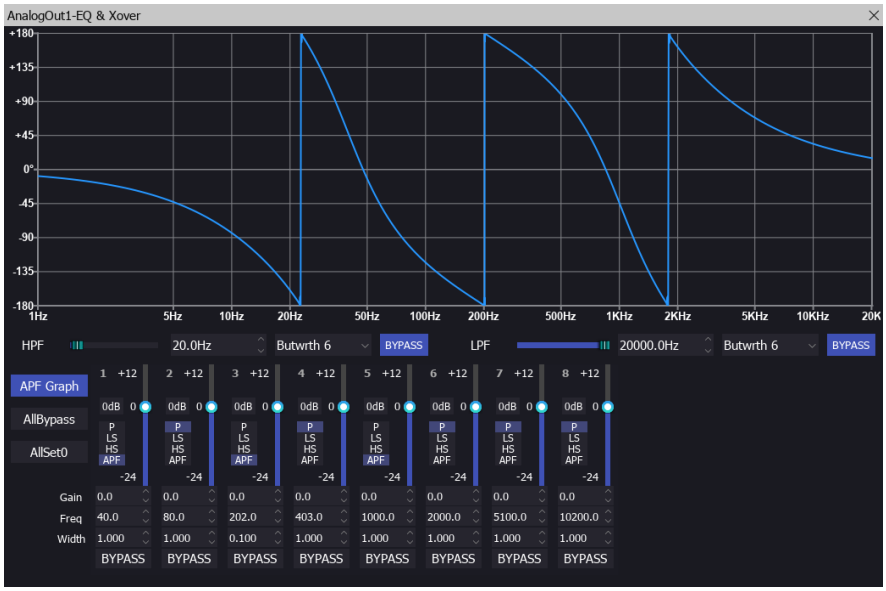


You can add 1-16 bands of EQ per channel before and/or after the matrix.
Toggle the **`AllBypass`** button to enable or disable the EQ. While the EQ is enabled, using any band's **`BYPASS`** button, you can toggle them individually on or off.

`EQ Types:` Parametric (**`P`**), Low Shelf (**`LS`**), High Shelf (**`HS`**)

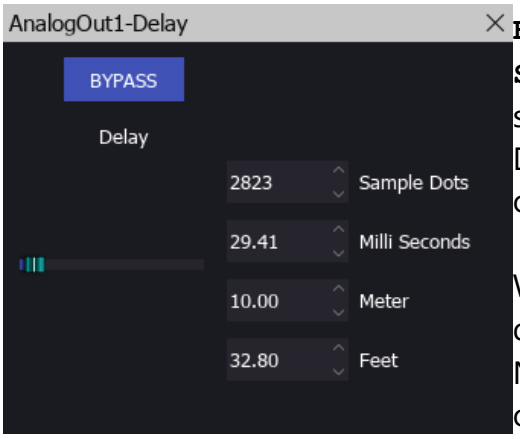
`Gain:` Set the gain of the corresponding channel between -24 and +12 dB
`Freq:` Set the desired frequency in Hz. (Same range as HPF/LPF: 19.7-20200Hz)
`Width:` Set the desired bandwidth for the band. This value can range from 0.017 to 4.75.

To use a band as an All Pass Filter, select the **`APF`** from the EQ types, then toggle the **`APF Graph`** view:



Here, only the **`Width`** and **`Freq`** parameters are usable, to shift the phase of the chosen frequencies, as the APF Graph shows, between -180 and +180 degrees.

*5.3.4 **DELAY***

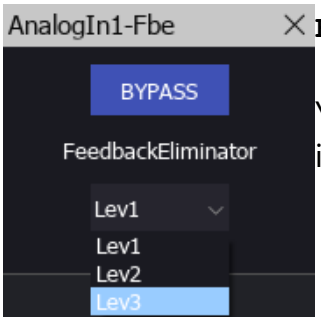


`BYPASS:` Toggle Bypass / Enable of the function.

`Sample Dots`, **`Milliseconds`**, **`Meter`** or **`Feet:`** You can set the delay time with either of these parameters. Defining a value in any box will automatically calculate every other parameter.

When selecting a delay function, you can set the maximum delay value from **`0.1s`**, **`0.5s`**, **`1s`** or **`2s`**.
Note, that all channels share a delay upper limit, depending on your latency settings. By default that is 2s.
(For example you can set 4 channels with 0.5s max delay each, but cannot set 2x 0.5s and add 2x 1s, because adding the last would exceed the 2s total.)

