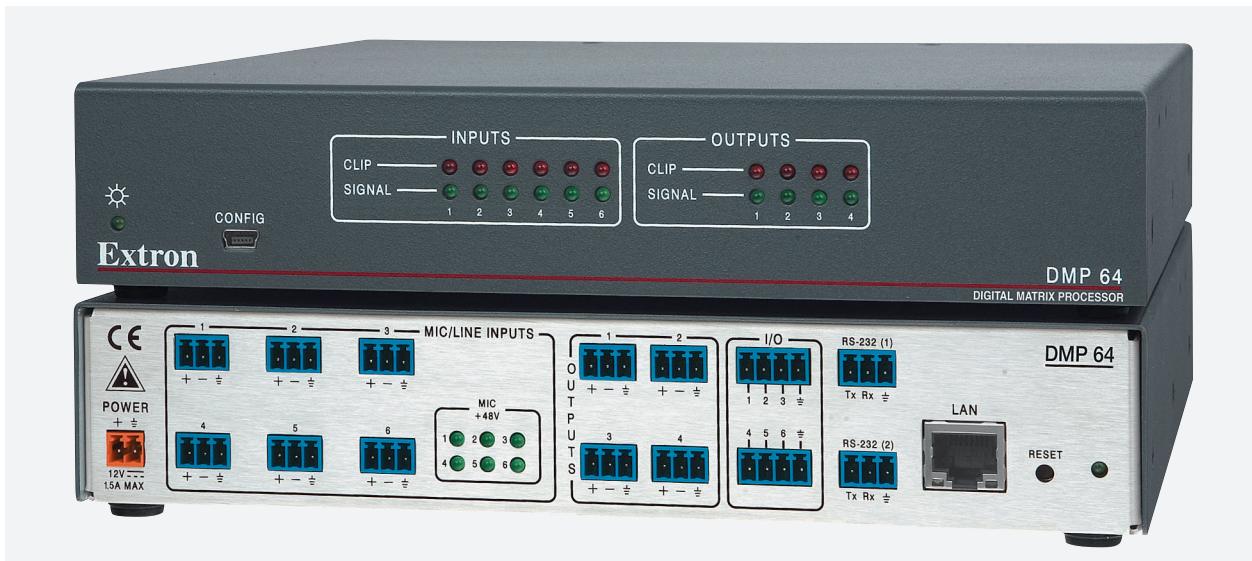


User Guide

Audio Products
Mixers and Processors

DMP 64

Digital Matrix Processor



Extron® Electronics
INTERFACING, SWITCHING AND CONTROL

Safety Instructions

Safety Instructions • English

WARNING: This symbol, , when used on the product, is intended to alert the user of the presence of uninsulated dangerous voltage within the product's enclosure that may present a risk of electric shock.

ATTENTION: This symbol, , when used on the product, is intended to alert the user of important operating and maintenance (servicing) instructions in the literature provided with the equipment.

For information on safety guidelines, regulatory compliances, EMI/EMF compatibility, accessibility, and related topics, see the Extron Safety and Regulatory Compliance Guide, part number 68-290-01, on the Extron website, www.extron.com.

Instructions de sécurité • Français

AVERTISSEMENT: Ce pictogramme, , lorsqu'il est utilisé sur le produit, signale à l'utilisateur la présence à l'intérieur du boîtier du produit d'une tension électrique dangereuse susceptible de provoquer un choc électrique.

ATTENTION: Ce pictogramme, , lorsqu'il est utilisé sur le produit, signale à l'utilisateur des instructions d'utilisation ou de maintenance importantes qui se trouvent dans la documentation fournie avec le matériel.

Pour en savoir plus sur les règles de sécurité, la conformité à la réglementation, la compatibilité EMI/EMF, l'accessibilité, et autres sujets connexes, lisez les informations de sécurité et de conformité Extron, réf. 68-290-01, sur le site Extron, www.extron.fr.

Sicherheitsanweisungen • Deutsch

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Instrucciones de seguridad • Español

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Chinese Simplified (简体中文)

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Chinese Traditional (繁體中文)

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Japanese

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Korean

경고: 이 기호 , 가 제품에 사용될 경우, 제품의 인클로저 내에 있는 접지되지 않은 위험한 전류로 인해 사용자가 감전될 위험이 있음을 경고합니다.

주의: 이 기호 , 가 제품에 사용될 경우, 장비와 함께 제공된 책자에 나와 있는 주요 운영 및 유지보수(정비) 지침을 경고합니다.

안전 가이드라인, 규제 준수, EMI/EMF 호환성, 접근성, 그리고 관련 항목에 대한 자세한 내용은 Extron 웹 사이트(www.extron.co.kr)의 Extron 안전 및 규제 준수 안내서, 68-290-01 조항을 참조하십시오.

FCC Class A Notice

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to part 15 of the FCC rules. The Class A limits provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause interference; the user must correct the interference at his own expense.

NOTE: For more information on safety guidelines, regulatory compliances, EMI/EMF compatibility, accessibility, and related topics, see the "[Extron Safety and Regulatory Compliance Guide](#)" on the Extron website.

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Conventions Used in this Guide

Notifications

The following notifications are used in this guide:

DANGER: A danger indicates a situation that **will** result in death or severe injury.

WARNING: A warning indicates a situation that has the **potential** to result in death or severe injury.

CAUTION: A caution indicates a situation that **may** result in minor injury.

ATTENTION: Attention indicates a situation that may damage or destroy the product or associated equipment.

NOTE: A note draws attention to important information.

TIP: A tip provides a suggestion to make working with the application easier.

Software Commands

Commands are written in the fonts shown here:

```
^AR Merge Scene,,Op1 scene 1,1 ^B 51 ^W^C  
[01] R 0004 003000400008000600 [02] 35 [17] [03]
```

Esc **X1** * **X17** * **X20** * **X23** * **X21** CE ←

NOTE: For commands and examples of computer or device responses mentioned in this guide, the character “Ø” is used for the number zero and “0” is the capital letter “o.”

Computer responses and directory paths that do not have variables are written in the font shown here:

```
Reply from 208.132.180.48: bytes=32 times=2ms TTL=32  
C:\Program Files\Extron
```

Variables are written in slanted form as shown here:

```
ping xxx.xxx.xxx.xxx -t  
SOH R Data STX Command ETB ETX
```

Selectable items, such as menu names, menu options, buttons, tabs, and field names are written in the font shown here:

From the **File** menu, select **New**.

Click the **OK** button.

Specifications Availability

Product specifications are available on the Extron website, www.extron.com.

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Introduction

This section describes this manual and the DMP 64, including:

- [About This Guide](#)
- [About the DMP 64 Digital Matrix Processor](#)
- [Features](#)
- [DMP 64 Application Diagram](#)

About This Guide

This guide contains installation, configuration, and operating information for the Extron Electronics DMP 64 ProDSP™ Digital Matrix Processor, software-controlled digital audio processor.

In this guide, the DMP 64 can also be referred to as “the mixer” or “device.”

About the DMP 64 Digital Matrix Processor

The DMP 64 is a standalone audio matrix processor with six microphone/line inputs and four line outputs. Using high-quality 24-bit A/D converters sampling at 48kHz, input signals are converted into the digital domain where Digital Signal Processing algorithms process and mix the signals using Extron floating point ProDSP™ technology. The DMP 64 uses a dual-matrix design providing virtual processing busses, with audio signal processing available in any of the input, virtual, and output signal paths. A dual matrix mixer with virtual paths provides extremely flexible architecture, allowing for versatile processing, mixing, and routing scenarios.

The DMP 64 is IP Link®-enabled, connecting to a host computer via the Ethernet port for fast configuration and setup. Dual RS-232 ports on the rear panel, plus a USB port located on the front panel provide convenient high-speed access. Six digital I/O ports permit connection of switches and sensors to provide input to the system for triggering a variety of actions within the device.

The DMP 64 has no front panel controls therefore, all configuration is performed using the Extron DSP Configurator™ program from a host computer connected by any of the communication ports: RS-232, USB or Ethernet. Signal present and clip LEDs for the six input channels and four output channels are provided on the front panel.

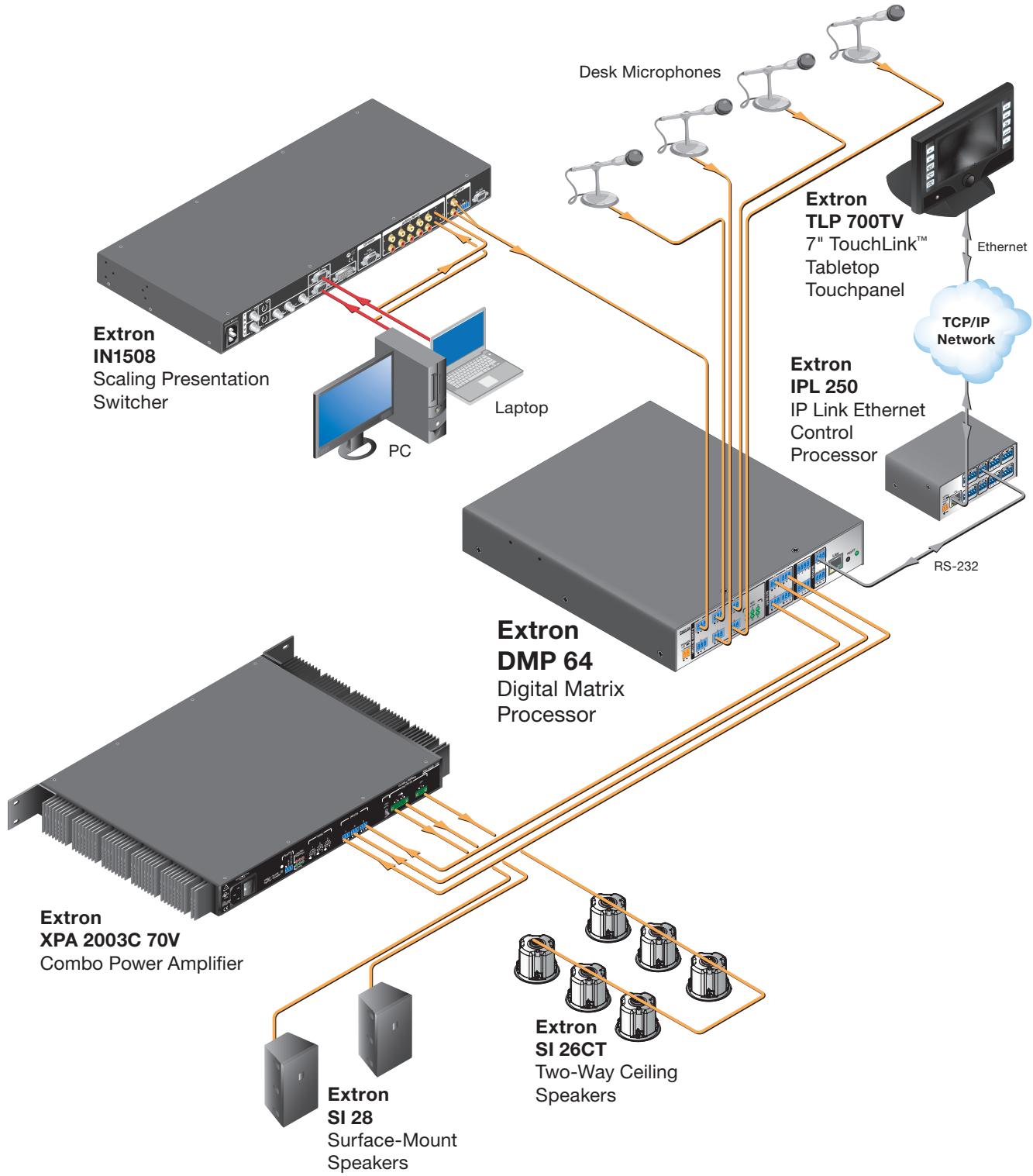
Two operational modes, Live and Emulate, allow a user to work offline from the device to set up a configuration and create presets and group controls as needed before placing the configuration in the DSP 64. DSP Configurator settings developed offline can be saved to disk as a job file to be uploaded to the device at a later time, or can be transferred directly to the device by switching to Live mode. Up to 32 full or partial presets and up to 32 group master controls can be created, loaded into, and stored in the DMP 64. The presets and controls can then be recalled through DSP Configurator, or a control system using Simple Instruction Set (SIS™) commands. Control systems connected to the device either by RS-232 or Ethernet can control a limited subset of DMP 64 functions using SIS commands.

