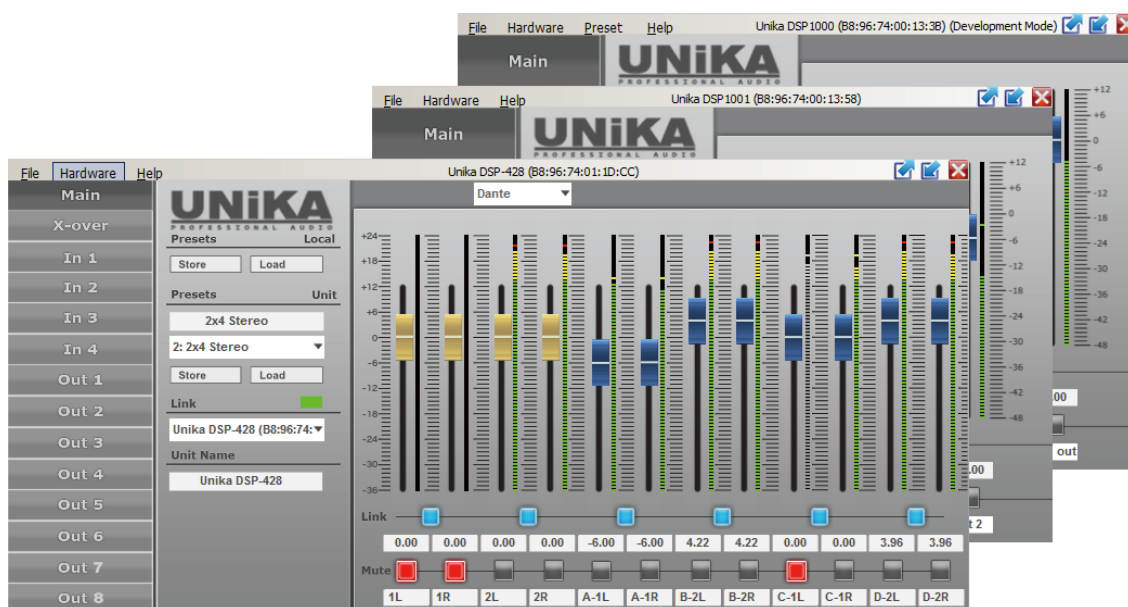# UNiKA

PROFESSIONAL AUDIO

DSP Audio Processor »»»

Operation Manual

Control Software and Device Front Panel Operation

Unika DSP-1000 | Unika DSP-1001 | Unika DSP-428

DSP Audio Processor

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DSP Audio Processor

Introduction

The UNIKA DSP speaker management processors can be operated from their respective front panels or control panels to a varying degree, depending on model. However, for maximum flexibility, remote operation from a Windows PC as well as from an Apple computer running OSX is also possible; for this remote functionality, the device must be connected via a wired Ethernet or a USB connection to the same network as the control computer. The control computer can be connected to the network wired or through Wi-Fi.

Getting Started

● It is recommended that a router with DHCP server is connected to the network so that it assigns the unit an IP address.

● When the controller is connected to the network, you can check or set its IP address via the unit window under Hardware - > Configure - > Network Settings → IP address.
  The default setting is DHCP (Auto IP).

● If Fixed IP address has been set, and this is no longer known or been forgotten, it can be requested via the display on the device. Press the menu button four times and turn the encoder wheel to the right three times to display the IP address.

● Note: you cannot connect the unit to the network through the USB or Ethernet at the same time. You can only control the unit from one platform at the same time – if you switch from one platform to another, you MUST shut down the control software at the previous platform first.

● Local connectivity of the device to the control computer from a front-panel USB connector is also possible to download initial settings or as a fallback in case of network failure.

● This Software Manual explains the remote setup and operation of the system through a network. The operation is basically same for all platforms; where differences exist they will be noted.

Installation and Setup

● Download and install the latest software from the UNIKA website. Note that for a Windows computer, there are separate applications for Windows 32 bit and 64 bit.
  http://www.unika.com.tw

● The software consists of a network window that list all known UNIKA DSP devices on the local network, and one or more unit windows for settings of a device. In the following sections we will discuss the Unika DSPControl program for Windows.

● The mac version works very similar to the Windows version.
  The mac menu bar is different from Windows for most program, the same differences are present with DSPControl.

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Unika DSPControl

● The Windows Unika DSPControl Network window is shown in Figure 1: DSPControl network panel. During installation your firewall setting are configured to allow communication over the network.

● Switch on the DSP and launch the DSPControl software. It will show the devices found on the network together with their mac address and IP address, similar to Figure 1.

● **Note**: you can control the unit from only one PC at the same time. If you want to switch control from one to the other, shut down the Unika DSPControl software on the previous control platform first.

● The '>' icon at the right side may be either > or >. When >, the unit is on line and can be activated by clicking the >.
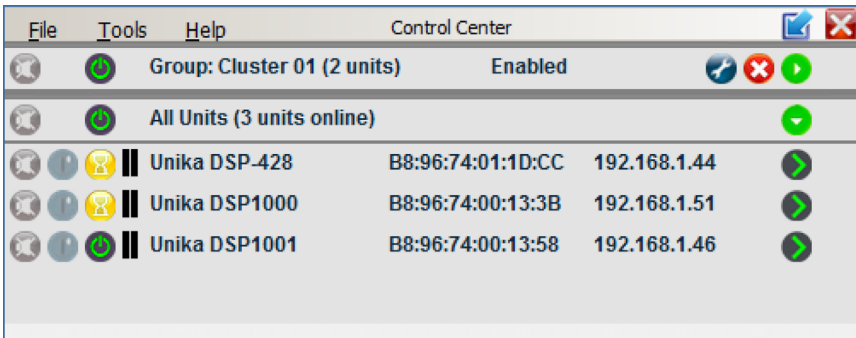

Figure 1: DSPControl network panel

The main purpose of the control app is to select a particular unit for manipulation and/or to group several units in a Group.

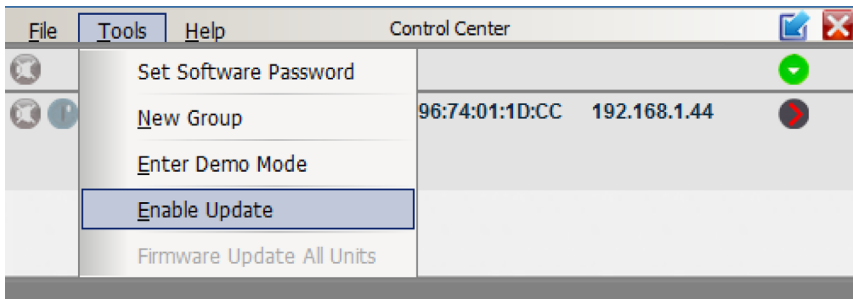The available Tools commands are shown in Figure 2: Tools menu items.
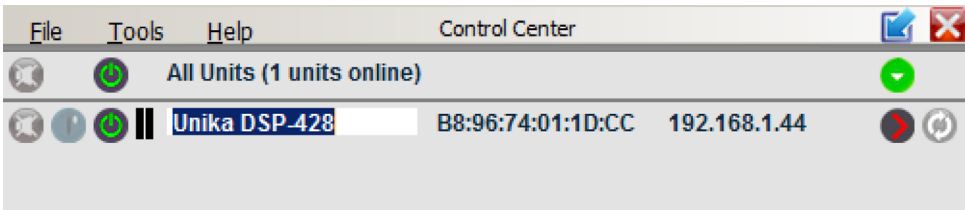

Figure 2: Tools menu items

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1. Tools

● 1.1 Tools | Set Software Password

Allows you to set up a password to access the graphical DSP software unit window. The DSPControl software has a separate, extensive system of operation modes.

Note: you can modify the Unit Name in the Unika- DSPControl program slot for the connected unit by just typing a new Name.

● 1.2 Tools / New Group

This command allows you to combine several on- line units into a named group. The group will be shown in the Control screen, see Figure 3.
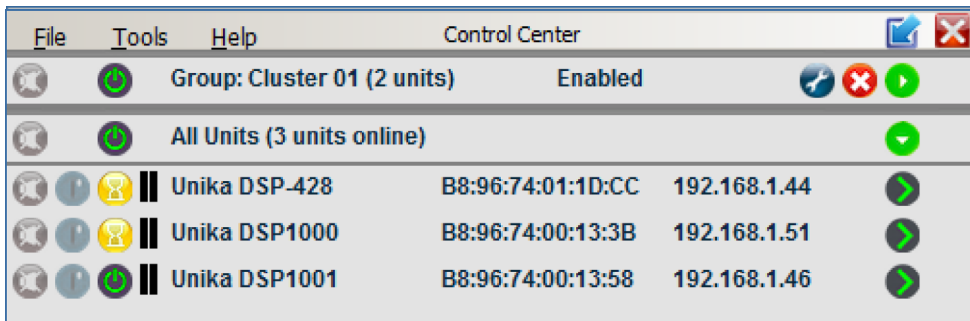

Figure 3: network window showing a named Group (Cluster 01) consisting of two units

Clicking the cross icon on the right deletes the Group; clicking the blue tool icon opens a screen where you can select the members of the Group; you will be able to select any unit which is on-line to become a member of the group.

A further selection from this screen opens a dialog where you can specify which parameter, for which setting and for which channel you wish to link within the Group.

For instance, you can link only gain, or gain and PEQ settings for Input & Output channels.

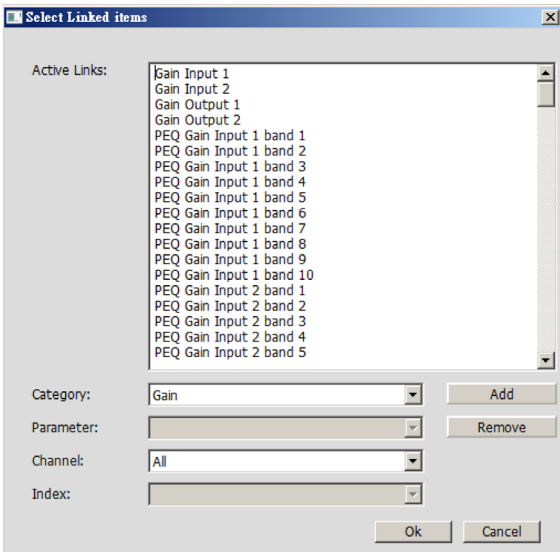An example Group setting screen is shown in Figure 4.

Figure 4: Sample selecting dialog box showing some linked items for a Group

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● 1.3 Tools / Enter Demo Mode

Allows you to exercise the functionality of an Unika DSP unit without hardware present for training and familiarization purposes.

● 1.4 Tools / Disable / Enable Updates

When set to Enable Updates, there will be an indication when firmware updates for the unit will be available. This is shown on the right edge of the unit identification in the control program with a 'rotary' icon, see Figure 5. The icon turns yellow color when updates are available. Clicking the icon will initiate the update.
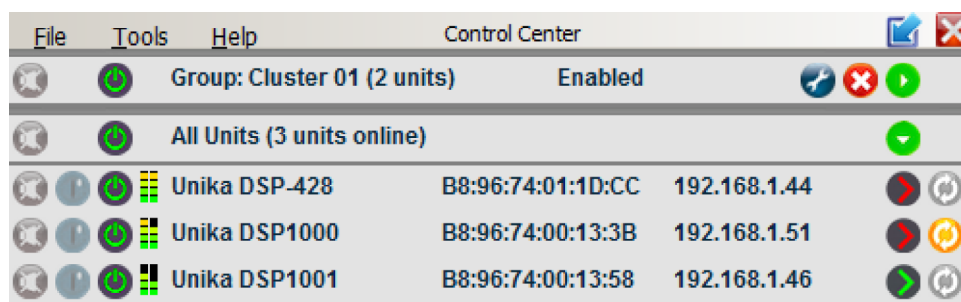

Figure 5: rotary' icon yellow color at right indicates firmware update for this unit is available

This is a 'brute force' update which will also restore the unit after a crash or other unrecoverable situation. All presets and other settings will be lost however.