*5.3.5 **FBE** - Feedback Eliminator*



`BYPASS:` Toggle Bypass / Enable of the function.

You can choose between 3 levels. The higher you choose, the more it impacts the signal.

*5.3.6 **COMPRESS / COMPRESS-II** – Compressor / Multi-Segment DRC*



`BYPASS:` Toggle Bypass / Enable of the function.

`Threshold:` Set where should the compression take effect.
The value can range between -100 and 0 dBFS

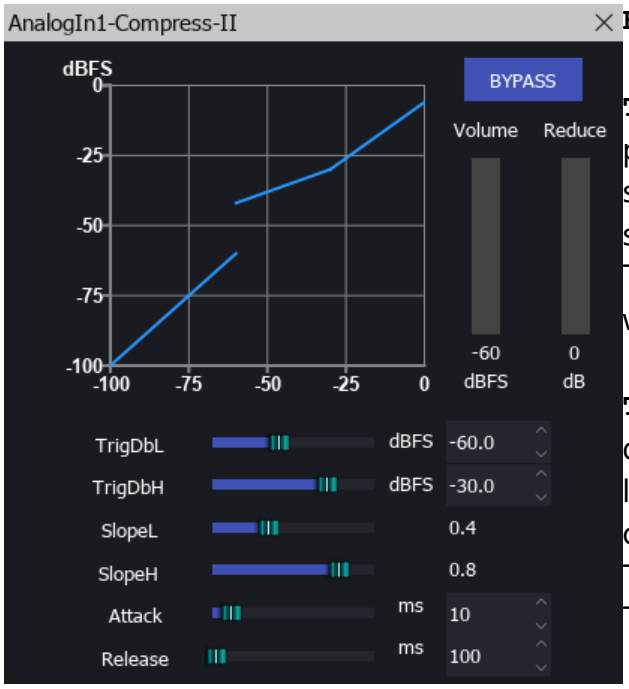
`Ratio:` Set the compression ratio (slope of the knee).
The value can range between 1.2:1 and 128:1 ratio.

`Attack:` Set the time, how soon should the compression take effect once the signal is above the given threshold.
The value can range between 1 and 100ms.

`Release:` Set, how much time has to pass after the signal is below the given threshold, for the compression to return to normal.
The value can range between 100 and 6000ms.

Compress-II is a multi-segment Dynamic Range Compressor that combines noise gating, expansion, and peak compression into a single interface. By setting independent lower and upper

thresholds, you can isolate background noise, manage mid-level dialogue or instruments, and safely limit loud audio peaks using a customizable input-to-output transfer curve.



`BYPASS:` Toggle Bypass / Enable of the function.

`TrigDbL:` Set the lower threshold level where the processor transitions from noise reduction to active signal path. Signals falling below this level will be sharply attenuated to silence background hum.
The value can range from -100 to 0 dBFS, capped by where TrigDbH is set.

`TrigDbH:` Set the upper threshold level where heavy dynamic control begins. When signals exceed this level, compression is actively applied to manage overall loudness and protect against clipping.
The value can range from -100 dBFS (or where TrigDbL is set) to 0 dBFS.

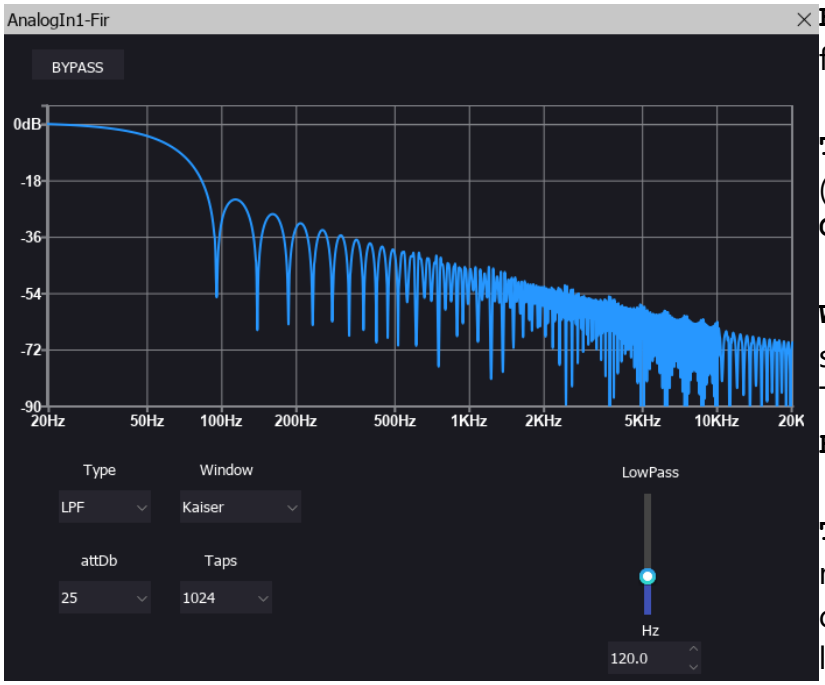
`SlopeL:` Adjust the gain transfer ratio for the quiet-to-medium signal range (between TrigDbL and TrigDbH). Lower values compress the mid-level dynamics more tightly, while a value of 1 provides fully linear, unaltered audio tracking.
The value can range from 0.1 to 1.