Features

- **Consumer and professional audio compatibility** — Input and output line level can be set to consumer (-10 dBV) or professional (+4 dBu).
- **Inputs** — Six balanced or unbalanced mic/line on 3.5 mm, 3-pole captive screw connectors.
- **Outputs** — Four balanced or unbalanced on 3.5 mm, 3-pole captive screw connectors.
- **ProDSP™ audio signal processing** — Features 32- and 64-bit floating point audio DSP processing providing wide dynamic range and maintaining audio signal transparency while preventing DSP signal clipping and simplifying management of gain staging.
- **Pro-grade hardware** — Studio grade 24-bit, 192 kHz analog-to-digital and digital-to-analog converters sampling at 48 kHz.
- **Low latency DSP processing** — The DSP engine supports a large array of concurrent audio processing within an audio channel and across multiple channels, while maintaining extremely low latency from input to output.
- **DSP Configurator software** — Powerful, user-friendly PC-based software tool for managing all audio operations of the DMP 64. Enables complete setup and configuration of digital audio processing tools on the ProDSP platform, as well as routing and mixing.
- **Intuitive graphical user environment** — The DSP Configurator Software features a Graphical User Environment with a clear view of all input and outputs, audio processing blocks, routing, mix points, and virtual routing in a single window. This allows a designer or installer to quickly view all audio activities without having to access multiple windows or menus.
- **SpeedNav keyboard navigation** — SpeedNav enables user-friendly, keyboard-based navigation of the DSP Configurator Software without the need for a mouse or touchpad. Using keyboard navigation keys and shortcuts, a user can access any input or output, mixing points, and all audio DSP tools. Using only the keyboard for software access can help expedite audio system setup and commissioning while on-site using laptop PCs.
- **Copy and paste for processing blocks** — To help speed audio system design and setup, parameter settings can be quickly copied between individual processing blocks or identical groups of blocks within the Graphical User Environment, using conventional cut-and-paste commands.
- **32 DSP Configurator presets** — Using the DSP Configurator software, parameters for DSP processing, signal levels, or audio routing can be saved as presets. These settings can be saved for the entire system, or any selected group of inputs, outputs, mixing points, and DSP blocks.
- **Six digital I/O ports for remote control or feedback** — Six configurable digital I/O ports are provided that enable the DMP 64 to sense and respond to external triggers such as mic activation and muting.
- **Dual matrix design** — The DMP 64 employs a dual matrix design with substantial flexibility to rout, mix, and process audio input sources. A primary matrix routes each input to any or all four outputs. If desired, any of the six inputs can first be directed into a secondary matrix, which routes the inputs to four virtual buses before being mixed back to the outputs via the primary matrix. Virtual buses allow for inputs to be grouped together and then processed with the same DSP settings and parameters, simplifying system setup and control.

- **Group masters** — The DMP 64 provides the capability to consolidate gain or mute control throughout the system. Any gain or mute block within the Graphical User Environment can be selected and added to a group master, which can then be controlled by a single master fader and mute control. Each group master can have up to 16 blocks, and up to 32 group masters can be created.
- **Soft limits** — Soft limits can be applied to group master faders. Minimum and maximum limits can be specified and controlled via RS-232 serial control.
- **Source signal presence and clipping LEDs** — The DMP 64 provides indicator LEDs on the front panel for each input and output providing real-time monitoring of signal presence. A separate set of LEDs illuminates as a warning when signal clipping is detected.
- **Flexible control options** — The DMP 64 can be controlled using the DSP Configurator Software and a PC connection to the IP Link Ethernet port, the RS-232 serial port, or the USB 2.0 port on the front panel.
- **Front panel USB configuration port** — Enables configuration without having to access the rear panel.
- **RS-232 serial control port** — Using serial commands, the DMP 64 can be controlled and configured via the Extron Windows®-based control program, or integrated into third-party control systems using Extron Simple Instruction Set (SIS™) commands. With two RS-232 serial ports plus the IP Link Ethernet port, the DMP 64 offers possibilities for control in single and divisible room applications.
- **Two RS-232 ports** — The DMP 64 is equipped with both primary and secondary RS-232 serial ports for divided room applications.
- **IP Link® Ethernet monitoring and control** — Engineered to meet the needs of professional A/V environments, IP Link enables the DMP 64 to be proactively monitored and managed over a LAN, WAN, or the Internet, using standard TCP/IP protocols.
- **Versatile mounting options** — Rack-mountable 1U, half rack width metal enclosure.
- **Universal power supply** — The 100-240 VAC, 50-60 Hz, international power supply provides worldwide power compatibility.

DMP 64 Application Diagram



Installation

This section describes the installation of the DMP 64, including:

- **Mounting the DMP 64**
- **Rear Panel Features and Cabling**

Mounting the DMP 64

The 1U high, half rack width, 9.5 inch deep DMP 64 Digital Matrix Processor can be:

- Set on a table,
- Mounted on a rack shelf,
- Mounted under a desk or tabletop, or
- Mounted on a projector bracket.

For detailed mounting options and UL rack mounting guidelines (see **Mounting the DMP 64** on page 139).

Rear Panel Features and Cabling

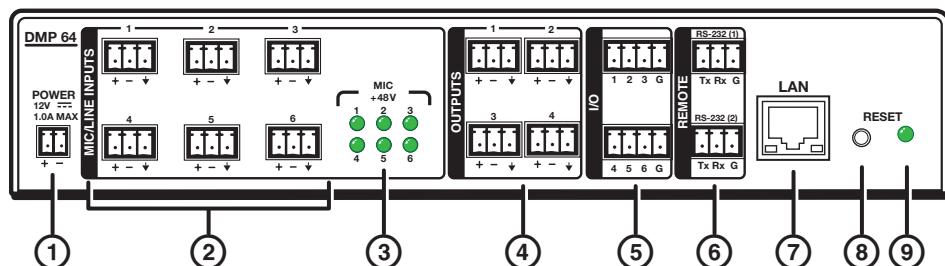


Figure 1. DMP 64 Rear Panel

NOTE: Control signal ground pins may be labeled as \pm or “G”. Audio ground pins may be labeled as \pm or \downarrow .

The wiring and function are the same, whichever way your product is labeled.

- ① Power connector** — Connect the included 12 VDC external power supply into the 2-pole 3.5 mm captive screw connector. Be careful to observe the correct polarity.

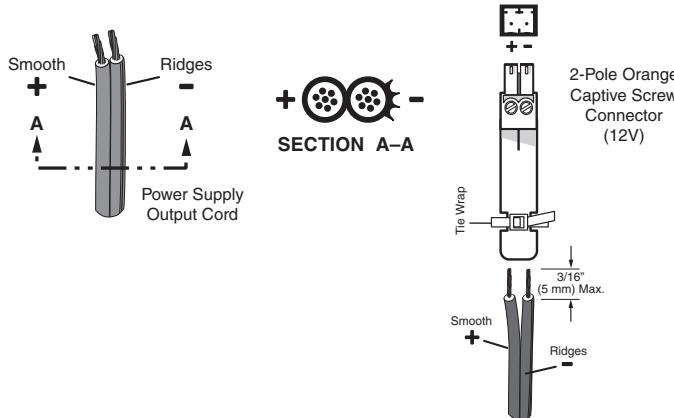


Figure 2. Power Supply Wiring

Use the supplied tie-wrap to strap the power cord to the extended tail of the connector.

ATTENTION:

- The two power cord wires must be kept separate while the power supply is plugged in. Remove power before wiring.
- Always use a power supply specified by Extron Electronics for the DMP 64. Use of an unauthorized power supply voids all regulatory compliance certification and may cause damage to the supply and the DMP 64.
- When connecting the power supply, voltage polarity is extremely important. Applying power with incorrect voltage polarity could damage the power supply and the DMP 64. Identify the power cord negative (ground) lead by the ridges on the side of the cord or a black heat shrink wrapping around it.
- Unless otherwise stated, the AC/DC adapters are not suitable for use in air handling spaces or in wall cavities. The power supply is to be located within the same vicinity as the Extron AV processing equipment in an ordinary location, Pollution Degree 2, secured to the equipment rack within the dedicated closet, podium or desk.
- The installation must always be in accordance with the applicable provisions of National Electrical Code ANSI/NFPA 70, article 75 and the Canadian Electrical Code part 1, section 16. The power supply shall not be permanently fixed to building structure or similar structure.

NOTES:

- The length of the exposed (stripped) copper wires is important. The ideal length is 3/16 in (5 mm). Longer bare wires can short together. Shorter wires are not as secure in the direct insertion connectors and could be pulled out.
- Do not tin the stripped power supply leads. Tinned wires are not as secure in the captive screw connectors and could be pulled out.
- To verify the polarity before connection, check the no load power supply output with a voltmeter.
- To avoid losing adjustments when configuring the DMP 64 via SIS commands, issue a 2FF or if using the Extron DSP Configurator, select **Tools > Save changes to device** to store the latest changes to the device. Wait several minutes **after** saving the adjustments **before** disconnecting power.

- ② Mic/Line 1-6 input connectors** — 3-pole 3.5 mm double-stacked captive screw connectors accept balanced or unbalanced mono mic or line level signals. Mic/line inputs provide gain settings to accommodate consumer (-10 dBV) and professional (+4 dBu) operating line level sources, plus microphone level sources. Up to six mono microphones or line inputs, balanced and unbalanced in any combination can be connected to these inputs. See the following diagram for wiring instructions.