Clicking the green > symbol at the right of the connected unit info line in the Unika DSPControl window will load the software for the unit. At this point, the software will go through a synchronization cycle and, if a Personal Identification Number (PIN) was defined before, will ask for the PIN.

The Main graphical interface screen will come up similar to Figure 6; this is the opening screen of the actual application controlling the settings of the unit.

● 1.5 Other Settings

● 1.5.1 Power-On Preset

In the menu item Hardware - > Configure - > Power On Preset, a preset can be selected, which is automatically loaded when the device starts.

● 1.5.2 Read Only Preset Range

When you are logged on as Developer or higher, you can block a certain preset range from being altered.
In Hardware - > Configure - > Read-Only Preset. If the value is, for example, set to 10, the first 10 memory locations in the preset list for User and Admin mode are read only and the user cannot save presets to those locations.

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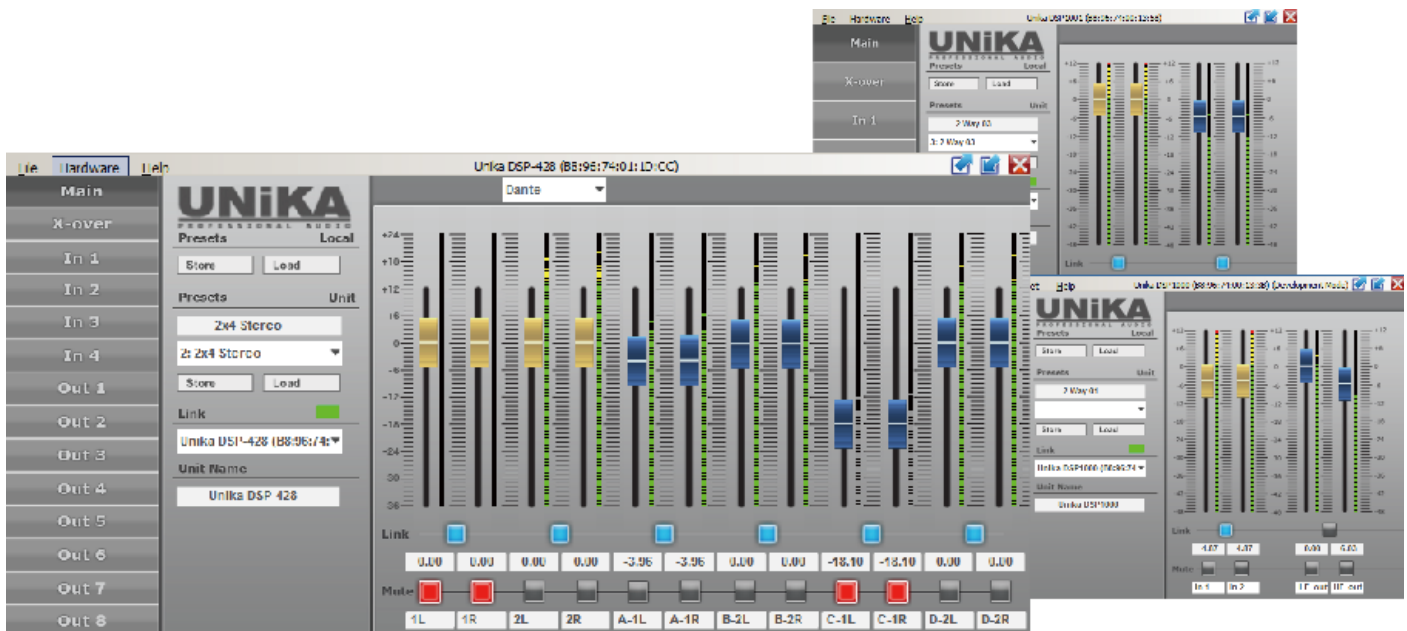
2. Unit Window: Main Panel

Figure 6: unit window

● 2.1 Unit Window Menu

Figure 6: unit window shows the unit window from which all setup screens and options can be reached. Clicking the exit cross will close the software but leave the network Control program running.

● 2.2 File Commands

● 2.2.1 File / Open and File / Save

Presets are the snapshot of all settings for a particular configuration. With these commands you can save a preset, and reload it at a later time. In addition it offers the choice to use the .preset file format or the .txt format. Text files are a way to archive settings for documentation purpose.

● 2.2.2 File / Backup presets and File / Restore presets

These commands operate similarly to Save and Open, but will save all presets in the unit in one directory, or load all presets from a directory.

● 2.2.3 File Quit – self-explanatory

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● 2.3 Hardware Commands

The Hardware menu offers several functions as shown in Figure 7: unit panel Hardware menu items.
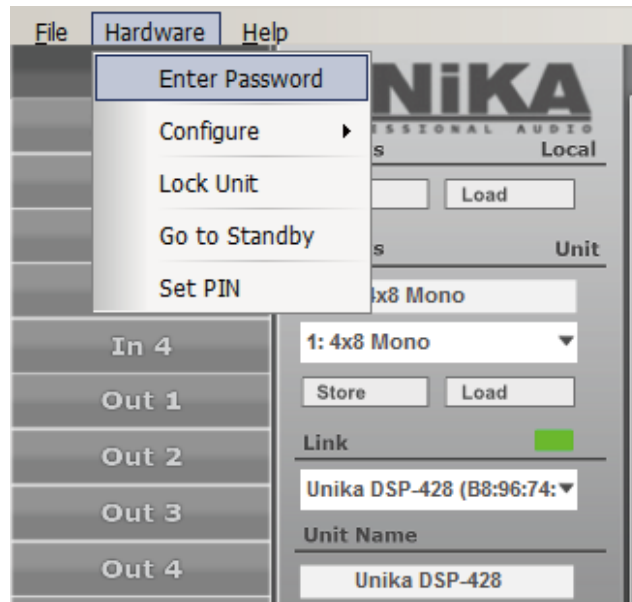

Figure 7: unit panel Hardware menu items

● 2.3.1 Hardware / Enter Password

The unit has three access levels plus locked. Entering the right access level password unlocks that access level. Please see password protection, access control & locking.

● 2.3.2 Hardware / Configure

The Configure menu item opens an additional screen where several options are presented. However, the options shown will depend on the access level currently in force – to be discussed.

● 2.3.3 Hardware / Configure / Power On Preset

Allows you to select a preset that will be automatically loaded at power-on;however any changes to this preset will not be carried over to the next power-on cycle unless you have saved the new setting, i.e. it will revert to the saved preset.

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● 2.3.4 Hardware / Lock Unit

Clicking this menu item will lock the unit. When locked, the unit window shows input levels but no controls or other settings, and nothing can be changed, as shown in Figure 8: detail of unit panel in locked state.

An attempt to lock the unit will generate a warning that unlocking will only be possible with a valid password. The actual operation of this function depends on the access level as described later.

Figure 8: detail of unit panel in locked state.

● 2.3.5 Hardware / Go to Standby

Will place the unit in Standby. In this state, outputs are disabled but all settings can still be manipulated.

● 2.3.6 Hardware / Set PIN

Clicking this option allows you to enter a 4-digit PIN number. When this is set, you will need to enter the PIN number at any time you connect to the unit. **Note:** the PIN control can be switched off by setting the PIN to 0000.

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3. Main Setup Screen Functions

● **3.1 Unit Window Functionality**

On the left side, the Unit window (Figure 6) gives access to the cross-over settings screen as well as all settings screens for each individual Input- and Output channels. These will be discussed later. The following commands are found in the center of the Unit window:

● **3.1.1 Presets - Local**

These commands allow you to save and re-load configurations as .preset files on the connected PC.

● **3.1.2 Presets - Unit**

These commands are similar as the other Save and Load commands but with the distinction that presets can be saved to and reloaded from the unit internal memory. The drop-down menu under Default Preset shows a list of memory slots where named presets can be stored and recalled. Since this is internal in the unit they will be available independent of the stored presets in a computer.

● **3.1.3 Link**

Under Link there are several status related indications. The indicator at the right will be green for an active link with the unit or red when the unit is off-line. The name and MAC address of the connected unit is also shown.

● **3.1.4 Unit Name**

Shows the unit name of the connected unit, which can be edited by typing in a new name. The new name will be reflected in the Link slot as well as in the name field in the Unika DSPControl window.
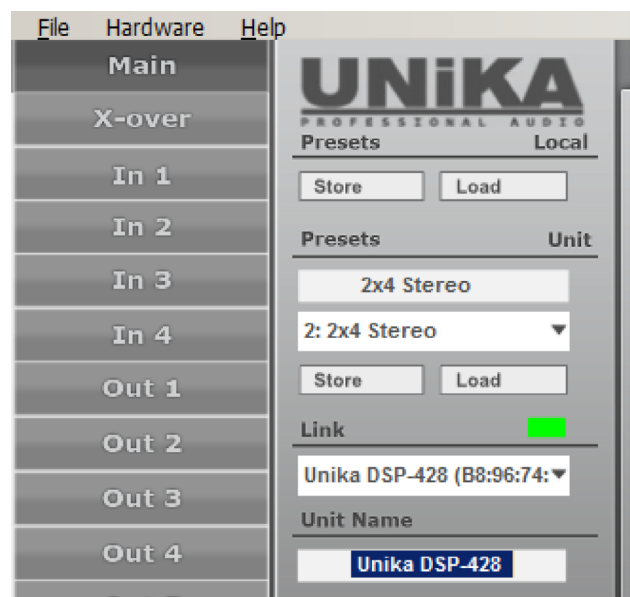

Figure 9

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● **3.2 Unit Window Channel Settings**
  On the right side are the Channel settings.This area is divided into several subsections:

● **3.2.1 Level Faders for Input Channels**
  These faders control the level from each of the physical input connectors as sent to the Input processing channels. Channel names are shown at the bottom line. Level settings are also shown numerically below the faders, and can also be set by entering a specific value in the corresponding text field.

● **3.2.2 Input Selection for Input Faders**
  When your unit is equipped with AES3 (AES/EBU) or Dante inputs, you can select the source for inputs 3 and 4 from this pull down menu.

● **3.2.3 Level Faders for Output Channels**
  These faders control the levels for the physical Output channels at the rear-panel output connectors. Level settings are also shown numerically below the faders, and can also be set by entering a specific value in the corresponding text field.

Faders auto-zero: Whenever you double-click on a fader, that fader moves to the default position. This is usually 0dB (unity gain) for gain faders. This operates on all views and screens. When two channels are linked, the odd-numbered channel can be used to manipulate both linked channels.

● **3.2.4 Link Buttons**
  When activated, all settings for the related pair of channels pairs are linked together and can only be changed with the left (odd numbered) fader of the pair the right fader will follow. Similarly, entering numeric values in the level fields will only be accepted for the left channel of a pair, with the right-channel numeric value following, the even- numbered channel of a linked pair will be unresponsive but follow the odd-numbered channel settings.
  To reset a paired level to 0dB you must therefore double-click the odd-numbered fader.

● **3.2.5 Mute Buttons**
  Mute buttons allow muting of an Input- or Output channel on a per-channel basis, independent of any linking in effect.