`SlopeH:` Adjust the compression or limiting ratio for loud signal peaks that cross above TrigDbH. Lower values create a flatter, more restrictive ceiling to strongly clamp down on high-energy peaks, while a value of 1 allows peaks through uncompressed.
The value can range from 0.1 to 1.

`Attack:` Set the time, how soon should the compression take effect once the signal is above the given threshold.
The value can range between 1 and 100ms.

`Release:` Set, how much time has to pass after the signal is below the given threshold, for the compression to return to normal.
The value can range between 100 and 6000ms.

*5.3.7 **FIR / FIR-II** - Finite Impulse Response Filter*



BYPASS: Toggle Bypass / Enable of the function.

Type: You can select Low Pass Filter (**LPF**), High Pass Filter (**HPF**), **Band** or **Custom**

Window: You can set a window for the selected type of (not custom) filter
The options are: **Rectangle**, **Hanning**, **Hamming**, **Blackman** and **Kaiser**.

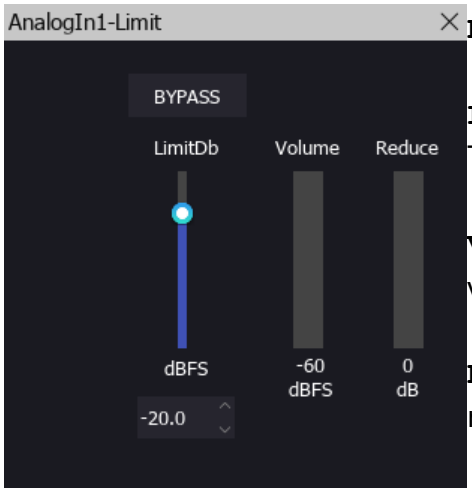
Taps: Even though you selected a maximum number of taps when choosing the FIR from the drop down list, here you can still change the resolution of the taps here.

attDb: This function is exclusive to the **Kaiser** window, as only that has means to modify the attenuation.
The values range from 21 to 120 dB.

Slider(s): If the selected type is not Custom, the respective low and/or high frequencies can be set with the sliders, or by typing the values manually.
The values can range: HPF: 20-19999.9Hz | LPF: 20.1-20000Hz | Band: Both HPF and LPF

Higher number of taps with low latency can be tolling on the CPU.
For this reason we added **FIR-II** which operates in the exact same way as FIR, but there is an **added 30 ms processing delay**, thus supporting higher taps on more channels.

*5.3.8 **LIMIT** – Peak Limiter*



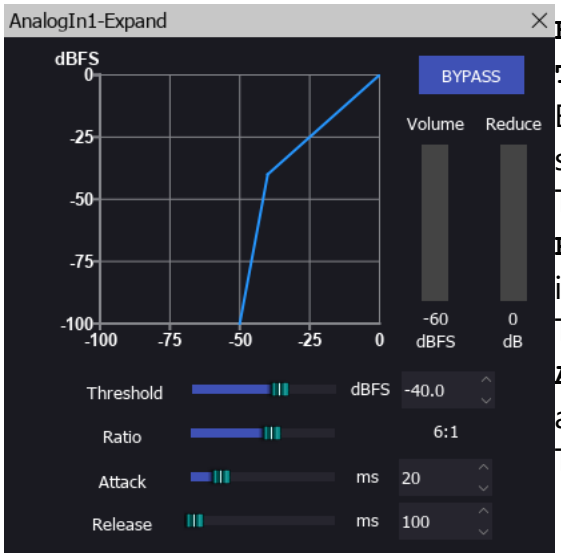
BYPASS: Toggle Bypass / Enable of the function.