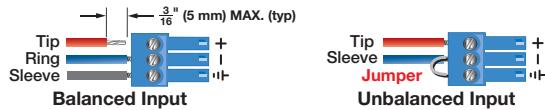


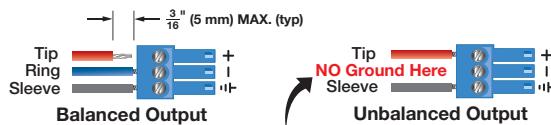
Figure 3. Balanced or Unbalanced Mic and Line Input Wiring

- ③ Phantom Power indicators** — These green LED indicators light when +48 V phantom power is placed on the corresponding mic/line input. Phantom power is not adjustable.

ATTENTION:

- Condenser microphones require phantom power. Dynamic microphones **do not** require power.
- Never set an unbalanced dynamic microphone to **48 V**. Doing so can damage the microphone. For condenser mics, verify the microphone will safely operate at +48 VDC.

- ④ Mono output connectors** — 3-pole 3.5 mm captive screw connectors provide balanced or unbalanced connections for mono line level output signals.



ATTENTION: For unbalanced audio, connect the sleeve to the ground contact.
DO NOT connect the sleeve to the negative (-) contacts.

Figure 4. Output Connector Wiring

- ⑤ Digital I/O output connectors** — A double-stacked 4-pole 3.5 mm captive screw connector provides six configurable digital input or output ports allowing connection to various devices such as motion detectors, alarms, lights, LEDs, buttons, photo (light) sensors, or temperature sensors.

Digital I/O ports are used to monitor or drive TTL level digital signals. The inputs can be configured to operate in one of two modes: digital input or digital output. In OUTPUT mode, the device can source up to 250 mA at +5 V. In INPUT mode, voltages greater than 1 V indicate a logic 'high' signal while voltages less than 1 V indicate a logic 'low'.

All digital I/O ports are tied to a common ground (one common ground for each 4-pole connector), but can be individually configured to operate in one of two modes: digital input or digital output.

NOTE: These ports can be configured via the DSP Configurator (see [Digital I/O Ports](#) on page 75 for additional information).

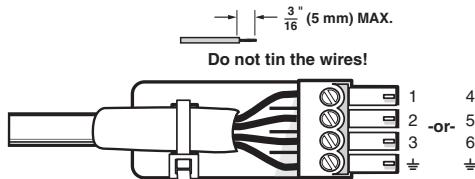


Figure 5. Digital I/O Wiring

- ⑥ RS-232 connector** — Two stacked 3-pole 3.5 mm captive screw connectors, labeled RS-232 (1) and RS-232 (2), are available for bi-directional RS-232 (± 5 V) serial control. Default baud rate is 38400.

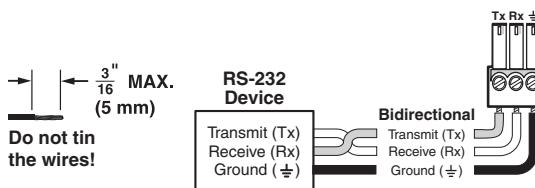


Figure 6. RS-232 Wiring

- ⑦ LAN (RJ-45) connector** — A standard RJ-45 jack accepts an RJ-45 plug for Ethernet connection.
- A yellow (ACT) LED indicates data activity on the connection.
 - A green (Link) LED indicates the jack is connected properly to the network.
- See [SIS Programming and Control](#) on page 92 for additional information on Ethernet cabling.
- ⑧ Reset button** — The reset button is used to return the DMP 64 to different tiers of default states and to place the unit into an event recording mode for troubleshooting (see [DMP 64 Hardware Reset Modes](#) on page 138).

USB Configuration Port (Front Panel)

A front panel configuration port uses an Extron USB A Male to USB Mini B Male Configuration Cable (part number **26-654-06**) for connection to a PC computer and the USB port (see [Install the USB Driver](#) on page 17 for USB driver installation details).

Hardware Operation

This section describes the the operation of the DMP 64, including:

- [DMP 64 Operation](#)
- [Front Panel Operation](#)
- [Rear Panel Operation](#)

DMP 64 Operation

The DMP 64 does not have physical controls. Configuration and operation are accomplished using a PC running Windows XP or newer and the DSP Configurator software (available on the included disc or at www.extron.com), an embedded web page using Windows Internet Explorer, or the Extron Simple Instruction Set (SIS™) using hyper-terminal or DataViewer.

The DMP 64 has several front and rear panel operational indicators described in the following pages.

Front Panel Operation

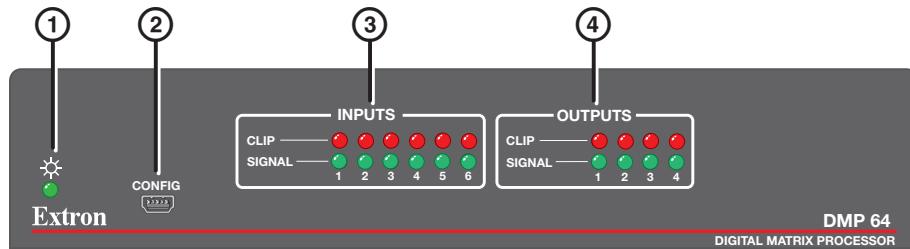


Figure 7. DMP 64 Front Panel

- ① **Power LED** — The power indicator lights when the DMP 64 is operational.
- ② **Configuration connector** — The USB 2.0 port uses a mini type-B connector to connect to a host computer for control. The DMP 64 USB driver must be installed prior to using the port (see [Install the USB Driver](#) on page 17).

The DMP 64 appears as a USB peripheral with bi-directional communication. The USB connection can be used for software operation (see [Windows-based Program Control](#) on page 15), and SIS control (see [Software Control](#) on page 14).
- ③ **Input Indicators** — Stacked red (signal clipping) and green (signal present) LEDs for inputs 1–6 . Each column represents one input channel.

The green signal LED varies in brightness corresponding to the real-time input signal level. It begins to light at –60 dBFS increasing in fifteen steps to full intensity as the signal level increases. When the signal reaches –3 dBFS or above, the red clipping LED lights and remains lit as long as the signal remains above –3 dBFS. When it falls below that level, the red LED remains lit for 200 milliseconds, after which the display resumes real-time monitoring of the signal level.
- ④ **Output Indicators** — Stacked red (signal clipping) and green (signal present) LEDs for outputs 1 through 4. Each LED column represents one output channel.

The green signal LED varies in brightness corresponding to the output signal level. It begins to light at –60 dBFS increasing to full intensity corresponding to signal level increases. When the signal level reaches –3 dBFS or above, the red clipping LED lights and remains lit as long as the signal remains above –3 dBFS. When it falls below that level, the red LED remains lit for 200 milliseconds, after which the display resumes real-time monitoring of the signal level.

Rear Panel Operation

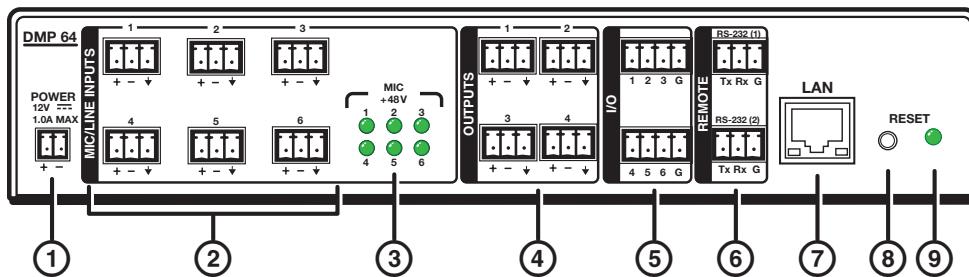


Figure 8. DMP 64 Rear Panel

① ② ④ ⑤ ⑥ See [Rear Panel Features and Cabling](#) on page 5 for details.

③ **Phantom Power indicators (MIC +48V)** — These green LED indicators light solid when +48 V phantom power is placed on the corresponding mic/line input. Phantom power is not adjustable.

ATTENTION:

- Condenser microphones require phantom power. Dynamic microphones **do not** require power.
- Never set an unbalanced dynamic microphone to **+48V**. Doing so can damage the microphone. For condenser mics, verify the mic will safely operate at +48 VDC.
- When a line level source is connected, be certain the +48V phantom power is off (unchecked).

⑦ **LAN** — The LAN connector has a green LED that lights solid to indicate proper connection to an active LAN and a yellow LED that blinks to indicate data activity.

⑧ **Reset** — The reset actuator initiates system resets (see [Reset Actuator and LED Indicator](#) on page 12).

⑨ **Power/Reset LED** — The green LED indicator adjacent to the reset button duplicates the front panel LED operation (see [Reset Actuator and LED Indicator](#) on page 12).

Power Cycle

Current mixing and audio processor settings (the current state of the device) are saved in nonvolatile memory. When the unit is powered off, all settings are retained. When the unit is powered back on, it recalls settings from the nonvolatile memory. If a configuration was in process during the power down, these saved mix, audio level, and audio DSP processor settings become active.

On power up the unit performs a self-test. The front and rear power indicator LEDs flash during the test, then light solid when the unit is available for operation or programming.

Firmware Updates

The firmware of the DMP 64 can be updated through Ethernet, USB, or RS-232 connection. The user can obtain new firmware from the Extron website, or from an Extron Applications Engineer via e-mail. After obtaining the new firmware, upload it to the unit via the served web pages (see [HTML Operation](#) on page 120), using the **Firmware Loader** in the DSP Configurator program (see [DMP Software](#) on page 14), or using the Extron standalone Firmware Loader software application available on the included disc or at www.extron.com.

Reset Actuator and LED Indicator

A recessed button (⑧) on the rear panel initiates several reset modes. The rear panel LED (⑨) blinks to indicate the reset mode.

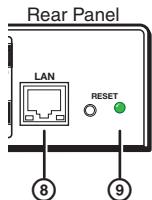


Figure 9. Reset button and LED

Hardware Reset Modes:

NOTE: The reset modes listed below will close all open IP and Telnet connections, and close all sockets.

The reset modes have separate and distinct functions. Additional information is available (see [DMP 64 Hardware Reset Modes](#) on page 138).

Each reset LED flash lasts for 0.25 seconds.

MODE 1 — Firmware reset:

1. Disconnect power to the DMP 64.
2. Press and hold the reset button.
3. Apply power to the device while holding the reset button.

Mode 1 will:

- Return the firmware to the version shipped with the unit from the factory. This allows recovering a unit with incorrect or corrupt firmware.
- Maintain all user files and settings.

NOTE: Some user web pages may not work correctly if returning the unit to an earlier firmware release.

MODE 3 — Events reset:

1. Apply power to the DMP 64.
2. Press and hold the reset button until the reset LED blinks once (~3 seconds). Release the reset button,
3. Within one (1) second press the reset button again to toggle events on or off, depending on the current state.
 - If event logging is currently stopped, following the momentary (<1 sec.) press, the reset LED will flash twice indicating events logging has started.
 - If any events are currently running, following the momentary press, the reset LED will flash three times indicating that events logging has stopped.

If a momentary press does not occur within 1 second, the events logging status before entering reset will remain.

MODE 4 — IP Address reset:

- 1.** Apply power to the DMP 64.
- 2.** Press and hold the reset button about 6 seconds until the reset LED blinks twice. Release the reset button.
- 3.** Within (1) second, press the reset button again to reset the IP settings.
If a momentary press does not occur within 1 second, the reset will be ignored.