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● **3.3 Digital Inputs (Option)**

When your device is equipped with AES3 (AES / EBU) and/or Dante input, you can set the source of input channels 3 and 4.

The AES-3 (AES / EBU) input signal can be fed via AES XLR socket in the controller. The Dante input is fed to the units via the same RJ45 connector that is also used for control.
Available input sources: Analog, AES-3 and DANTE.

● **3.3.1 AES Failover**

In the menu item Hardware -> Configure -> AES analog failover, a radio button can be turned on so that when there is no AES / EBU signal, the unit automatically switches to the analog input.
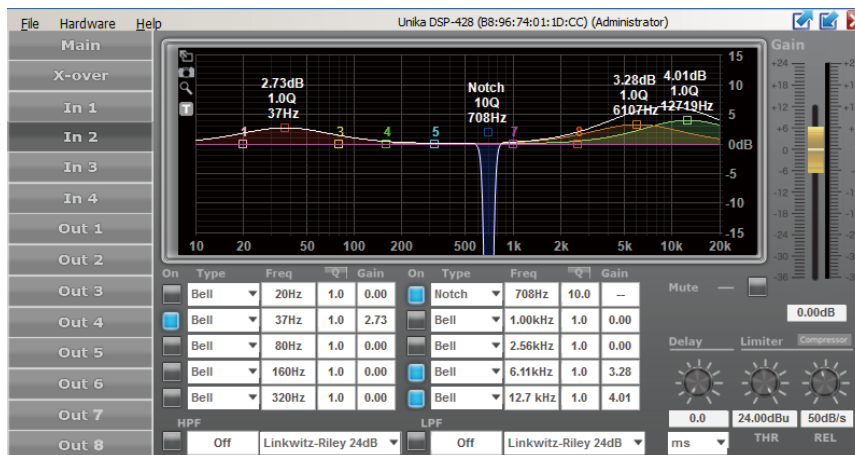
4. Input Channel Settings Screen
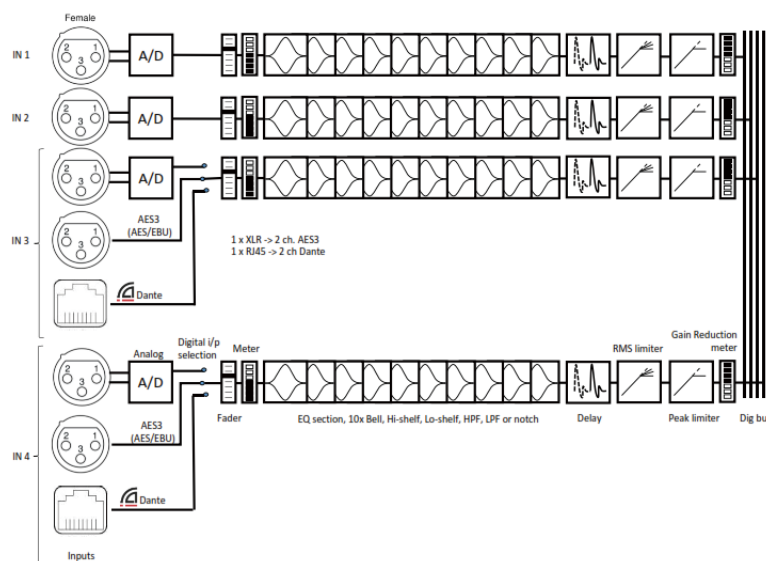

Figure 10: Input channel settings window

Figure 11: Input signal flow

Clicking on an Input channel number in the Unit window will bring up the input channel settings screen see Figure 10. In this screen, individual input channels can be equalized and/or filtered.

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● 4.1 Filtering and EQ

All input channels can have filtering and equalization applied individually, or as linked pairs. Input channels can be linked in the Unit window, and in that case, all EQ and filter settings for the pair will be linked.

If input channels are linked, EQ and filters for each pair can only be set for the odd number – for example, setting an EQ for input channels 1 and 2 can thus only be done in the Input 1 screen.

EQ can be set either in the graphical screen by cursor dragging, or in the numerical fields and drop- down lists below the graphical screen.
Type, Freq., Q/BW and Gain can be set for each input or pair of linked inputs.
All settings are reflected real-time in the graphical screen.
The Q/BW column toggles from indicating Q or bandwidth (BW) every time the column label is clicked.

● 4.2 Input Channel Transfer Function Curve

On the graphical screen, a white curve represents the compound transfer curve for that input channel.
The various elements that make up the curve like EQ settings and the crossover filters are indicated by their respective colors and EQ numbers.

There are two ways to toggle the contribution of an EQ or crossover element to the white compound curve on or off. You can either click the black/blue button next to each EQ or the HPF or LPF filter at the bottom of the screen, or you can double-click on the numbered square of the elements in the graphical screen. Even if the effect of the element on the compound curve is removed, the colored non-contributing curve of the element remains visible. This is a useful functionality to review the contribution of several interacting filter and EQ settings on a channel.

● 4.2.1 Screen Expansion and Legends

On the left-hand side of the graphical screen are four icons to help precise settings. The top Arrow icon expands the graphical screen to the size of the window. Clicking this icon again from the expanded screen expands it to the full display size. In either of the expanded settings, clicking the X-icon at the top left will collapse the graphical area to the default setting.
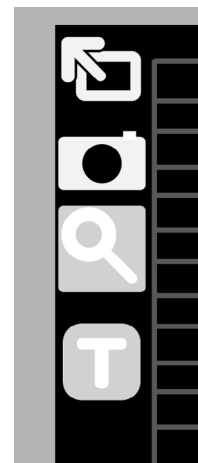

Figure 12: precise setting tools

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The Camera icon will generate a .png picture file similar to a screen shot which you can save for documentation or instruction purposes.

The T (text) icon will place the frequency, Q and gain values of each EQ and filter setting in the graphical screen. These settings will anyway be visible when changes are being made, but with the T icon activated, they will be visible continuously. This duplicates the numerical indications in the lower part of the screen but it will be useful with the expanded screens mentioned before.

For very accurate settings of EQ or filtering you can click the Looking Glass icon. All movements of the cursor now have a much smaller effect and you can drag curves with much greater precision, clicking the Looking Glass again reverts to normal operation.

● 4.2.2 Input Gain, Mute

The right hand side of the screen shows the fader to set Input channel gain and a Mute button. Settings are individual per channel unless input channels have been linked in the Unit window.

● 4.2.3 Delay, Limiter, Compressor

In the bottom right of the Input screen there are settings for channel delay, limiter threshold (-48 to +24 dBu) and Limiter release rate (10 to 100 dB/s) settings plus a button to enter the RMS Compressor menus. Channel delay can be set in a variety of distance or time units from the drop-down list.
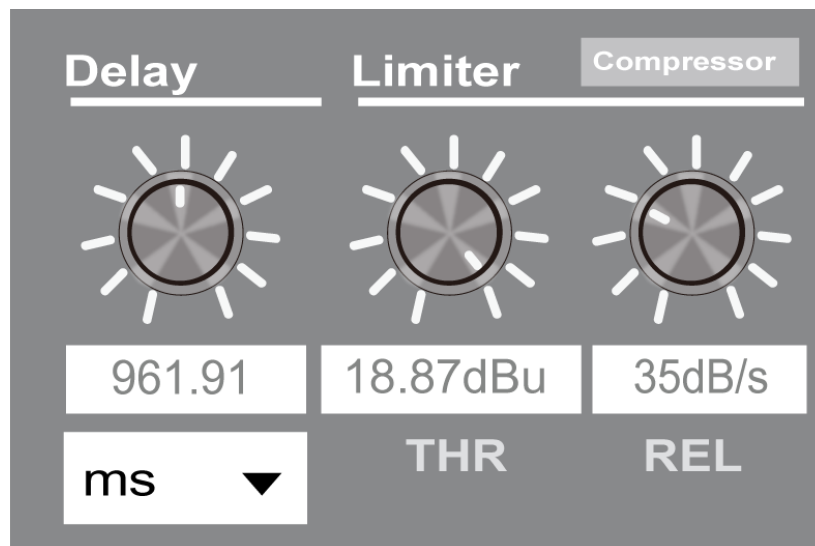

Figure 13: Detail of Delay, Limiter & Compressor in input- window

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● 4.2.4 RMS Compressor

Clicking on the compressor button brings up the compressor menu Figure 14.

In the compressor menu you can set the parameters of the RMS Compressor, the threshold, attack, hold, and release time, the compression ratio and the make-up gain.

Next to the controls is a VU meter that also displays the gain reduction (GR). Be sure to use the 'Back' button to return to the output menu.
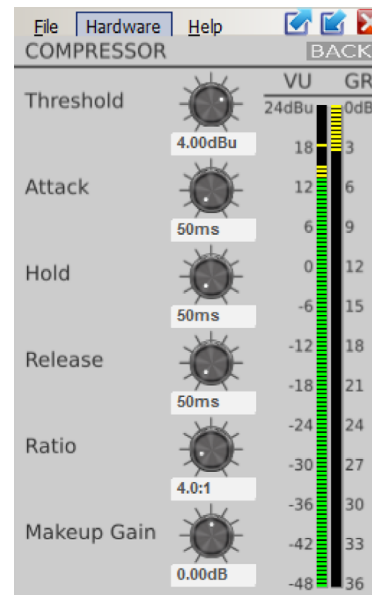
Figure 14: Compressor settings window

5. Crossover Settings Window
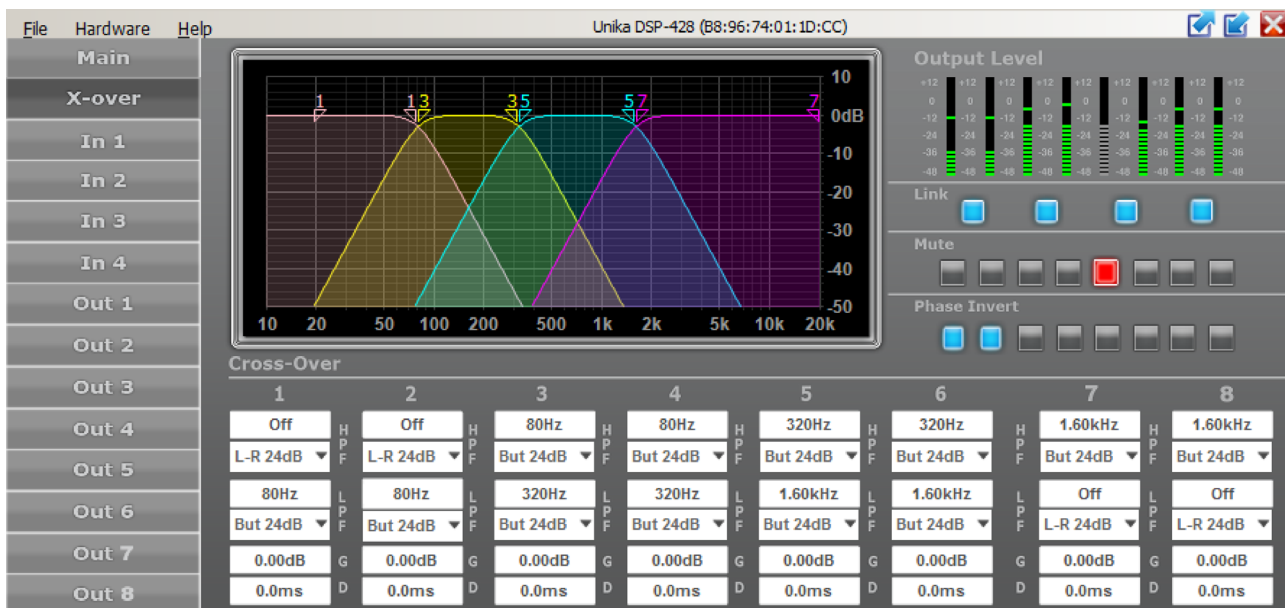
Figure 15: Crossover window

When you click X-over in the left unit window function list, most of the screen is dedicated to the controls and indicators for setting up the cross over filters as shown in Figure 15.