LimitDb: Set the ceiling for the limiter to take effect from.
The values range from 0 to -100dB.

Volume: Here you can monitor the current incoming signal volume that the limiter receives (in dBFS).

Reduce: This shows how many decibels of volume is being reduced from the signal that is exceeding the limit (in dB).

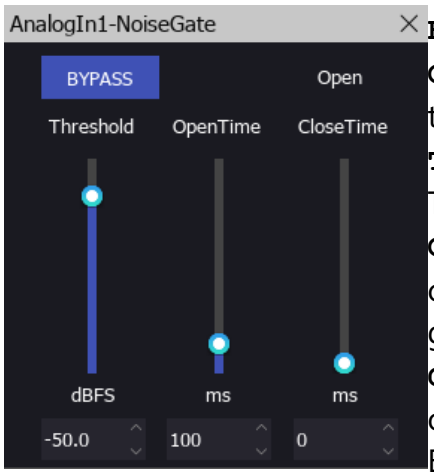
*5.3.9 **EXPAND** - Expander*



BYPASS: Toggle Bypass / Enable of the function.
Threshold: Set the value from which below the Expander should further lower the volume of the received signal.
The value can range between -100 and 0 dBFS
Ratio: Set how much the signal is reduced for every dB it is below the threshold.
The value can range between 1.2:1 and 128:1 ratio.
Attack: Set the time how quickly the gain reduction is applied after the signal drops below the threshold.
The value can range between 1 and 100ms.

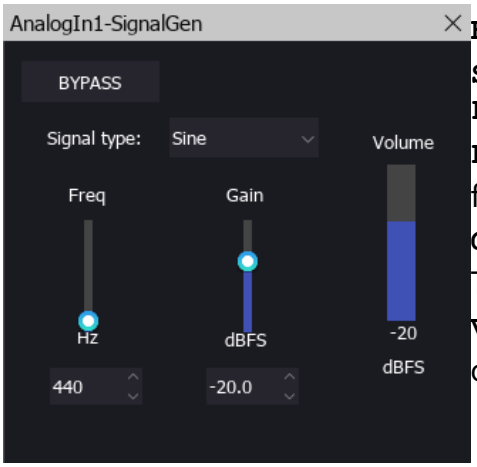
Release: Set the time how long it takes for the signal's gain to return to normal after it goes back above the threshold.
The value can range between 100 and 6000ms.

*5.3.10 **NOISEGATE** - Noise Gate*



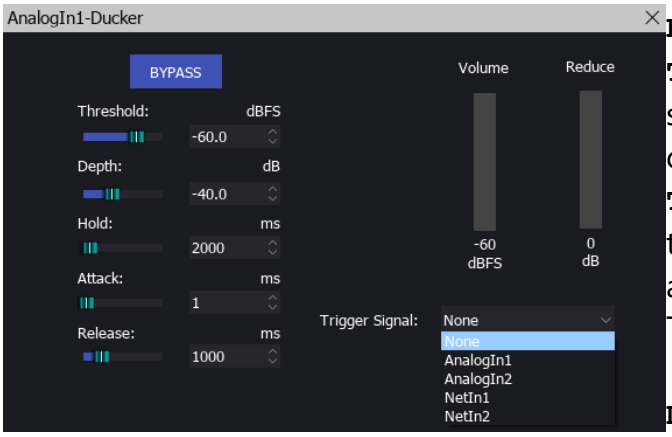
BYPASS: Toggle Bypass / Enable of the function.
Open/Close: On the top right, you can see the current state of the Noise Gate.
Threshold: Set the value from which below, the signal is muted.
The value can range between -42 and -100dBFS.
OpenTime (aka Attack): Set the time it takes for the gate to go from completely closed to fully open after the input signal exceeds the given threshold.
CloseTime (aka Release): Set the duration the gate remains fully open after the signal falls below the threshold.
Both Open and CloseTime values can range between 0 and 1000ms.

*5.3.11 **SIGNALGEN** - Signal Generator*



BYPASS: Toggle Bypass / Enable of the function.
Signal Type: Choose from 3 options: **Sine**, **White Noise** or **Pink Noise**
Freq: When **Sine** is selected, you can manually change the frequency of the wave from 1 to 20000 Hz.
Gain: Set the volume of the selected signal type.
The value can range between 0 and -60dB.
Volume: Monitor the current volume on the signal generator's output.