Mode 4 will:

- Enable ARP program capability
- Set the IP address back to factory default (192.168.254.254)
- Set the Subnet back to factory default (255.255.0.0)
- Set the Gateway back to factory default (0.0.0.0)
- Set the Digital I/O Port mapping back to factory default
- Turn DHCP off
- Turn Events logging off

MODE 5 — Factory default reset:

- 1.** Apply power to the DMP 64.
- 2.** Press and hold the reset button until the reset LED blinks 3 times (~9 seconds), then release.
- 3.** Momentarily (<1 second) press the reset button to return the DMP 64 to factory default conditions.

If a momentary press does not occur within 1 second, the reset is exited.

The default (reset) state of the device is:

- All mix-points set to 0 dB gain and muted.
- All outputs active (unmuted, 100% volume).
- No inserted or active DSP processing.
- All audio inputs are set to 0 dB gain and muted.
- All preset and group master memory is clear (empty).

Digital I/O Ports

The dual 4-pin Digital I/O ports monitor or drive TTL level digital signals. The ports consist of two banks of three I/Os with the fourth pin used as a ground, providing six ports total. The DSP Configurator software provides selection from a list of scripts, that can be loaded to the DMP 64. The scripts provide pre-configured sets of functions.

From the main structure menu, click **Tools > Configure Digital IO** to access the scripts (see [Digital I/O Ports](#) on page 75).

DMP Software

This section describes the control software for the DMP 64, including:

- [Software Control](#)
- [Embedded Web Pages](#)
- [Windows-based Program Control](#)
- [DSP Configurator Program Basics](#)
- [Audio level, Mix-point, Processing Blocks, and Signal Chains](#)
- [Mic/Line Input Signal Controls](#)
- [Ducker Tutorials](#)
- [Line Output Channels](#)
- [Virtual Bus Returns](#)
- [Primary Mix Matrix](#)
- [Secondary Mix Matrix](#)
- [Group Masters](#)
- [Digital I/O Ports](#)
- [Emulate Mode and Live Mode](#)
- [Presets](#)
- [Protected Configuration](#)
- [DSP Configurator Windows Menus](#)
- [Optimizing Audio Levels](#)

Software Control

The DMP 64 can be controlled using the DSP Configurator software, SIS commands through hyper terminal or DataViewer, or using embedded WebPages. IP Link functions will be available through network connection including global viewer functionality.

The DMP 64 has the following connection options:

- **RS-232** — One single stack 3-pole, 3.5 mm captive screw connector is used for bi-directional RS-232 (± 5 V) serial control.
See [Rear Panel Features and Cabling](#) on page 5, for additional details on connecting the RS-232 port.
- **LAN** — 10 Mbps, 100 Mbps, half duplex, full duplex connections are supported. Two LEDs indicate connection and activity status. The device has the following default Ethernet configurations:

IP Address: 192.168.254.254

Default Gateway: 0.0.0.0

Subnet Mask: 255.255.0.0

DHCP: OFF

See [Rear Panel Features and Cabling](#) on page 5, and [Connection Options](#) on page 92 for additional details on connecting the LAN.

- **USB 2.0** — A Mini B-type USB connector located on the front panel provides high-speed USB 2.0 connectivity to a host computer, backward compatible to 1.0.

Embedded Web Pages

The embedded web pages, accessible by LAN using a web browser, include the following information, available in a tabbed interface.

- **System Status** — The opening web page, displaying a report of system status parameters.
- **Configuration** — This tab contains the following menu items.
 - System Settings. Contains IP address and date/time settings.
 - Passwords. Enter/re-enter admin and user password fields to set up password protected access.
 - Firmware Upgrades. Browse/upload firmware to the device.
- **File Management** — Delete or upload files
- **Control** — Contains the following menu items:
 - Audio Settings. Includes mix matrix, input and output gain control.
 - Group Controls. Provides access-only to the group controls.
 - Presets. Used to save new presets created on Audio Settings page.

See [HTML Operation](#) on page 120 for further details.

Windows-based Program Control

The DSP Configurator Control Program is compatible with Windows 2000, Windows XP, and Vista, and provides remote control of the input gain/attenuation, output volume output adjustment, and other features.

DSP Configurator can control the DMP 64 using any of the three control ports, RS-232, USB, or LAN.

Updates to this program can be downloaded from the Extron website at www.extron.com.

Installing the DSP Configurator Program

The program is contained on the Extron Software Products disk.

Install the software as follows:

1. Insert the disk into the drive

- Click the **Software** tab or software icon.

NOTE: If the DVD setup program does not start automatically, run **Launch.exe** from the DVD ROM directory using Windows "My Computer".



Figure 10. DVD Software Menu

- Scroll to the DSP Configurator program and click the **Install** text to its right.

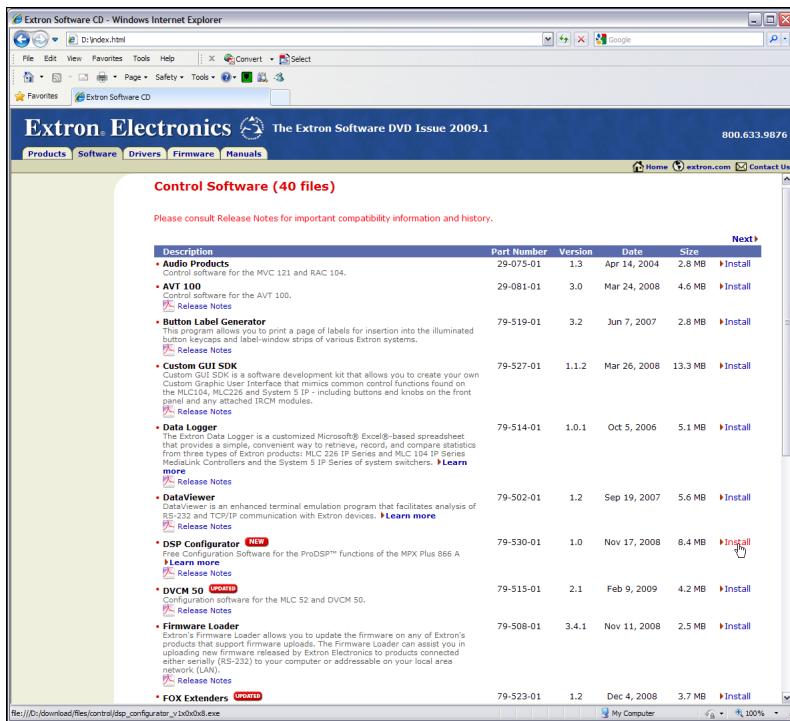


Figure 11. DVD Control Software Menu

- Follow the on-screen instructions. By default, the installation creates a **C:\Program Files\Extron\DSP_Configurator** folder for the DSP Configurator program.
- When the DSP Configurator installation is complete, the USB Installer starts automatically (see **Install the USB Driver** on page 17). Extron recommends the USB drivers be installed whether they are used immediately or not.

Install the USB Driver

When the USB installer begins, follow these instructions.

1. When the driver installation window appears (see figure 12), click **Next** to proceed.

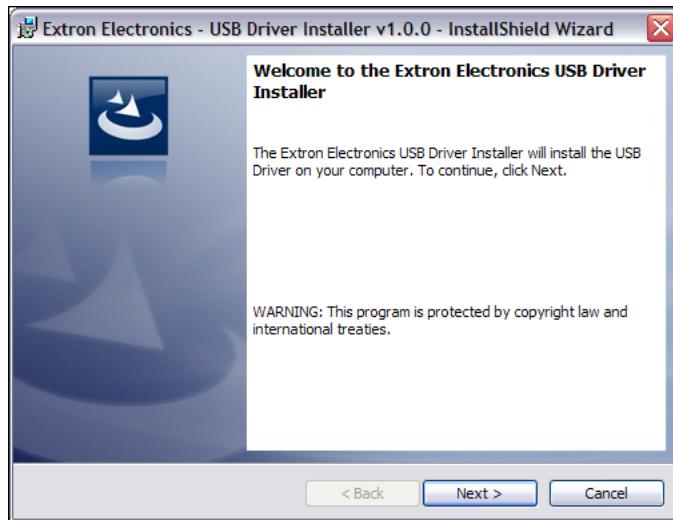


Figure 12. USB Installer Splash Screen

2. The driver installer launches (see figure 13).



Figure 13. USB Driver Installation

3. When the installer has completed the installation of the USB drivers, the following screen appears (see figure 14):

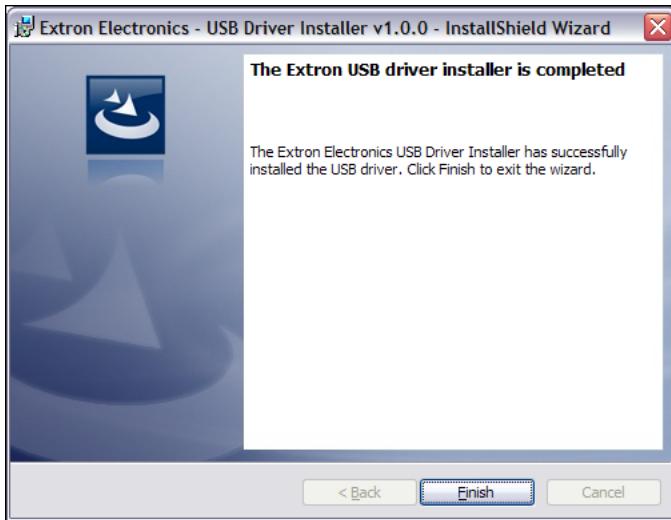


Figure 14. Successful USB Driver Installation

4. Click **Finish**.

USB driver installation is complete.

DSP Configurator Program Basics

Starting the Program

NOTE: Extron recommends connection via the Ethernet LAN port for running the DSP Configurator program.

To run the DSP Configurator Program, click

Start > Programs > Extron Electronics > DSP Configurator > DSP Configurator.



The DSP Configurator program starts in **Emulate** mode (see figure 15, next page). Also (see **Emulate Mode and Live Mode** on page 76) for details of mode operation.

Using the Program

In **Emulate** mode, audio parameters can be selected, then transferred to the DMP 64 by switching to **Live** mode (while connected to a DMP 64). Audio settings can also be tailored while connected to the DMP 64 which allows real-time auditioning of the audio output as adjustments are made (see **Emulate Mode and Live Mode** on page 76).