● 5.1 Graphical Control

The graphical window shows the crossover settings for each of the output channels in dedicated colors. The numbers for each channel have a triangle shape and the direction of the triangle denotes whether the triangle controls the upper or lower corner frequency setting.

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For each channel, the cross-over frequency on either side of the band can be set with the cursor or by entering a value in the numeric fields below the channel. To change the frequency setting by mouse, move the cursor over the number of the channel you want to change in the graphical screen until the cursor changes to a 4-way arrow. Now you can click- and-drag the number triangle and the frequency value will change real time, and will be reflected both in the screen curve and the numerical value. Frequency limits are 20Hz and 20kHz in each case.

● 5.2 Filter Type

The filter type and slope for each Output channel and at each end of the band can be set by selecting the desired type from the drop-down menu below the graphical area, for that channel. Separate filter types are available for the HP and the LP setting.

● 5.3 Gain

The Output channel gain, in dB, can be set by entering a numerical value in the field indicated by "G".

Note: the Gain value entered here will be reflected in the setting for the channel in the Unit window level setting; Alternatively, changing the Gain setting for a channel in the Unit window will be reflected in the Crossover screen gain setting!

● 5.4 Delay

For each channel a delay can be introduced in the numeric field indicated in milliseconds.

● 5.5 Link

When two channels are linked by activating the related Link button in the top right-hand side of the screen, the settings for that pair are combined and only a single color and number icon for the pair is shown in the graphical screen (only the odd number of the pair is shown). Also in this case, all numerical values and filter types for the two channels are linked and can only be changed in the left hand side (odd numbered) channel of the linked pair.

Note: Changing the Link status of a pair of Output channels will also be reflected in the Link status for that pair in the Unit window; alternatively, changing the Link status for a pair in the Unit window will also carry over to the Crossover screen!

● 5.6 Mute, Phase Invert

The top right side of the screen shows these additional options which are self-explanatory. Changes to the Mute status of a channel or channel pair will also reflect in the Unit window and vice versa. Mute status of an individual channel can be changed independent of whether a channel is linked to another channel. However, the Phase invert status will be link for two channels if the channels are linked.
The phase invert button changes the polarity of the output channel. The signal is changed 180 degrees.

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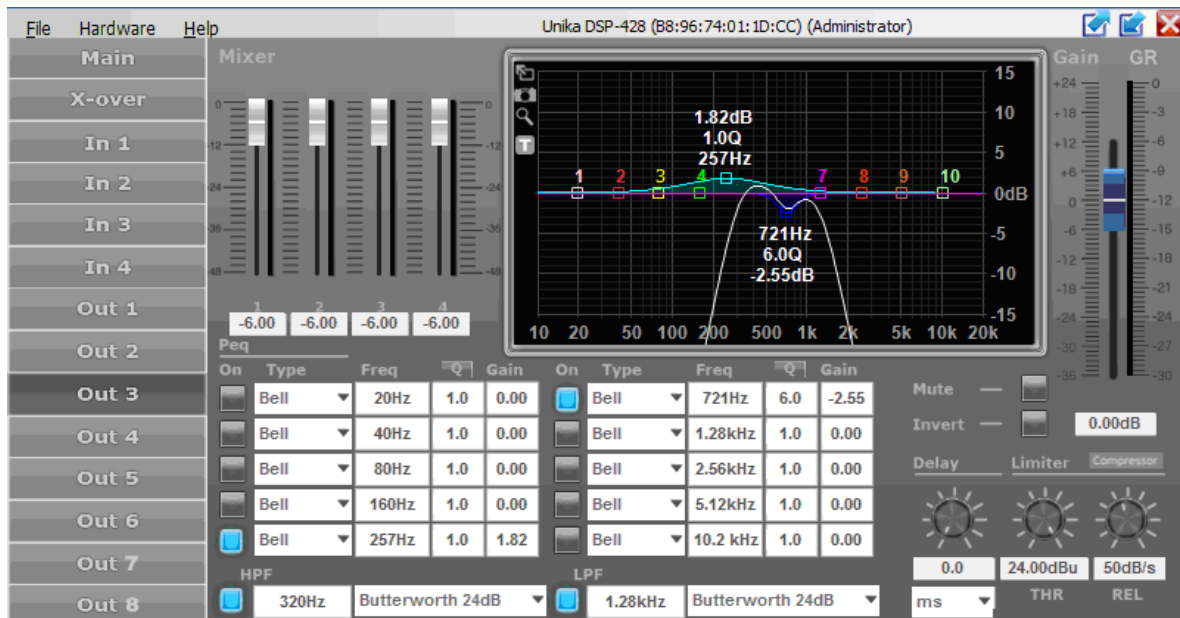
6. Output Channel Settings Window

Figure 16: Output channel window

● 6.1 Input Output Matrix

Figure 16: Output channel window, faders are shown for each of the input channels. This will allow mixing each of the input signals to each individual Output channel. Even if Input channels are set to Linked in the Unit window, it is still possible here to mix individual Input channels to an Output channel in any desired ratio. Level settings are shown numerically below each fader. The available range is from -48 dB to 0 dB, as in other screen, double- clicking a fader resets it to default.

● 6.2 Graphical Screen

A complete control screen is available for each Output channel. The graphical area reflects the Crossover settings for the channel in addition to any equalization settings set up in this screen, and is very similar to the available settings in the Individual input channel screens. The status of an element is always reflected in the blue on/off button next to each element.

● 6.3 Relationship between Input- and Output Settings

At the bottom of the Output channel settings screen (Figure 16) you will see the high-pass and low-pass settings from the cross-over settings for the particular Output channel repeated. Both the slope and the filter type are shown, and the curves are also shown in the Output graphic screen. Any changes you make here are also reflected in the X- over screen. This way, you have control over the main crossover settings as well as the EQ and filtering for a specific Output from a single screen. In contrast, the crossover screen gives you the overview of all cross-over settings for all channels in a single screen.

Both the HPF and LPF filter can be disabled here, this will also be visible in the Crossover screen.

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● 6.4 Output Filtering and EQ

Similarly as described for the Input channels, each Output channel can have up to 10 EQ's and filters attached to it (Figure 16). For each of these, EQ type, Frequency, Q or BW and Gain can be set.

Each EQ can be set in the graphical screen by dragging the appropriate EQ number as described earlier, as well as by entering the desired numerical values in the fields. When using the mouse to drag settings, the actual numerical values for Gain, Q or BW and Frequency are shown in the screen.

Note: The four icons in the top left area of the screen (expansion arrow, camera, 'T' and Looking Glass) act as described for the Input screen.

● 6.5 Output Channel Transfer Curve

On the graphical screen, a white curve shows the compound transfer curve for that output channel.
The various elements that make up the curve like EQ settings and the crossover filters are indicated by their respective colors and EQ numbers.

Note: that the white compound transfer curve also reflects any cross-over settings made in the X-over screen, when activated.

● 6.6 Delay, Limiter, Compressor

Like the input channel, each output is also equipped with a dynamics processing section.

The compressor and limiter at the output limits the entire signal. In the bottom right of the output screen there are settings for channel delay, limiter threshold (-48 to +24 dBu) and Limiter release rate (10 to 100 dB/s) settings plus a button to enter the RMS Compressor menus. Channel delay can be set in a variety of distance or time units from the drop-down list.

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● 6.7 RMS Compressor

Clicking on the compressor button brings up the compressor menu.

In the compressor menu you can set the parameters of the compressor such as threshold, attack, hold and release times, ratio and makeup gain.

Next to the controls is a VU meter that also displays the gain reduction (GR). Be sure to use the '**Back**' button to return to the output menu.
Note, is selected equipment with power amplification built in, it is possible to set the threshold as an absolute (RMS) voltage. The system then displays the corresponding output power for 4 and 8 ohms loudspeaker system. This makes it easy to set the maximum power that power amplifier will apply to the connected loudspeaker making it easy to set the threshold so that the attached loudspeaker is not thermally overloaded.

Note: Applying too much electrical power is not the sole failure mode for a loudspeaker. Applying power outside of the rated frequency range can easily damage a loudspeaker at low power levels. UNIKA advises to carefully consider the correct settings for each driver. The RMS Compressor and Peak Limiter can be valuable tools to protect drivers, but will not guarantee that the driver cannot be damaged.
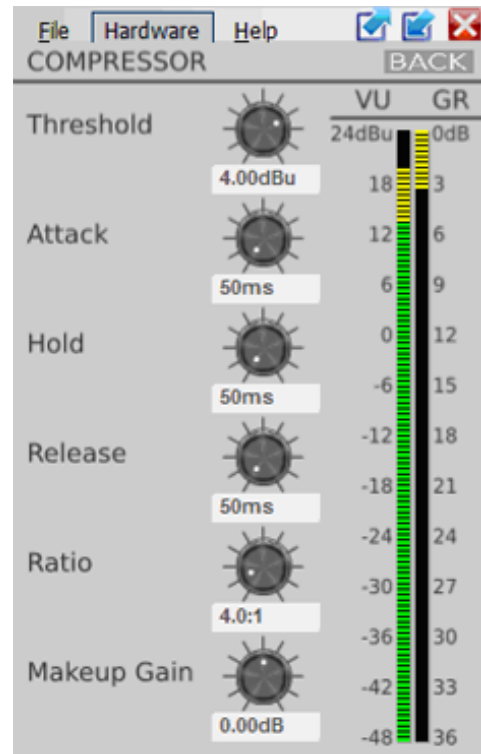

Figure 17: Output RMS Compressor

● 6.8 Change / Adjusting Values

To change a value in the input fields in a Windows PC, it is also possible to adjust the value using the arrow keys or the scroll wheel on the mouse.

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7. Access Control and Locking

This unit has three access levels, plus locked mode. It is intended to enable the right amount of control to each type user.

The three levels are:
- **User** (user of the system)
  The user is the person that makes use of the system that the DSP product drives. This can be the FOH engineer, venue engineer. This user level has the lowest amount of control of the configuration.

- **Administrator**, e.g. installer, sound system designer…
  The administrator is the person that changes the system. In some cases this can still be the DJ or FOH engineer, typically when they are familiar with the venue. It can also be the designated maintenance engineer.

- **Developer** (the person responsible for the acoustic performance)
  The developer tunes the crossovers, equalizes the outputs, and sets the output limiter so that it protects the system.

The standard passwords are as follows
User mode: U0000
Administrator mode: Adm0001
Developer mode: Dvl0001

Note: Please change these passwords as soon as possible to prevent unauthorized access to your system.

Note: Please store your changed passwords securely. There is no way to retrieve lost passwords. If you break into your unit by setting and forgetting a password, we cannot help you without an RMA process. Resetting your unit can only be done in the factory. We do not have or give out a master password. Please head this warning and keep your password in a secure location.

The privileges of each level are now explained in more detail.

At start-up the unit will be in User Mode by default, and will have access privileges as configured previously for that Mode. There is no specific indication that you are in User Mode. However, Administrator and Developer Mode will be indicated at the top of the window next to the unit name.