*5.3.12 **DUCKER** - Sidechain Attenuation*

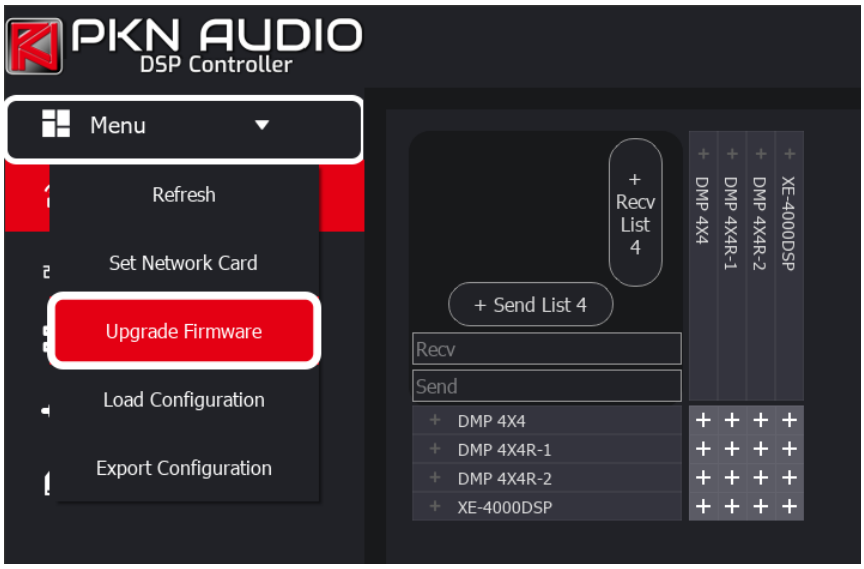


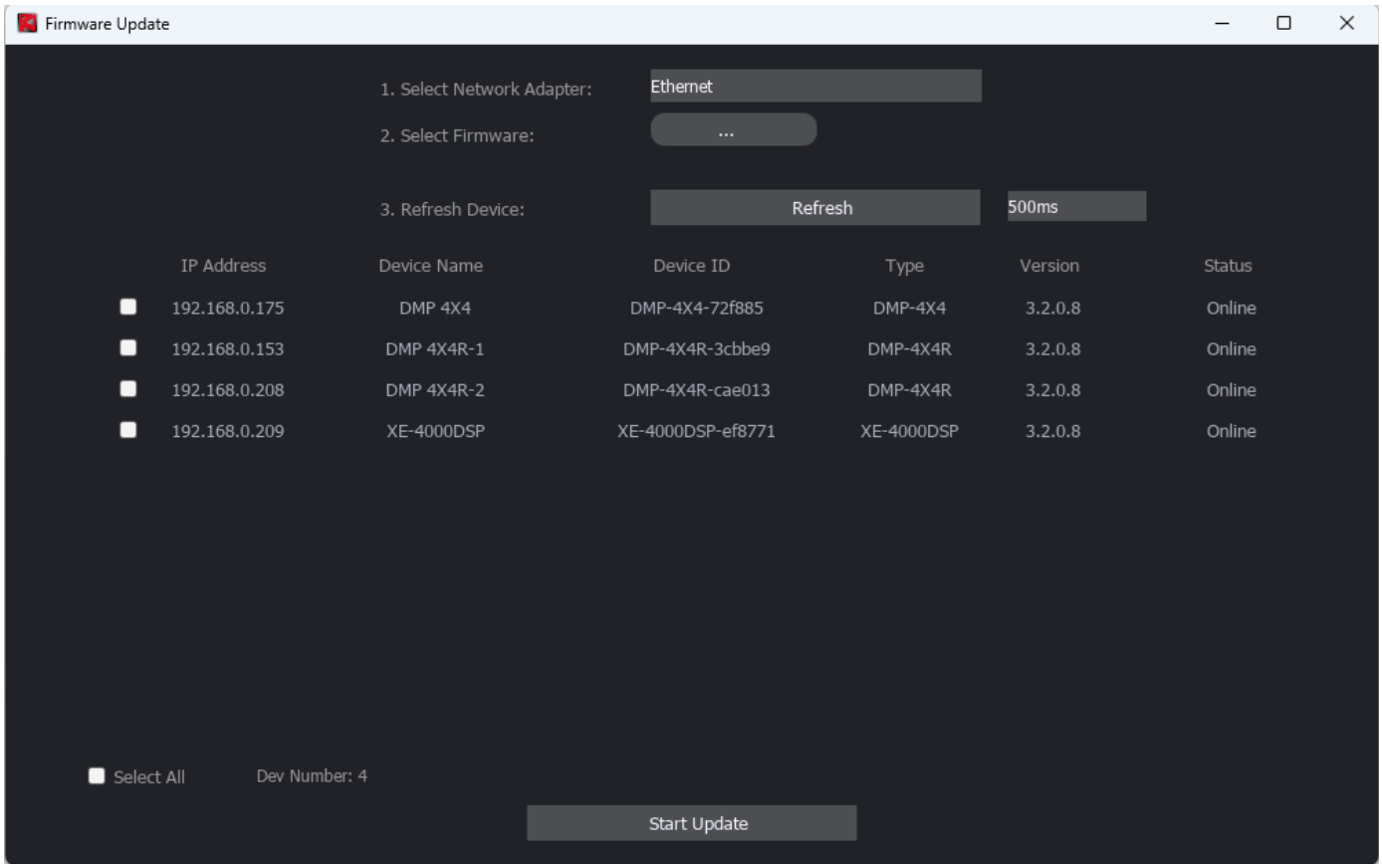
BYPASS: Toggle Bypass / Enable of the function.
Trigger Signal: Choose the sidechain input signal, which will determine the attenuation of the current channel.
Threshold: Set the volume that the selected trigger signal source should reach, to activate the attenuation.
The value can range between -80 and -50 dBFS.