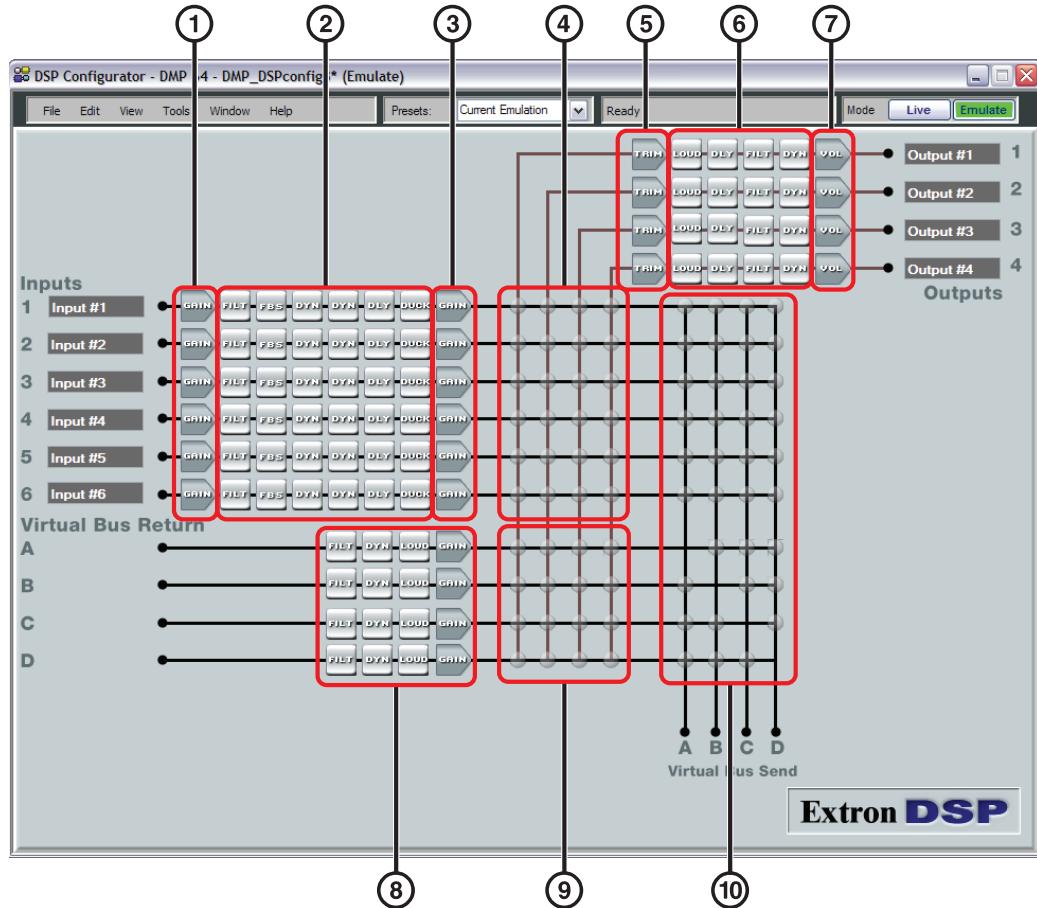


Figure 15. DMP 64 Configurator Program

The DSP Configurator program window consists of an input and virtual return signal processor chain, the main mixer, virtual send and receive (secondary) mixers, and an output signal processing chain.

- | | |
|--|--|
| <ul style="list-style-type: none"> ① Mic/Line Input Gain control ② Mic/line input signal processor chain ③ Mic/Line input pre-mixer gain ④ Primary Mix-points ⑤ Output trim control (post-mixer trim) | <ul style="list-style-type: none"> ⑥ Output signal processor chain ⑦ Output volume control ⑧ Virtual Bus signal processor chain ⑨ Virtual Bus Return (primary) mix-points ⑩ Virtual Bus Send (secondary) mix-points |
|--|--|

Navigation

There are two methods of navigation around the interface:

- Keyboard
- Mouse

One element in the user interface always retains focus. When a new DSP Configurator file is opened, the upper left element (Input #1 Gain) is focused by default.

Keyboard Navigation

All user interface elements, including mix-points, have the ability to receive focus using the tab and arrow keys or using the arrow keys following a single left-click. For additional details (see [Keyboard Navigation](#) on page 85).

Mouse Navigation

Left-click. A single click brings focus to a processor block, as well as other interface elements such as tabs, sliders, check boxes. Other left-click actions follow the Windows standard.

Right-click. A single right-click brings up a context menu specific to the processor block right-clicked. Other right-click actions follow the Windows standard.

Double-click. A double-click opens a dialog window from either the focused or unfocused state of an element.

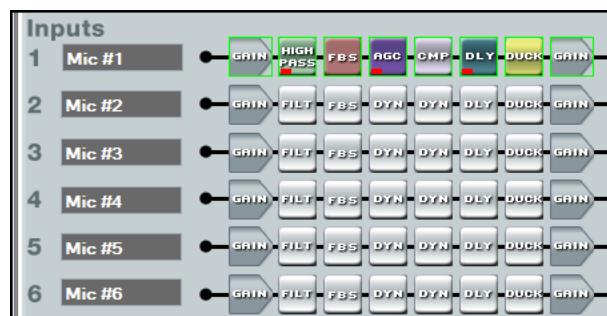
Cut, Copy, or Paste Functions

The user can cut, copy, or paste a processor. The actions can be performed from a:

- Context menu accessed by a right-click of the processor,
- Using the Edit menu,
- Using standard Windows keystrokes:
 - <Ctrl+X> = cut
 - <Ctrl+C> = copy
 - <Ctrl+V> = paste

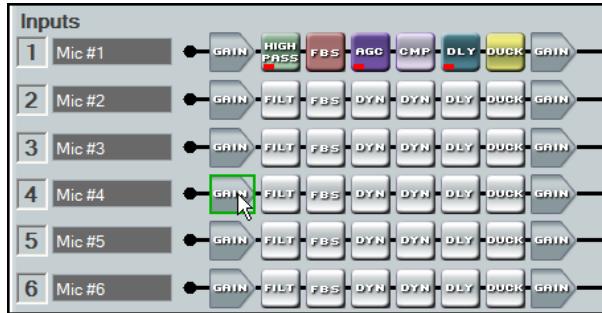
Multiple elements can be acted upon but the blocks copied must be compatible with the desired paste blocks. A highlighted group of elements can be cut or copied to a clipboard. The clipboard contents can then be pasted, but only succeeds if there is an exact one-to-one relationship between the clipboard contents and the block or blocks pasted.

In the following example, the Mic #1 input signal path is copied to Mic #5. First, click the mouse and drag it across the entire signal path. The selected blocks are highlighted in green. Press <Ctrl+C>, or use the **Edit > Copy** menu selection to copy the blocks.

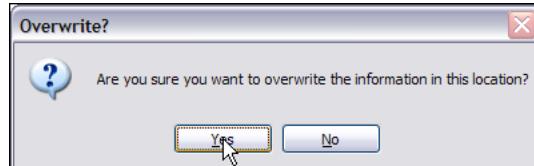


As shown below, the starting point for the paste, (the upper, leftmost element), must first be focused by left-clicking the mouse on it. Note the green focus outline that appears on the Mic #4 Gain block. The clipboard elements are pasted using the context menu **Paste** command, the **Edit>Paste** command, or <Ctrl+V>.

NOTE: A cut and copy of elements can be pasted to multiple locations. To copy the clipboard to an additional location, click on the leftmost block and paste again.



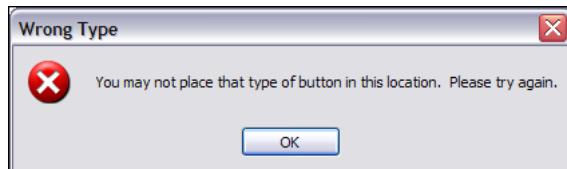
The program warns that all settings in the section being pasted to will be overwritten:



After clicking **Yes**, the entire Mic #4 input path is now identical to the Mic #1 input path including signal levels, parameter settings, and mute/bypass selections.

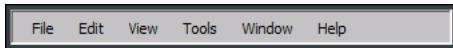


Any single processor block can be copied, then pasted to a similar processor block in the same or different input, virtual or output signal path. Mix-point gains can be copied from one to another; however, input gain, pre-mixer gain, post-mixer trim, and output volume cannot. Mix-point settings can be freely copied between mix-points. The user is always asked whether they want to overwrite the existing information. If an attempt is made to copy a processor block setting to an incompatible block, the user is advised the action cannot be completed.



DSP Configurator Toolbar Menus

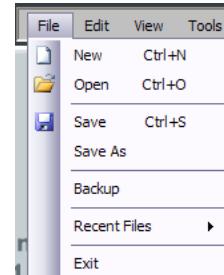
The DSP Configurator contains the following structural menus, arranged horizontally below the title bar:



File

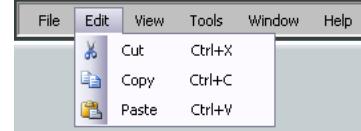
NOTE: **New**, **Open**, and **Recent Files** are unavailable in Live mode.

- **New** — Discards the current DSP configuration (after prompting to save any changes) and opens a blank configuration file.
- **Open** — Loads and activates a previously saved DSP configuration file.
- **Save** — Saves all changes to the current DSP configuration file under the current file name. If the file has not previously been saved, prompts for a file name.
- **Save As** — Saves all changes to the current DSP configuration file under a new file name.
- **Backup** — Transfers all partial presets plus the current configuration to a DSP configuration file within the DSP Configurator program.
- **Recent Files** — Opens a list of recently opened or saved DSP configuration files.
- **Exit** — Closes the DSP Configurator Program.



Edit

- **Cut** — Removes all parameters of a selected processor block or set of selected blocks to the clipboard. If not followed by a **Paste** command to a different block, the parameters are restored.



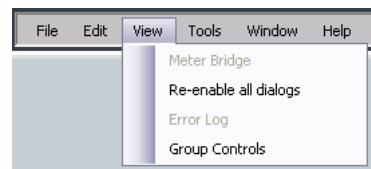
NOTE: Processor blocks are not removed from the processor stream after a **Cut** and a subsequent **Paste** operation. Only the parameters are moved.

Processor blocks and their parameters can be pasted only into another block of the same type. For example, the input 1 filter block and all of its parameters can be copied to the input 2 filter block but not to the input 1 delay block.

- **Copy** — Copies all of the parameters of a selected processor block, gain block, or set of selected blocks to the clipboard.
- **Paste** — Inserts processor blocks and their parameters from the clipboard into the selected location.

View

- **Meter Bridge** — Opens a **Meters** dialog box with real-time meters that monitor signal levels at each input and output.



NOTE: **Meter Bridge** is available in Live mode only while connected using the LAN port.

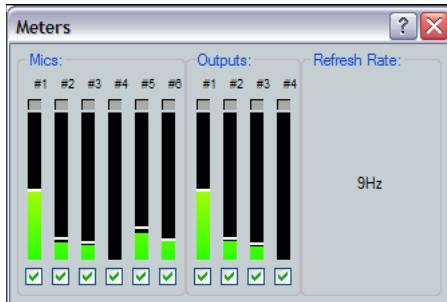


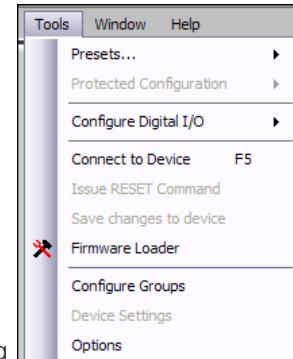
Figure 16. Meter Bridge

- **Re-enable all dialogs** — Re-enables all dialog boxes, the pop-up windows that allow changes to block parameters.
- **Error log** — Lists error messages as a troubleshooting tool.
- **Group Controls** — Opens the Group Controls dialog box (see **Group Masters** on page 69).