As developer, you have access to and can set all passwords, access levels and privileges.

You should receive a Developer password with your unit; if you log on with that password you can then set up a User and Administrator password and the privileges for each level as described in this Chapter.

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Note:that there is no functional relation between the software password in the network window of the control software, the PIN control system and the hardware-related access control functionality for the DSP software. Each operates separately.

If you enter your Developer password, the Hardware drop-down menu will show additional options that will not be visible in the other modes.
In addition to change your Developer password,you have the option to set the passwords for Administrator and User modes (Figure 18).
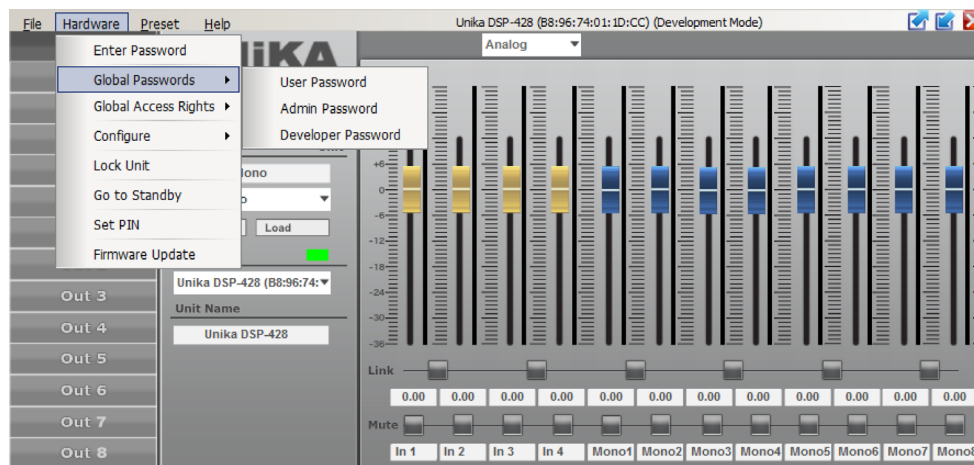

Figure 18: Additional functionality available in Developer mode

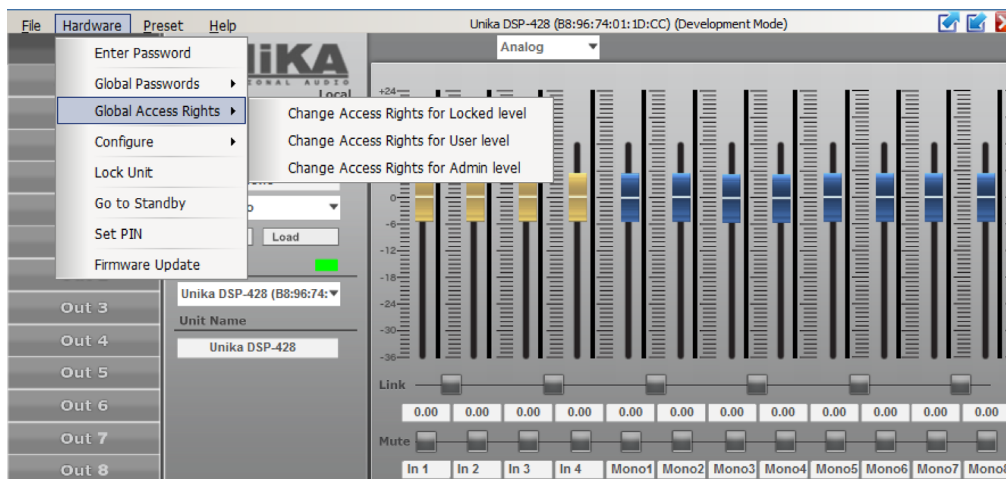You can also set all Access rights for the other modes as shown in Figure 19.

Figure 19: Developer can set Access rights for all other operation modes.

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For each level, there is a dialog screen with privileges that you can set for that level, accessed through the **Hardware / Access Rights** menu. You set the User Mode privileges that are required to work with the unit as a user, while with Administrator and Developer level access, you can allow progressively more detailed settings and adjustments to be made. For this reason, the Access Rights dialog window will show more privileges with higher access levels. The User Mode privileges list with a sample setting is shown in Figure 20.
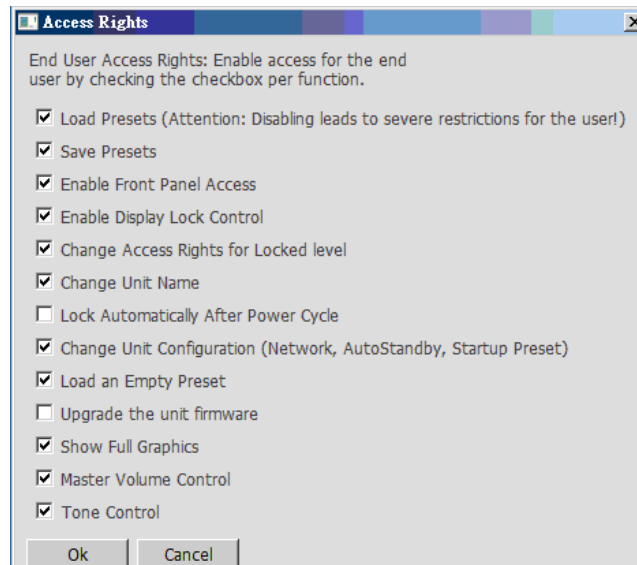

Figure 20: Sample User access level privileges dialog box

A similar screen will be shown for Administrator mode; Figure 21.

Figure 21: Administrator access level privileges dialog box.

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Finally, there is a dialog screen to set the access levels in locked level mode as shown in Figure 22.

Figure 22: Locked level access settings dialog box

A typical Locked screen is shown in Figure 23 Sample Locked screen.

Figure 23: Sample Locked screen

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Access rights

Most descriptions are self-explanatory, the following rights can be controlled:

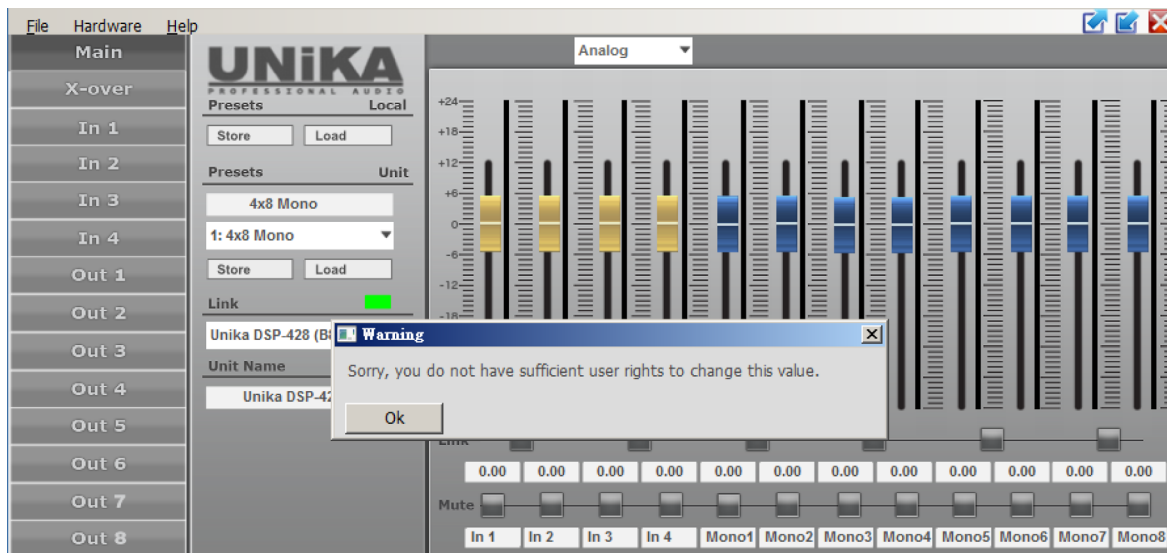● Load presets – loading presets from the device preset list

● Save presets – Saving presets to the device memory

● Enable Display Lock Control – Locking and unlocking the unit

● Change Access Rights of Locked level –change the access rights for lower level(s)

● Change Unit Name

● Lock Automatically after Power Cycle – The Unit can be set to lock automatically after a power cycle (be careful, if the user sets this to on he/she may be locked out after a power cycle)

● Change Unit Configuration – Change the configuration of the Network, Auto-standby, Startup preset setting. Again, mistakes here can lock a user out of the device control

● Load an empty Preset – For an experienced user, loading an empty preset can easily reset the unit for a clean slate. But for an inexperienced user, it can damage loudspeakers. So when unchecked, the user can only load presets that have been defined

● Upgrade unit firmware

● Show full graphics – when checked the full settings, like crossover, EQ, and limiter settings are visible to the user, even when they cannot be changed.

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8. Firmware Update Process

The controller firmware can be updated via the PC software. It is strongly recommended that the DSP is updated via the Ethernet interface.

The update is performed in administrator mode.
On models with a display, the firmware version can be checked via pressing the encoder button 4 times and turning the encoder until the firmware version is displayed.

You can also check the firmware version in the PC software via Help -> about -> version.
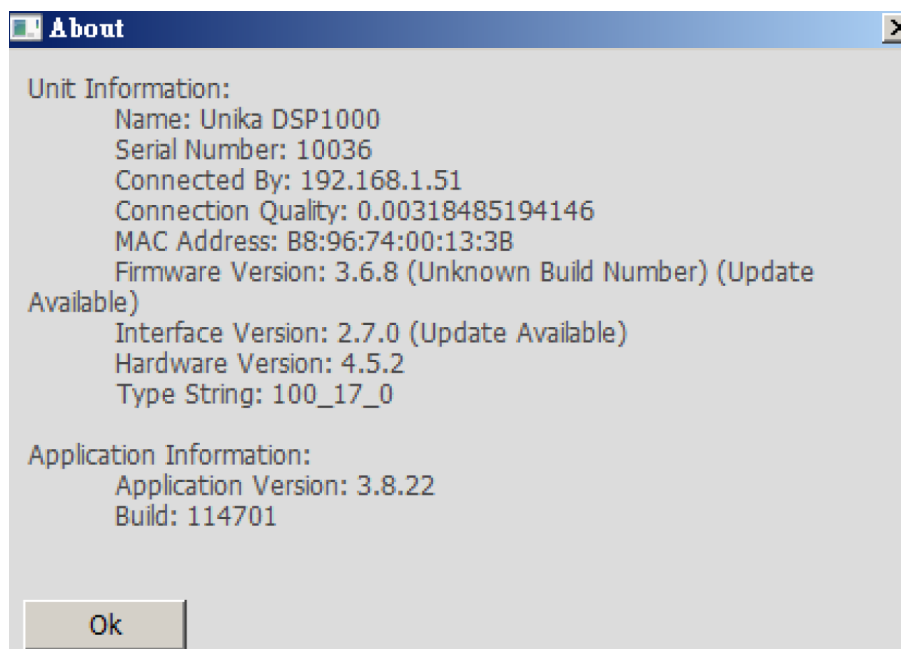

Figure 24: Update Available.

An available update is indicated by the suffix "(Update Available)". If an update is available, it can be performed as follows:
Hardware - > Firmware Update. The program may run in several stages through the update process.

If the update fails, restart the software and device and try again.
Through a firmware update (at least Admin or Developer mode), all presets from the DSP controller can be cleared. You will be prompted for the option to backup and restore all presets. If that is not done, the update removes all presets from memory.