Depth: Set the value where the volume should be reduced to, once the trigger signal source reaches the given threshold.
The value can range between -60 and 0 dB.
Hold: Set how long the signal remains at its attenuated level before the release.
The value can range between 1 and 50000ms.
Attack: Set how quickly should the ducker lower the signal volume once the trigger meets the threshold.
The value can range between 1 and 1000ms.
Release: The time it takes for the signal to return to its original level, after **Hold** time passes.
The value can range between 1 and 5000ms.

6. Firmware Update

Click `Menu → Upgrade Firmware`, to open the firmware upgrade window:





- `Select Network Adapter`: If you have multiple network cards, make sure to select the one that corresponds to the upgradable device's local area network.
- `Select Firmware`: Click on the ... to select the firmware upgrade package from your computer. You can check for new firmware versions on our Website!
- `Refresh Device:` If there are no devices listed, use this button to refresh the list.

  Using the checkbox, select which device(s) you would like to upgrade with the selected firmware, then click Start Update and wait until the process is finished.
  The Status should update accordingly to "InstallSuccess", and the device will automatically restart.

7. Troubleshooting

- *There is an error at software startup: xxx.dll library is missing!*

The prompt for the lack of .dll files indicates that the operating system lacks the necessary Microsoft runtime libraries. Downloading the latest `Microsoft VC++ runtime library collection` and installing it can solve the problem.

- *The software cannot detect the device!*

-Check if the network card selection is correct, make sure to select the network card in the same local area network as the device.
-Check if the firewall has blocked the software from connection. Be sure you allowed it the first time you started the software.
-Through the firmware upgrade feature, check if the device can be found. If the device is visible on the firmware upgrade view, but the software cannot detect it, first check the above two points.
-Check if the network port LED of the device is flashing. If it's not, be sure to check the hardware.

- *The software can detect the device, but cannot control it!*

-Check whether the network card selection is correct.
-Check whether the IP address of the computer and the IP address of the device are in the same local area network and domain.
-Check whether the computer is set to a fixed IP, resulting in the computer and the device not being in the same local area network.
-Check whether the network is normal with other devices.
-Check the "`Device Info`" menu, to see if the running time can be obtained. If the running time cannot be obtained, check the computer network and switch status (you can check for packet loss with a "ping" command from CMD towards the device's IP).
-Sometimes, when a USB network port adapter is used, there could be problems with poor network connectivity.

- *Firmware upgrade fails!*

-Check whether the network card selection is correct.
-Be sure to select the network card in the same local area network as the device.
-Check whether the firewall has blocked the software.
-If the firewall is enabled, the following programs need to be allowed for network access:

- ***PKNAudioDSP.exe***
- ***AES67Update.exe***
- ***gohttpserver.exe***