Tools

The Tools menu contains the following items and sub-menu:

- **Presets** — Provides three options:
 - **Mark All Items** — Mark (select) all parts of the current configuration (excluding presets), including processors and mix-points to save as a partial preset.
 - **Save Preset** — Save the currently marked processors, and mix-points as a partial preset.
 - **Clear Marked Items** — Unmark (deselect) all parts of the current configuration (excluding presets), including processors and mix-points.
- **Protected Configuration** — **Live** mode only. Allows a user (typically the installer) to save and recall a protected configuration. The protected configuration is useful to establish parameters and values (with the exception of the device IP address) in a known state, either as a troubleshooting tool or as a baseline configuration. A protected configuration, once saved in the device, is always present and cannot be overwritten without entering a user-defined Personal Identification Number (PIN) password. The protected configuration is restored without a PIN.



NOTE: The default PIN is **0000**.

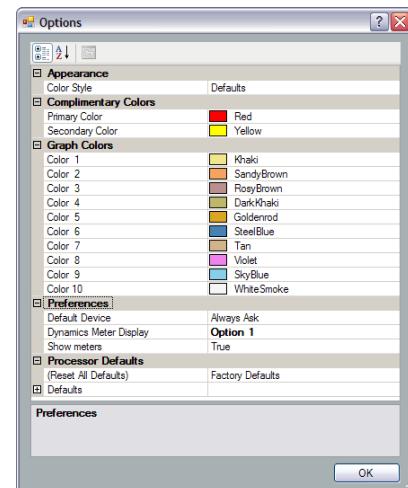
- **Save** — Save the current configuration (excluding presets), including processors and mixes as a password protected configuration. The DSP Configurator program prompts for a PIN to save.
- **Recall** — Recall the protected configuration.
- **Change PIN** — Change the PIN associated with the protected configuration.

- **Configure Digital I/O Ports** — Live mode only. Opens a utility to configure digital I/O ports. The DMP 64 provides six digital I/O ports that can be used to trigger external events from DMP 64 actions, or for external events to trigger DMP actions (see [Digital I/O Ports](#) on page 75).
- **Connect to/Disconnect from Device** (depending on **Emulate** or **Live** mode) — Performs the same functions as the Mode **Emulate** and Mode **Live** buttons.
- **Issue RESET Command** — Initializes and clears the following: mix-points, presets, processor blocks, and gain blocks. This reset is identical to the **Esc ZXXX← SIS** command (see [SIS Programming and Control](#) on page 92).
- **Save changes to device** — **Live** mode only. Saves configuration changes made in the DSP Configurator program to the DMP 64.
- **Firmware Loader** — Calls the Firmware Loader program, which allows updates without taking the DMP 64 out of service (see [Firmware Loader](#) on page 136).
- **Configure Groups** — Opens the configure groups dialog box (see [Group Masters](#) on page 69).
- **Device Settings** — Live mode only. Opens a dialog box providing a means to change the IP address, set administrator and user passwords, and select the serial port baud rate.
- **Options** — Opens a tabbed dialog box to customize the DSP Configurator appearance and operation.

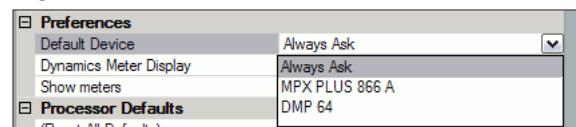
- **Colors** — Tailor the appearance of the various graphs and dialog boxes.

Appearance uses a selected color scheme for the complimentary and graph colors.

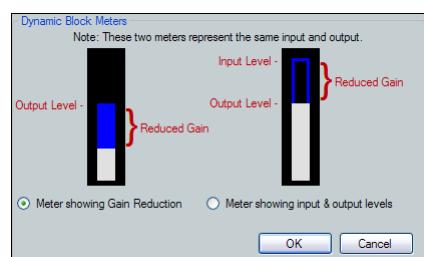
Complimentary Colors allows custom selection of colors used with the various graphs and dialog boxes. **Graph colors** change the row colors containing the information and descriptions of the graphs seen in the processor blocks.



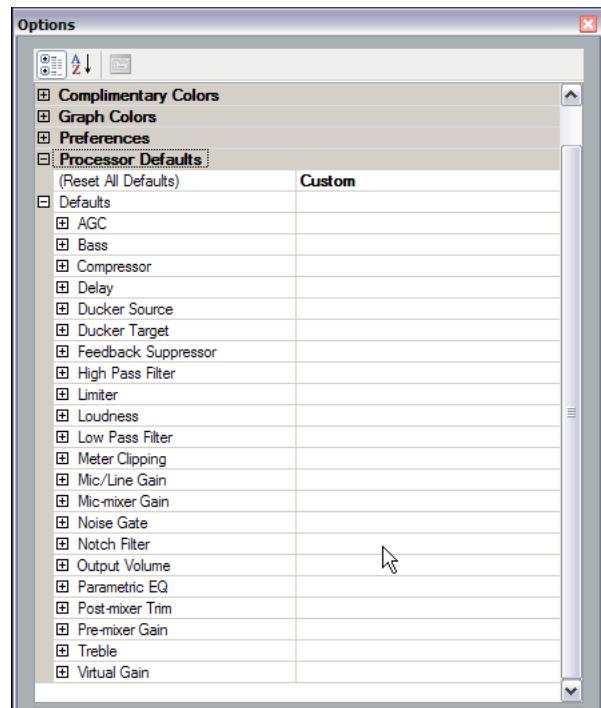
- **Preferences** — The startup splash screen contains options to select the device to connect to, or to “Always ask” on startup. The selection can be changed using **Default Device**.



- If **Show Meters** is set to **True**, **Dynamic Block Meters** is used to tailor the appearance of the dynamics meters to use the full meter to show input and gain reduction, or to show the level based on the output and gain reduction.

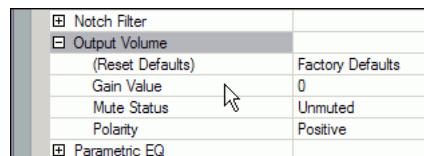


- Processor Defaults, Reset All Defaults** — Returns the DMP 64 processor and level control blocks to factory default settings. Each processor and gain/volume/trim block also has an individual default reset.
- Processor Defaults, Defaults** — Individually selects the default parameters for the various processor, trim, and gain blocks. Each row item contains default settings customized for the processor, filter, trim, or gain block it represents. Gain and volume blocks can be initially muted, while filter and dynamics processor blocks can be initially bypassed.



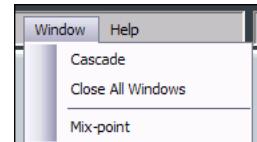
NOTE: The bypass function is labeled **Enable**.

- To view the individual processor defaults, press the + button on the name of the processor, trim, gain or meter device.



Window Menu

- Cascade** — Rearranges all open DSP Configurator program windows, including dialog boxes, in a cascading array.
- Close All Windows** — Closes all open dialog boxes.
- Individual Windows** — Brings the associated dialog box to the front of the desktop.



Help selection

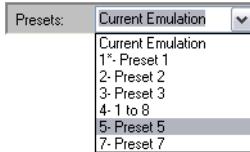
The **Help** menu contains the following elements:

- Contents** — Opens the Help file at the **Contents** tab.
- Search** — Opens the Help file at the **Search** tab.
- About...** — Displays the name of the application, the current version number, and copyright information.

NOTE: Help can be activated by pressing the <F1> key from the main screen.
When a dialog box is open the <F1> key opens context-specific help. .

Presets Drop-down

Displays a list of up to 32 presets. Select a preset from the list to display it in the window and either activate it (**Recall**), abort the selection without either recalling or deleting it (**Cancel**), or delete it (**Delete**).



NOTE: An asterisk in the drop-down list indicates a partial preset exists only in the DMP 64 and has not been uploaded to the DSP Configurator.

Mode Buttons

Provides selection between **Live** mode and **Emulate** mode. See [Emulate Mode and Live Mode](#) on page 76 for more information.



Backup

When in Live mode (connected to a DMP 64), if presets exist in the DMP 64 that are not present in the DSP Configurator program (indicated by an asterisk next to the preset name), the function halts and prompts the user to run a backup.

Backup (**File>Backup**) transfers all partial presets plus the current configuration from the DMP 64 to a DSP configuration (.edc) file within the DSP Configurator program and then displays a prompt to save the file to the hard drive. Backup is unavailable when the DSP Configurator program is in Emulate mode.

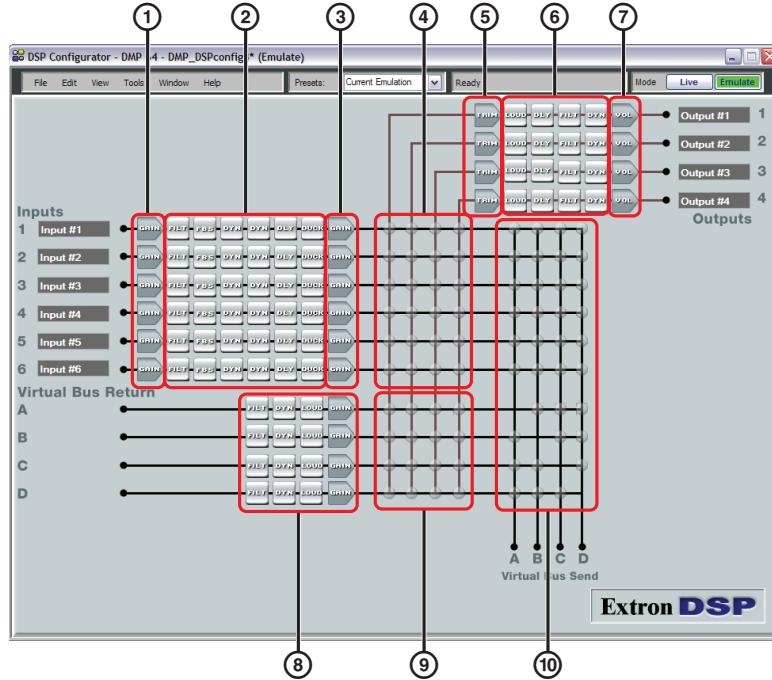


Figure 17. Control Blocks and Processor Chains

Audio Level, Mix-point, Processing Blocks, and Signal Chains

Outlined in red above (see figure 17), all control blocks on the main DSP user interface have one of three main functions in the overall signal chain:

- Level control (gain, trim, and volume),
- Mix-point (signal routing), or
- Signal processing (filter, feedback, dynamics, delay, duck, and loudness).

The signal chain varies depending on whether it is in the input, output, or virtual bus stage. The input chain begins with a level control (GAIN), filter (FILT), feedback suppression (FBS), two dynamics (DYN) and a delay (DLY) processor, followed by ducking (DUCK) and a pre-mixer gain (GAIN) control. The output chain begins with a level control (post-mixer trim [TRIM]), loudness (LOUD), delay (DLY), filter (FILT), and dynamics (DYN) processing blocks, and an output volume control (VOL). Each virtual bus chain has a filter (FILT), a dynamic processing block (DYN), loudness (LOUD), and output trim control (GAIN). All mix-points have a gain control.