In user-level "Developer", presets can be cleared "Preset" → "Clear all User Presets". All presets are deleted that are not protected by the "read-only preset range". To clear Read-Only Presets: In user- level "Developer", choose "Preset" → "Clear all Read-Only Presets".

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9. Front Panel Control Features

● 9.1 Function Keys and Rotary Encoder

To the right of the LCD display are function keys and a rotary Encoder, all audio channels and system parameters are selected and edited using these control keys, all parameter value can be adjusted up or down with the rotary Encoder.

The EXIT key will exit any activity and return to the top level showing the preset name and number also the device model.

Upon power ON, a status bar on the LCD show the progress of the device initialization process and after that the model name and the DEFAULT PRESET will show on the LCD screen.
The default preset is an "Empty Program" which means "flat" condition for all processes.

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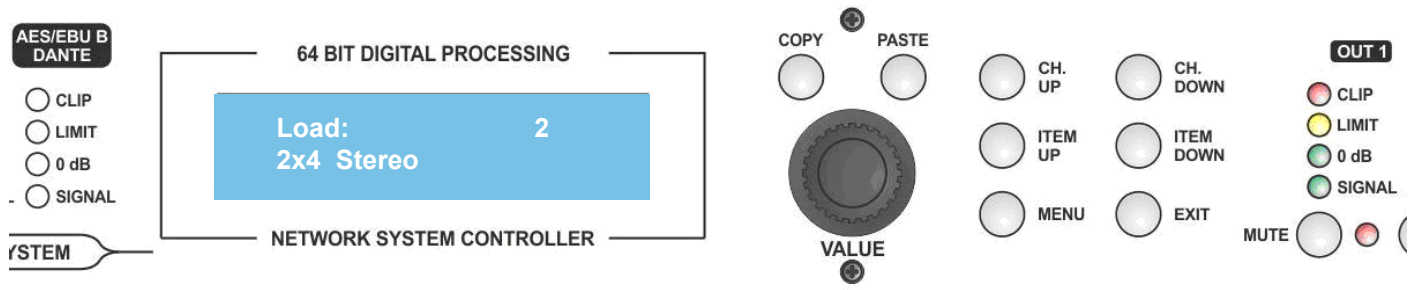## 9.2 The functionality of the relative keys will be shown in the following tables

(Table-1) The MENU key and the Rotary Encoder

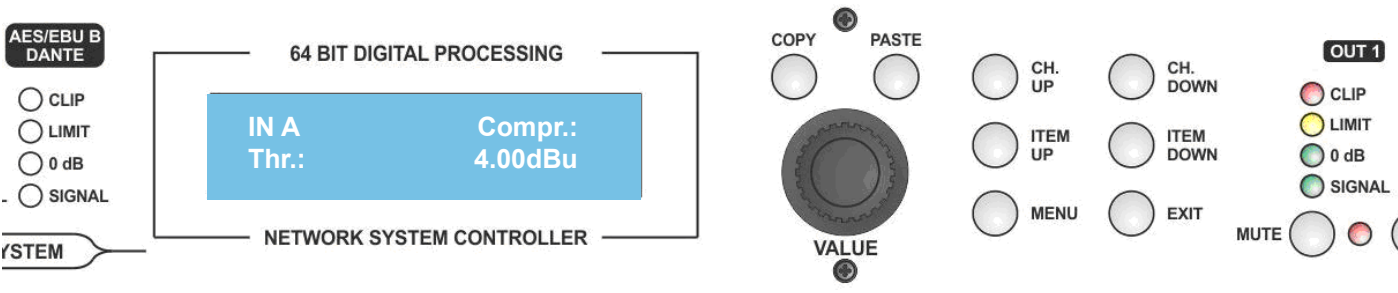| MENU (pressed to select items)	Rotary Encoder (rotated to select items)	Rotary Encoder (pressed to access)
LOAD	Rotate the encoder to select existed Presets (from 1 to 100)	1.Press and rotate to confirm Yes/No 2.Pressed to load the selected preset
SAVE	Rotate the encoder to select internal memories location (from 1 to 100)	1.Press and rotate to set Preset Name 2.Press Menu key & rotate the encoder to confirm Yes/No 3.Press Menu key or the encoder to save the current setting to the selected memories location.
ACCESS LEVEL	Rotate the encoder to select "Locked" or "Unlocked" for the device	
	When "Locked" is selected	1.Press then rotate to set password 2.Press MENU key to confirm the password 3.Press MENU key again the device will be locked
	When "Unlocked" is selected	1.Press then rotate to enter password 2.Press MENU key the device will be unlocked
VERSON INFO (the information for display only)	●Device S/N:000030xxxxx ●Firmware version:3.8.21.xxxxx ●Hardware version:4.9.x ●IP Address:169.254.xxx.xxx ●MAC Address: B896:74:01:1D:xx	●Device S/N:000030xxxxx ●Firmware version:3.8.21.xxxxx ●Hardware version:4.9.x ●IP Address:169.254.xxx.xxx ●MAC Address: B896:74:01:1D:xx
ROUTING (factory preconfigured templates for the ease of user initial setting)	●Mono 8-Way ●Stereo 1-Way ●Stereo 2-Way ●Stereo 3-Way ●Stereo 4-Way	The selected In/Out routing will not be carry out until the configuration has been saved.

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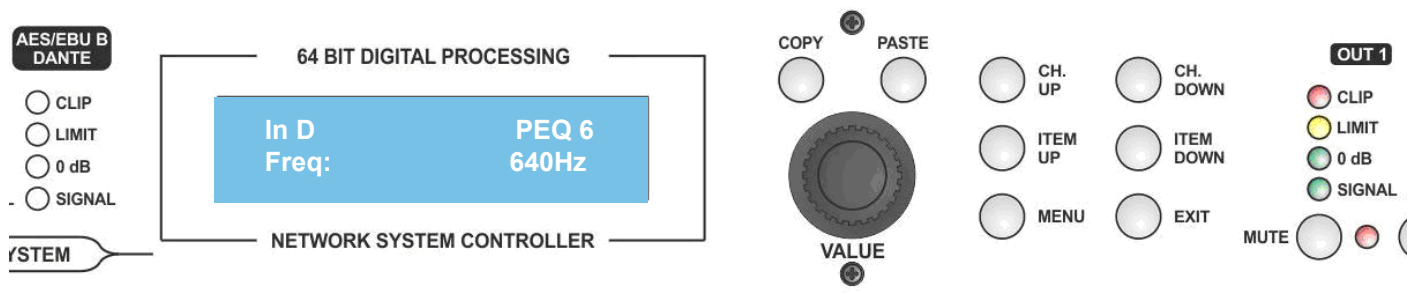
(Table-2) The **CH.UP & CH.DOWN** keys and the **ITEM-UP & ITEM-DOWN** keys

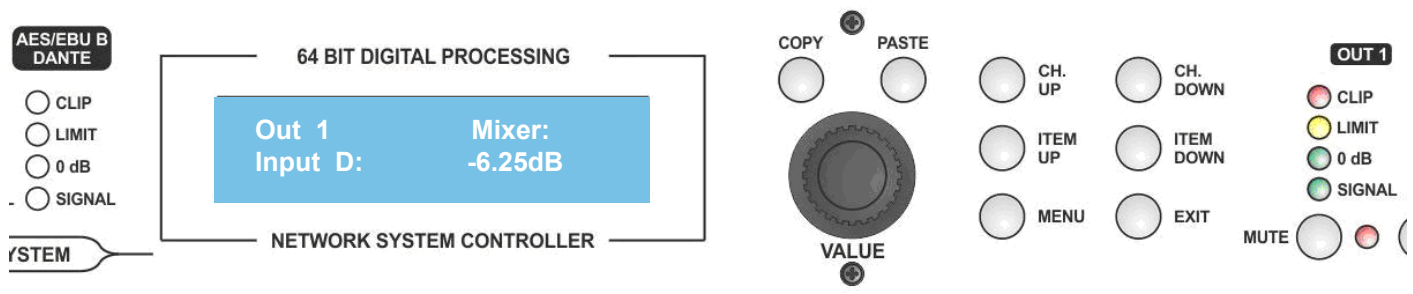| CH.UP or CH.DOWN (pressed to select items)	ITEM UP or ITEM DOWN (pressed to select items)	Rotary Encoder (Rotated to select items)	Rotary Encoder (pressed to access)
Press either one of the keys to cycle through to the Input or Output channel to be configured	Press either one of the keys to cycle through the parameters of the selected channel	Rotate the encoder to change the value of the parameter	
Input Channel Select Input A, B, C or D	**Gain**	Rotate the encoder to Mute the channel or adjust from -48dB to +12dB	
	Source	Only input channels C&D can select audio source from Analog input, digital AES3 or Dante network	(Input channels A&B and all Output channel's signal are Analog format and cannot be changed)
	Delay	Adjust the Delay from 0ms to 2000ms in 0.03ms/step	Press the encoder to change the time delay Unit as Meter, Millisecond or Feet
	Low-Pass Filter	●Adjust the cutoff Frequency (from 50Hz to 20KHz) ●Turn ON/OFF the LPF ●Select filter type: 1.Butterworth 6,12,18,24dB/oct 2.Bessel 6,12,18, 24dB/oct 3.Linkwitz Riley 12, 24dB/oct	Press the encoder to select: Frequency, Filter Type or enable ON/OFF the low pass filter
	High-Pass Filter	●Select the cutoff Frequency from 20Hz to 20KHz ●Enable ON/OFF the HPF ●Select filter type: 1.Butterworth 6,12,18,24dB/oct 2.Bessel 6,12,18, 24dB/oct 3.Linkwitz Riley 12, 24dB/oct	Press the encoder to select: Frequency, Filter Type or enable ON/OFF the high pass filter
	PEQ 1 to PEQ 10	●Select the Frequency to be adjusted from 20Hz to 20KHz ●Enable ON/OFF the PEQ. ●Select filter type: Bell, Notch, High Shelf, Low Shelf, All-pass, Bandpass, High pass, Low pass filters. ●Gain adjustment from -12dB to +12dB ●Q factor adjustable from 0.1 to 25 for filter type Bell, High Shelf, Low Shelf, All pass, Bandpass filters ●Q factor adjustable from 10 to 80 for Notch filter. ●Q factor adjustable from 0.5 to 2 for High pass & Low pass filters.	Press the encoder to select: Frequency, Filter Type, Q factor, Gain, or enable ON/OFF the Parametric Equalizer

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(Table-3) The **CH.UP & CH.DOWN** keys and the **ITEM-UP & ITEM-DOWN** keys

CH.UP or CH.DOWN (pressed to select items)	ITEM UP or ITEM DOWN (pressed to select items)	Rotary Encoder (Rotated to select items)	Rotary Encoder (pressed to access)
Press either one of the keys to cycle through to the Input or Output channel to be configured	Press either one of the keys to cycle through the parameters of the selected channel	Rotate the encoder to change the value of the parameter	
Input Channel Select Input A, B, C or D	**Compressor**	Rotate the encoder to change value: ●Threshold from -48dBu to +24dBu ●Attack time from 1ms to 10000ms ●Hold time from 1ms to 10000ms ●Release time from 1ms to 10000ms ●Compression Ratio from 1.2:1 to inf:1 ●Makeup Gain from -12dB to +12dB	Press the encoder to select: ●Threshold ●Attack ●Hold ●Release ●Ratio ●GAIN (Makeup gain)
	Limiter	Rotate the encoder to change value: ●Threshold from -48dBu to +24dBu ●Release time from 10dB/s to100dB/s	Press the encoder to select: ●Threshold ●Release
	Link	Rotate the encoder to turn ON/ OFF the Link for channels A&B or channels C&D.	

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(Table-4) The **CH.UP & CH.DOWN** keys and the **ITEM-UP & ITEM-DOWN** keys

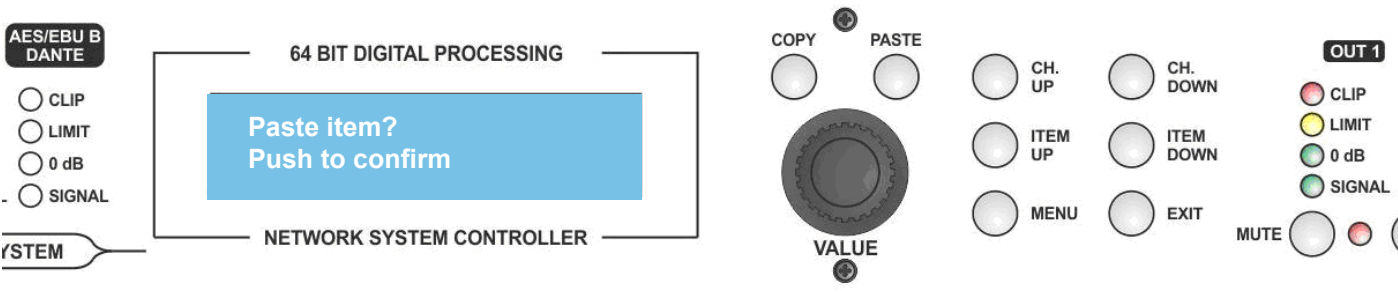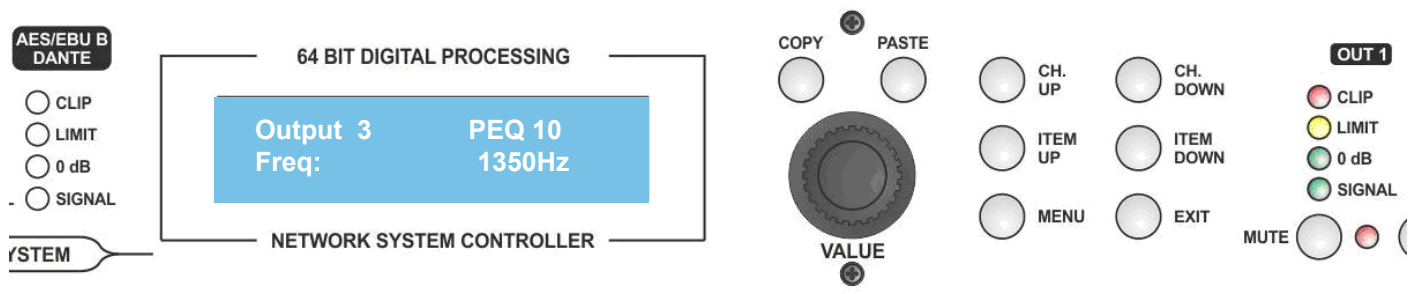| CH.UP or CH.DOWN (pressed to select items)	ITEM UP or ITEM DOWN (pressed to select items)	Rotary Encoder (Rotated to select items)	Rotary Encoder (pressed to access)
Press either one of the keys to cycle through to the Input or Output channel to be configured	Press either one of the keys to cycle through the parameters of the selected channel	Rotate the encoder to change the value of the parameter	
Output Channel Select Output 1 to 8	**Gain**	Rotate the encoder to Mute the channel or adjust from -48dB to +12dB	
	Delay	Adjust the Delay from 0ms to 2000ms in 0.03ms/step	Press the encoder to change the time delay Unit as Meter, Millisecond or Feet
	Low-Pass Filter	●Adjust the cutoff Frequency from 50Hz to 20KHz ●Turn ON/OFF the LPF ●Select filter type: 1-Butterworth 6,12,18,24dB/oct 2-Bessel 6,12,18,24dB/oct 3-Linkwitz Riley12,24dB/oct	Press the encoder to select Frequency, Filter Type or enable ON/OFF the low pass filter
	High-Pass Filter	●Select the cutoff Frequency from 20Hz to 20KHz ●Enable ON/OFF the HPF ●Select filter type: 1.Butterworth 6,12,18,24dB/oct 2.Bessel 6,12,18,24dB/oct 3.Linkwitz Riley12,24dB/oct	Press the encoder to select Frequency, Filter Type or enable ON/OFF the high pass filter
	Mixer	Rotate the encoder to adjust the level of the selected input channel that mix into the output channel	Press the encoder to select Input A, Input B, Input C or Input D

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(Table-5) The **CH.UP & CH.DOWN** keys and the **ITEM-UP & ITEM-DOWN** keys

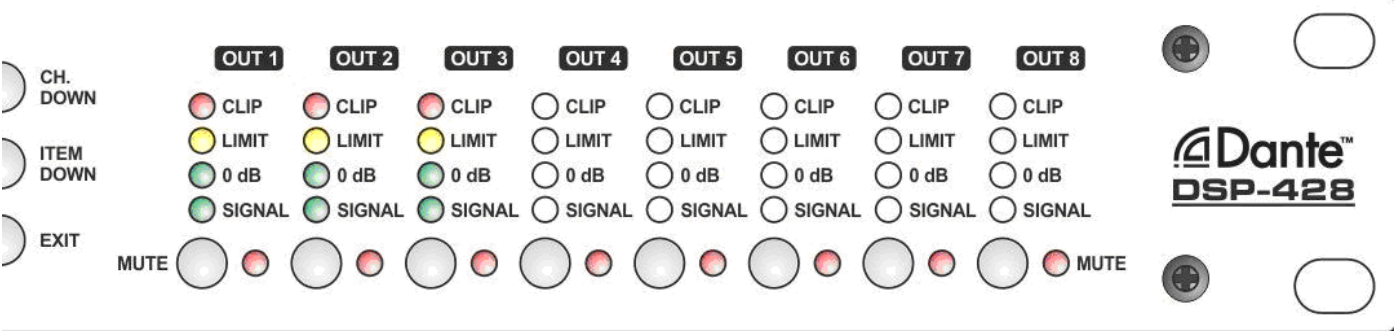| CH.UP or CH.DOWN (pressed to select items)	ITEM UP or ITEM DOWN (pressed to select items)	Rotary Encoder (Rotated to select items)	Rotary Encoder (pressed to access)
Press either one of the keys to cycle through to the Input or Output channel to be configured	Press either one of the keys to cycle through the parameters of the selected channel	Rotate the encoder to change the value of the parameter	
Output Channel Select Output 1 to 8	**PEQ 1 to PEQ 10**	●Select the Frequency to be adjusted from 20Hz to 20KHz ●Enable ON/OFF the PEQ. ●Select filter type: Bell, Notch, High Shelf, Low Shelf, All-pass, Bandpass, High pass, Low pass filters. ●Gain adjustment from -12dB to +12dB ●Q factor adjustable from 0.1 to 25 for filter type Bell, High Shelf, Low Shelf, All pass, Bandpass filters ●Q factor adjustable from 10 to 80 for Notch filter. ●Q factor adjustable from 0.5 to 2 for High pass & Low pass filters.	Press the encoder to select Frequency, Filter Type, Q factor, Gain, or enable ON/OFF the Parametric Equalizer
	Compressor	Rotate the encoder to change the value: ●Threshold from -48dBu to +24dBu ●Attack time from 1ms to 10000ms ●Hold time from 1ms to 10000ms ●Release time from 1ms to 10000ms ●Compression Ratio from 1.2:1 to inf:1 ●Makeup Gain from -12dB to +12dB	Press the encoder to select: ●Threshold ●Attack ●Hold ●Release ●Ratio ●GAIN (Makeup gain)
	Limiter	Rotate the encoder to change the value: ●Threshold from -48dBu to +24dBu ●Release time from 10dB/s to 100dB/s	Press the encoder to select: ●Threshold ●Release
	Link	Rotate the encoder to turns ON/OFF the Link of channels 1&2, 3&4, 5&6, or 7&8	
	Phase	Rotate the encoder to set the output signal phase to Normal or Inverted	

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(Table-6) COPY and PASTE keys

COPY (pressed to copy)	CH.UP / CH.DOWN ITEM-UP / ITEM-DOWN	PASTE (pressed to paste)	Rotary Encoder (paste confirm)
Press the key to copy the currently displayed item parameter values to the clipboard	Use these keys to navigate to the same item of the desired target channel to paste the data into it.	Press the key to paste the parameter values into the target channel item	Pressed the encoder to confirm completion

(Table-7) LED indicators

CLIP	The signal begins to distort when CLIP lights up
LIMIT	The Peak Limiter is activated when LIMIT lights up
0dB	The signal level is 0dBu (0.775Vrms) when 0dB lights up
SIGNAL	The signal level is -48dBu when SIGNAL lights up
MUTE	The channel has been muted when MUTE lights up

MUTE: Press the Mute button to cutoff the output signal, the red LED lit show that the channel has been muted.

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● **9.3 Features and Benefits**

● **9.3.1 Presets**

The DSP-428 is organized into 100 programmable presets, each completely defining the configuration of all inputs and outputs along with their respective audio components. DSP-428 has included 2 factory presets that preloaded into the unit which are simply starting points for common applications, and all can be modified, rename, and save to suit the end user. From the DSP Control software, a "Power ON Preset" can be configured that will be loaded upon powered up of the unit, it is the one selected out of the 100 existing presets, alternatively, a so called "Last-Setting" preset can be choose as "Power ON Preset", which is a constantly refreshed working preset used to take a "snapshot" of all current settings should the unit be turned off before changes can be saved.
Any modifications made to the currently use preset before saving, it will remain in the working preset until either the modified preset is saved, or a fresh preset is recalled.

Note: that preset recall will overwrite the working preset, so make sure the current configuration is saved before continuing or it will be lost.

When modifications made to the currently use preset without saving, on the front panel LCD an asterisk mark (＊) will show up at the right end side of the preset name for 3 seconds and the preset number will changed to an exclamation mark (!). If choosing one of the existing presets as a "Power ON Preset", beware of that if you change any setting without saving, this change will be void after power cycling the unit and the original preset will be loaded again. The presets can also be saved and recalled as preset files on the connected PC.

● **9.3.2 Input Channel A, B, C, D; Analog Audio, AES-3 Digital Input and Optional Dante Network**

The DSP-428 has provided four audio input channels, all equipped with XLR connectors, each input is processed independently and may be routed to one or several output, signal routing occurs in the output section.
Any input channel can be selected to edit its Gain, PEQ, and Delay, Compressor or Limiter functions etc.
The unit is also equipped with an AES-3 (AES/EBU) digital input and optional Dante network input, the Dante network input is fed to the unit via the same RJ-45 connector that is also used for control.
Analog audio, AES-3 digital or Dante network audio sources can be chosen for input channel C & D but input channels A & B can accepted only analog audio source.

● **9.3.3 Output Channel-1 to 8**

There are eight analog outputs on the DSP-428, all equipped with XLR connectors, each output can obtain its audio source from any input or multiple input mixes, any output channel is processed independently to edit its Source, Gain, Polarity, PEQ, Delay, Cross-over, Compressor or Limiter functions etc.

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● 9.3.4 LED Indicators

Each input and output channel has a 4 segment LED array for audio level display, ranging from -48dB (SIGNAL) through clipping (CLIP); the 0dB LED indicate that the signal level is at 0dBu (0.775Vrms), the LIMIT LED light up indicate the limiter is activated; The red LED aside the MUTE button indicate that the output channel has been muted when light up.