Each of the three signal processing chains; Input (①, ②, ③), Output (⑤, ⑥, ⑦), and Virtual (⑧) (see figure 17), consist of a series of control blocks of two basic types specific to that chain: level control (gain, trim, and volume control), and signal processors (frequency filters, feedback suppression, dynamics, delay, ducking, and loudness). Both types of control blocks are always present in the chains. Gain controls default to unmuted and processor blocks are bypassed upon insertion.

Gain, trim and volume blocks can be muted and processor blocks (after being inserted) can be bypassed for signal comparison. Mutes and bypasses are shown by a red indicator in the lower left of the block.



Figure 18. Input Gain Control Muted, Dynamics Processor Bypassed

Level Control Blocks

To access a gain, trim or volume control to view a setting, make a change, or observe a live audio meter (input gain and output volume blocks only), double-click the gain block icon (see figure 19). This action opens a dialog box that contains the fader for that control.

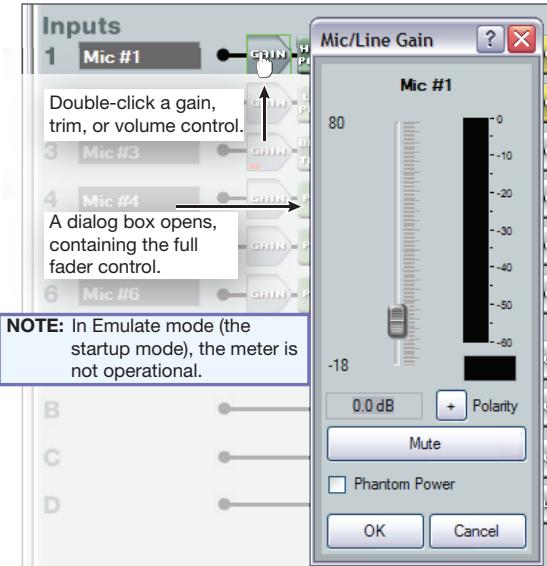


Figure 19. Accessing a Typical Gain Control Dialog Box

Processor Blocks

Each processor block represents a menu of one or more processors that can be inserted into the audio stream. For blocks that provide more than one processor, only one can be selected. Each block can be inserted by a double-click or right-click>Insert then selecting the desired processor (see figure 20). When a block is inserted, the selected processor is displayed in the block and the block changes color. Processor blocks default to bypassed. To have them default to “not bypassed,” see [Tools](#) on page 23.

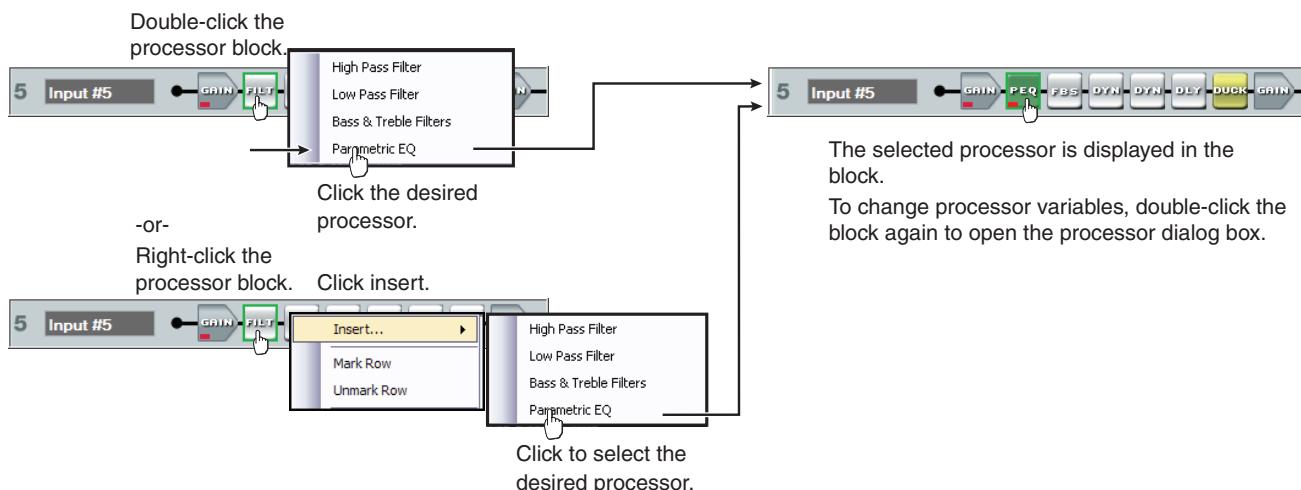


Figure 20. Selecting a Processor Block

Once a processor is inserted, to view associated parameters that define the selected processor (such as a frequency curve) or to remove the bypass, double-click on the processor block. This action opens a new window with a dialog box that contains parameters for the processor (see figure 21).



Figure 21. Sample Processor Dialog Box

- The **Set Defaults** button discards all custom settings and reloads the default parameters.
- The **Bypass** button temporarily suspends the processing without removing the processor block. Red indicates the processor is bypassed.

By default, each processor block is bypassed when inserted (the **Bypass** button in the processor dialog box is red). This can be changed for each processor block type, see **Tools>Options** and the specific defaults for the processor types.

NOTE: Figure 21 is an example of one type of dialog box. Contents and appearance of each dialog box are unique to the processor type.

The block can be removed from the signal chain by selecting it with a single mouse click and pressing the keyboard <Delete> key or by right-clicking and selecting **Delete**.

Mic/Line Input Signal Controls



The input signal processor chain allows adjustments to program or microphone audio material before input to the main mixer.

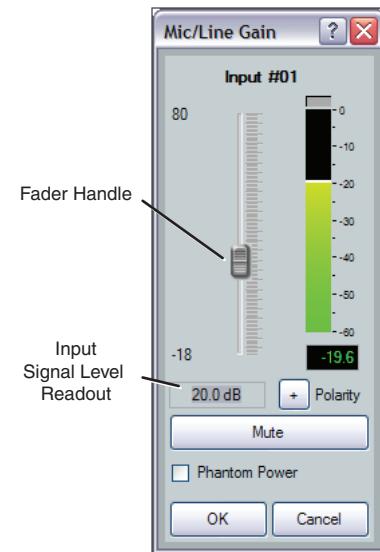
Gain Control (GAIN)

The gain control provides a single long-throw fader with a range of -18 dB to +80 dB, adjustable in 1 dB increments with the fader, or in 0.1 dB increments using direct entry in the input signal level readout below the fader. The peak reading meter holds the peak level for one second, displaying it numerically in the box below the meter. The default setting is unity gain (0.0 dB).

The **Phantom Power** checkbox, accessible in the dialog box, toggles the +48 VDC phantom power on and off. Phantom power is typically used to power a condenser microphone.

The **Mute** button, accessible in the dialog box, silences the input.

The **Polarity** button, accessible in the dialog box, allows the polarity of the wires connected to the audio connectors (+/tip and -/ring) to be flipped to correct for miswired connectors.



Filter (FILT)

Each filter block allows a total of five filters. The first filter is inserted from a processor list that appears when the block is double-clicked or from a context window that shows a processor list when the block is right-clicked.

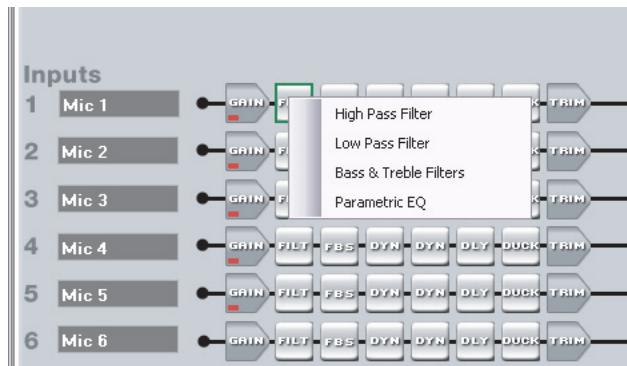


Figure 22. Insert Filter Menu

Once inserted, double-click the processor block to change parameters of the filter. After the first filter is inserted, up to four additional filters can be added to the filter block using the dialog box. Select the desired filters from the following list using the drop-down boxes:

- **High pass filter** — A high pass filter passes a band of frequencies extending from a specified cutoff frequency (greater than zero) up toward the high end of the frequency spectrum. All frequencies above the specified cutoff frequency are allowed to pass, while all frequencies below are attenuated.
The default cutoff is 100 Hz.
- **Low pass filter** — A low pass filter passes a band of frequencies extending from a specified cutoff frequency (less than infinite) towards the lower end of the frequency spectrum. All frequencies below the specified frequency are allowed to pass, while all frequencies above are attenuated.
The default cutoff is 10 kHz.
- **Bass and treble filters** — Also known as shelving or tone controls, the separate bass and treble filters provide the ability to cut or boost gain linearly above or below a specific frequency, with the end-band shape giving the visual appearance of a shelf.
The bass default frequency is 100 Hz and the treble default is 8 kHz.
- **Parametric equalizer filter** — The parametric filter is a frequency equalizer that offers control of all parameters, including amplitude (the amount of gain/boost or gain reduction/cut applied), center frequency (frequency), and range of affected frequencies (Q) around the center frequency.

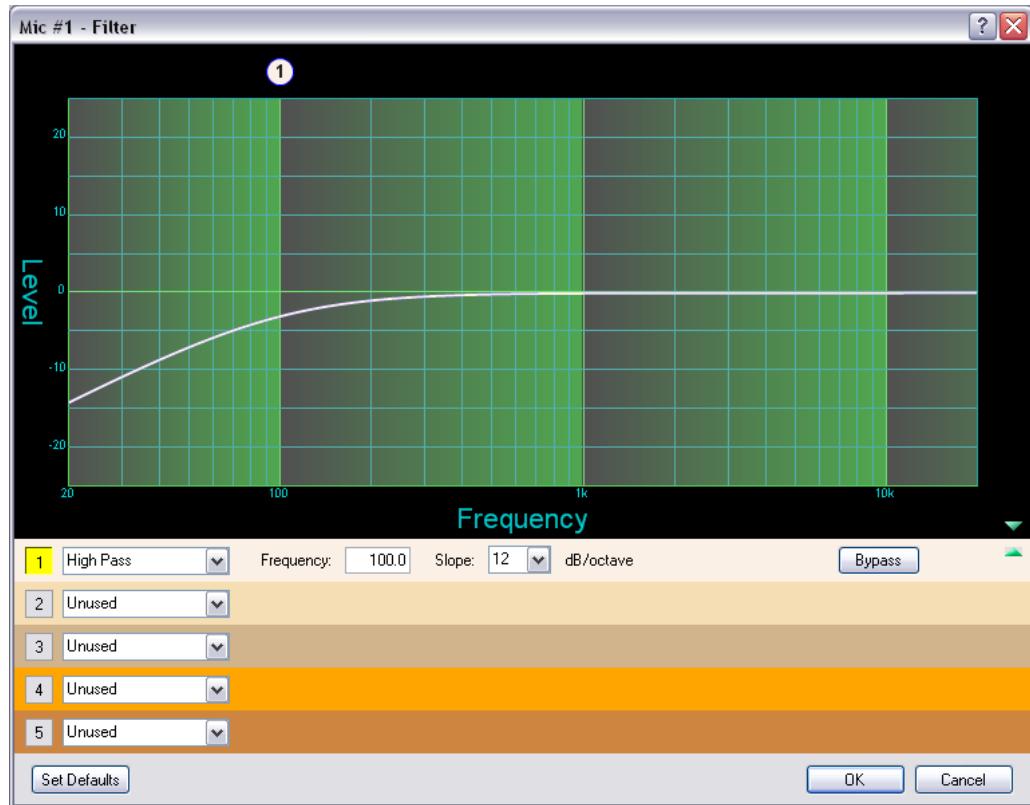


Figure 23. Filter Block Dialog Box

Additional filters are inserted using the open filter block dialog box, and selecting a filter type from the drop-down filter selection list. All filter parameters are modified using the Filter block dialog box. Each filter is loaded with all applicable default parameters displayed to the right of each drop-down filter selection list.