The LED array of AES / EBU and DANTE lights up indicate that AES-3 digital audio or DANTE network audio is connected and the audio signal is fed into the unit.

● 9.3.5 Audio Functions

● 9.3.5-1 Gain

Input and Output Gain are separately adjustable from -48dB to +12dB in 0.1dB increments. The output gain menu also provides for selection of input source(s) for a given output channel, as well as polarity of the outgoing signal. Any input or combination of inputs can be routed to any or all outputs. Beware of that two signals which share significant content, such as a stereo source, will be up to 6dB louder when combined.

● 9.3.5-2 EQ

The DSP-428 EQ section offers full parametric EQ filters on inputs and outputs, each input and output channel has 10 selectable EQ filters, filter type is selectable between Bell, Notch, Low Shelf, High Shelf, All Pass, BandPass, Low Pass and High Pass.
Filter's Center/Cutoff/Corner frequency is adjustable from 20Hz to 20 kHz; ±12dB of boost/cut control range and -40dB of attenuation range for the Notch filter.

Q factor or BW (Bandwidth) of Bell type filter is adjustable from Q= 0.2 to 25, for Notch filter Q=10 to 80, for Low & high Shelf filter Q=3 to 25, for Low-pass & High-pass filter Q=0.5 to 2, for All-pass & Bandpass filter Q=0.2 to 25.

● 9.3.5-3 Delay

Within the DSP-428, up to 2000 milliseconds of time delay are available for input channels, allowing secondary loudspeaker clusters to be time aligned with the primary speakers up to 2,252 feet (686m) away from the main stage area.

Output channels have the same time delay as well; output delay is best used to align discrete drivers within a speaker cabinet or cluster, the delay time unit can be selected as Second, Millisecond, Feet, Inch, Mil-inch, Meter and Millimeter.

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● 9.3.5-4 Crossover (X-over)

Crossover functions are available only on the output channels. Every channel's crossover consists of a high pass filter (HPF) and a low pass filter (LPF), along with the frequencies and filter types used.

The frequency range for the high pass filter (HPF) is adjustable from 20Hz to 20 kHz, with an option to turn the filter off at the low end of the frequency selection, the low pass filter (LPF) offers the same adjustable frequency range, with the "off" option at the high end of the frequency selection.

There are 10 types of filters available in the crossover section:
1-Butterworth 6dB/Oct
2-Butterworth 12dB/Oct
3-Butterworth 18dB/Oct
4-Butterworth 24dB/Oct
5-Bessel 6dB/Oct
6-Bessel 12dB/Oct
7-Bessel 18dB/Oct
8-Bessel 24dB/Oct
9-Linkwitz-Riley 12dB/Oct
10-Linkwitz-Riley 24dB/Oct

● 9.3.5-5 Butterworth Filter

Butterworth filters individually are always -3dB at the displayed crossover frequency and are used because they have a "maximally flat" passband and sharpest transition to the stopband. When a Butterworth HPF and LPF of the same crossover frequency are summed, the combined response is always +3dB.
With 12dB per octave Butterworth crossover filters, one of the outputs must be inverted or else the combined response will result in a large notch at the crossover frequency.

● 9.3.5-6 Bessel Filter

Bessel filters individually are always -3dB at the displayed crossover frequency and are used because they have a maximally flat group delay, in other words, they have the most linear phase response.
When a Bessel HPF and LPF of the same crossover frequency are summed, the combined response is +3dB for 12dB/oct and 18dB/oct Bessel filters, and -2dB for 24dB/oct Bessel filters. One of the outputs must be inverted when using either 12dB/oct or 18dB/oct Bessel crossover filters or else the combined response will have a large notch.

● 9.3.5-7 Linkwitz-Riley Filter

The 12 dB/oct and 24dB/oct Linkwitz-Riley filters individually are always -6dB at the displayed crossover frequency.
When a Linkwitz-Riley HPF and LPF of the same displayed crossover frequency are summed, the combined response is always flat. With 12dB/oct Linkwitz-Riley crossover filters, one of the outputs must be inverted or else the combined response will have a large notch at the crossover frequency.

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● **9.3.5-8 Compressor & Limiter**

A full function RMS Compressor and a Peak Limiter are included on each input and output channel.

The limiter is commonly used to prevent transient audio signal spikes from damaging loudspeakers, the compressor is used to maintain the dynamic range consistency of the audio signal and to "thicken" the sound of the audio source; the adjustable parameters include Threshold, Ratio, Attack Time, Release Time and make-up Gain.

The DSP-428 limiter threshold range is adjustable from -48dB to +24dB, this setting determines the signal level above which gain reduction begins, and is indicated by the yellow (LIMIT) LED in the output meter section.

The increase of the audio level exceeding the threshold will be limited because the limiter ratio is fixedly set to INF: 1, and the Release time can be adjusted from 10dB/sec to 100dB/sec.

The DSP-428 compressor threshold range is adjustable from -48dB to +24dB, the compression ratio control determines the amount of gain reduction above threshold; the compression ratio can range from a mild 1.2: 1 adjustment to a brick wall INF: 1.

The Attack time and Release time settings adjust the time it takes the compressor/limiter to engage and then disengage when the signal increases above threshold and then subsequently falls back below threshold.

The attack time is adjustable from 1.0ms through 10.0s, while the release time ranges from 1.0ms through 10.0s. Compressor's Hold time is adjustable from 1.0ms to 10.0s and make-up Gain from -12dB to +12dB.

● **9.3.6 Copy and Paste**

The Copy and Paste function is used to quickly transfer the settings from a currently selected item's parameter to another input or output channel.

Press the COPY key to copy the currently displayed item parameter values to the clipboard.

Use the CH.UP or CH.DOWN keys to navigate to the same item of the desired target channel to paste the data into it.

Press the PASTE key to paste the parameter value into the target channel item, and then press the encoder to confirm completion.
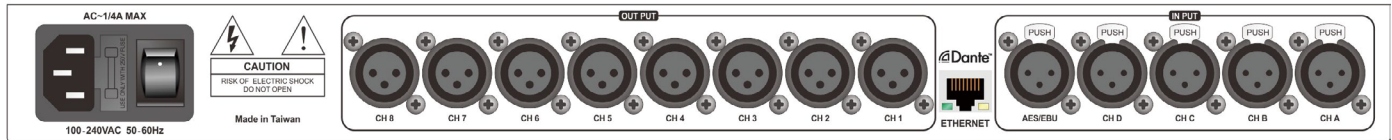
● **9.3.7 Mute**

Mute button allow muting of an Output channel, pressing the Mute button will toggle its mute function, the output's red Mute LED will lit when muted.

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● **9.3.8 Interconnection**

● **9.3.8-1 Audio Connections**

The DSP-428 analog audio connections use three pin XLR connectors, with pin 2 (+), pin 3 (-), and pin 1(G). Inputs and outputs are electronically balanced.
If an unbalanced signal is fed to an input, the signal should be on the (+) connection (pin 2) and pin 3 must be tied to ground or significant signal loss will result.
The AES-3 digital input signal can be fed via (AES / EBU) XLR socket on the controller rear panel.

Note: This unit is designed to clip at signal levels above +20dBu = 7.75Vrms which places the noise floor lower than -90dBu. The optimum input signal level which should be fed into the DSP-428 is 0dBu = .775Vrms. This input level will allow 20dB of headroom while giving a nominal signal that is >90dB above the noise floor.

● **9.3.8-2 Ethernet & Dante Network Connections**
A RJ45 Ethernet port is provided on the rear panel for networked control.
The Dante input is fed to the units via the same RJ45 connector that is also used for control.
For detailed information regarding the implementation of Ethernet control please see the UNiKA DSPControl Software section.
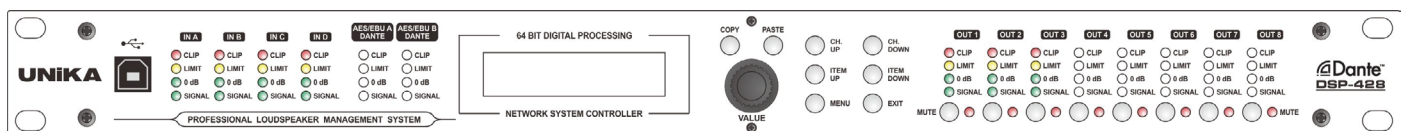
● **9.3.8-3 USB Connection**
The DSP-428 has provided an USB port on the front panel, a USB-A to USB-B cable is provided with every DSP-428 unit to connect to a computer running Unika DSPControl software.

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Specification

1. Analog Electronically Balanced inputs, 20KΩ.
2. Analog Electronically Balanced outputs, 100Ω.
3. AES-3 Digital input, 110Ω.
4. Input channel C and D can be set to accept Analog audio, AES-3 digital audio or Dante network audio.
5. 118dB dynamic range (inputs) / 114dB dynamic range (outputs) /20Hz - 20KHz unweighted.
6. Input Gain range:……...-36dB to +12dB
7. Output Gain range:……-36dB to +12dB
8. Maximum input level:…………+22dBu
9. Maximum output level:………..+24dBu
10. 64 bits digital processing.
11. Audio Sampling Rate:………. 48KHz
12. S/N Ratio 103.5dB @ 24dBu output.
13. THD<0.01% @ 1 KHz, +12dBu
14. Frequency response 10Hz to 22 kHz ±0.25dB.
15. 10 parametric filters per input and output
16. Parametric EQ filters can be set as Bell, High Shelf, Low Shelf, Notch, BandPass, High Pass, Low Pass and All-Pass.
17. Parametric EQ filter Boots/Cut range ±12dB.
18. Parametric EQ filters Q value adjustable from 0.2 to 25.
19. Parametric EQ filter frequency adjustable from 20Hz to 20KHz.
20. Notch filters Q value adjustable from 10 to 80.
21. Butterworth, Bessel, Linkwitz-Riley cross-over filters up to 24dB / Octave.
22. 2000ms Delay per input, 2000ms Delay per output, adjustable in 0.001ms/step or 0.343mm/step or 13.514mil/step.
23. RMS Compressor for each input and output:
    - Threshold range from -48dBu to+24dBu.
    - Ratio range 1.2:1 to Inf.:1.
    - Attack time range 1ms to 10sec.
    - Release time range 1ms to 10sec.
    - Compression Hold time 1ms to 10sec.
    - Makeup gain -12dB to +12dB.
24. 180° phase invert switchable for outputs.
25. Peak Limiter for each input and output
    - Threshold range from -48dBu to +24dBu
    - Release rate from 10dB/s to 100dB/s.
26. Latency 640 microseconds
27. Mute control on inputs and outputs.
28. Adjacent channels can be linked as Stereo channel.
29. 100 presets can be stored & recalled on the unit and backup to the PC.
30. Default set to load the last working setting upon startup.
31. USB and Ethernet Ports are provided for full configuration and real-time monitoring via PC.
32. A backlit 2x20 characters LCD displays channel and function settings.
33. LED indicators of Signal, Limit, Clip and Mute for all the input and output channels.
34. Languages: English and Spanish (others on request)
35. Universal Power Supply, 100-240VAC, 50/60Hz, 20Wmax.
36. Supports streaming audio of 44.1KHz/48KHz/96KHz sample rate on Dante input.
37. One 2m USB2.0 type A to type B cable, one1.6m CAT5E UTP with RJ45 connectors cable and one AC power cord will be shipped along with the device.

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PROFESSIONAL AUDIO