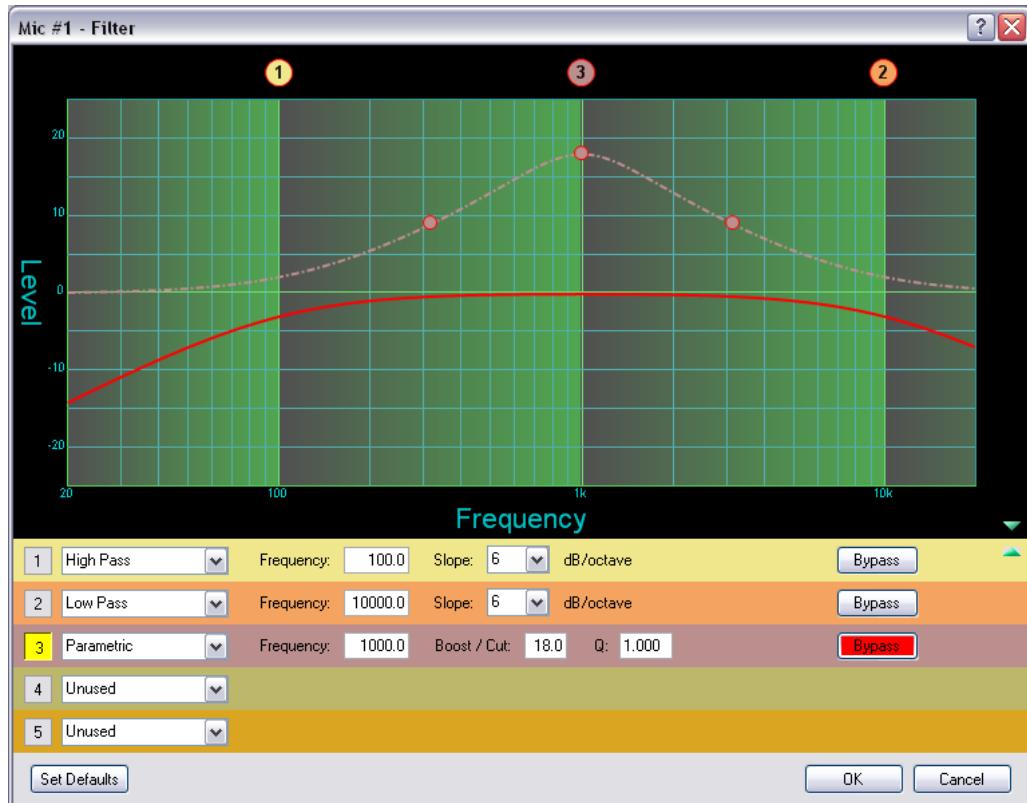


Figure 24. Filter Dialog Box, Filters Added

Within the dialog box, a filter is focused when a filter type is inserted, or is focused by clicking the filter number to the left of the filter selection drop-down list. Note how box 3 (see figure 24) is highlighted in yellow, indicating it is the filter in focus. When bypassed, the results of the filter in focus (independent of other filters) will show in the graph as a dotted line the same color as its filter row. When active (not bypassed), the line is solid.

When multiple filters are enabled, the graph indicates the focused filter result (independent of other filters) in the color of the filter row in the type/parameters table below the graph. The composite response of all filters is displayed in red.

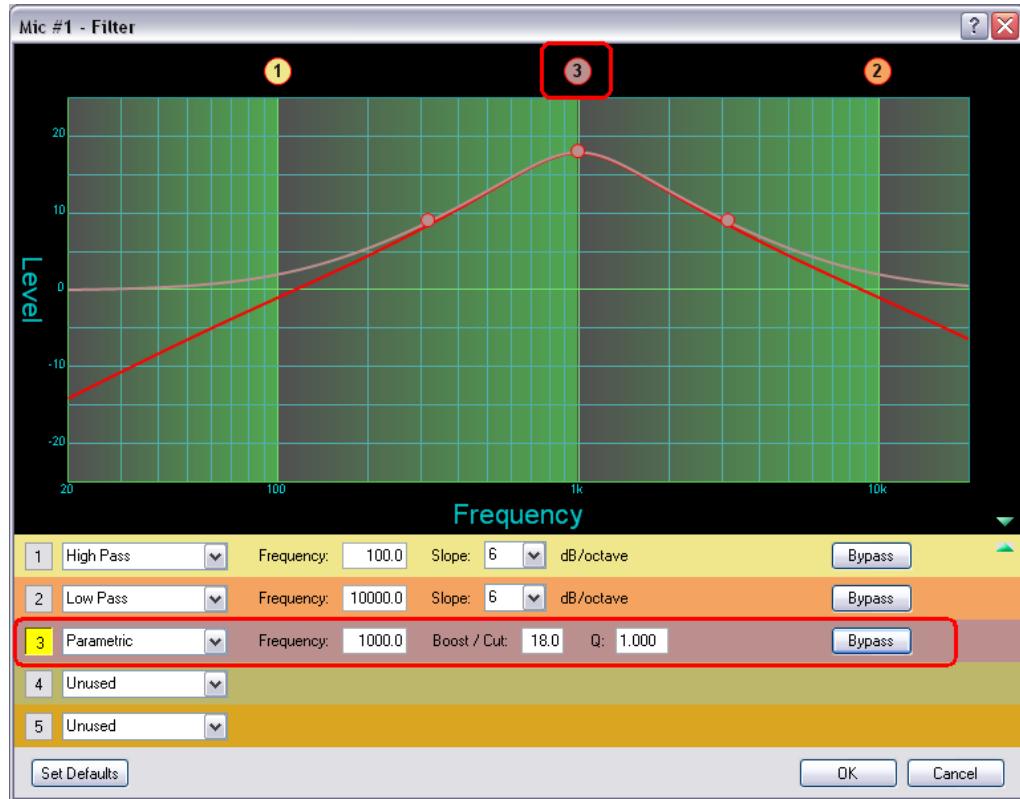


Figure 25. Filter Dialog Box, Filter Not Bypassed

Above the graph, each filter has a "handle" (circled in red above for the parametric EQ filter) placed directly above the cutoff or center frequency. The handle number corresponds to the filter number (also outlined in red). Clicking a handle or clicking the table row brings focus to that filter. <Click+hold+dragging> the handle horizontally changes the cutoff or center frequency to a new position on the x axis.

The table below shows each filter type with default parameter settings. The table immediately following shows the range for each parameter.

Type	Frequency	Parameter 1	Parameter 2
Parametric	1000.0 Hz	Boost/Cut: 0.0 dB	Q: 1.0
High Pass	100.0 Hz	Slope: 6 dB	N/A
Low Pass	10000.0 Hz	Slope: 6 dB	N/A
Bass (Tone)	100.0 Hz	Boost/Cut: 0.0 dB	Slope: 6 dB
Treble (Tone)	8000.0 Hz	Boost/Cut: 0.0 dB	Slope: 6 dB

Filter Parameter	Settings Range
Frequency	20 Hz to 20 kHz
Tone (Boost/Cut)	-24 dB to +24 dB
Q (Parametric EQ only)	0.707 to 15.000
Slope (HP & LP filters only)	1st Order (6 dB) and 2nd Order (12 dB)

High Pass

The high pass filter allows frequencies below the specified frequency to pass unattenuated. All frequencies below the cutoff are attenuated.

The default cutoff is 100 Hz.

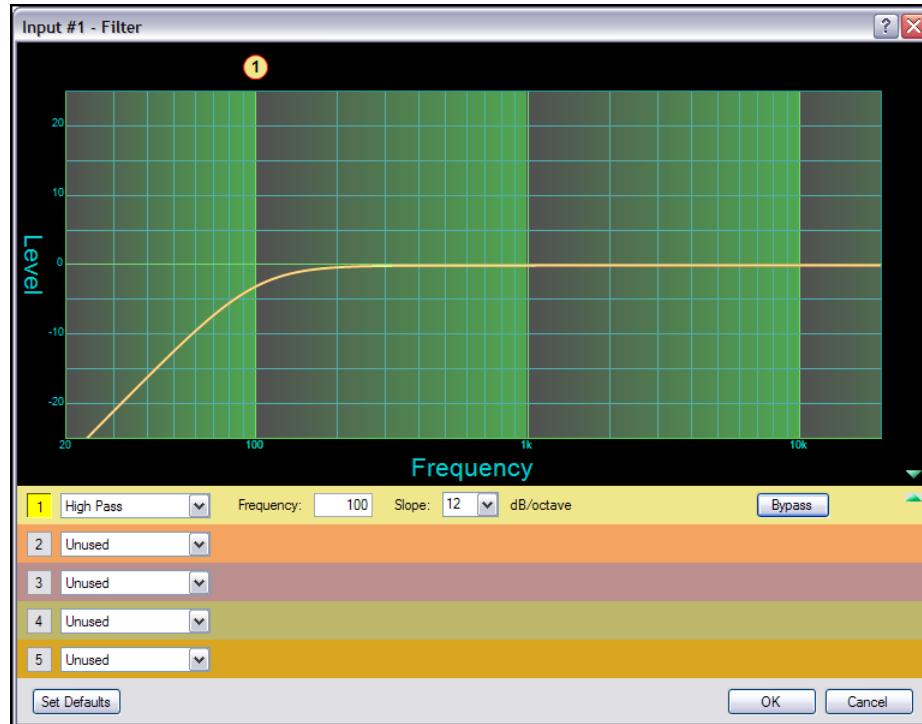


Figure 26. High Pass Filter Response Curve

All frequencies lower than the specified frequency (in this example, 100 Hz) are attenuated leaving the upper frequency response flat. Also note that at the specified frequency (100 Hz) the signal is 3 dB down, typical operation for high pass filters. The 3 dB down point will remain constant regardless of the slope setting. Only the steepness of the frequency attenuation curve changes.

Low Pass

The low pass filter is the opposite of the High Pass filter. All frequencies above the specified frequency are attenuated allowing lower frequencies to pass.

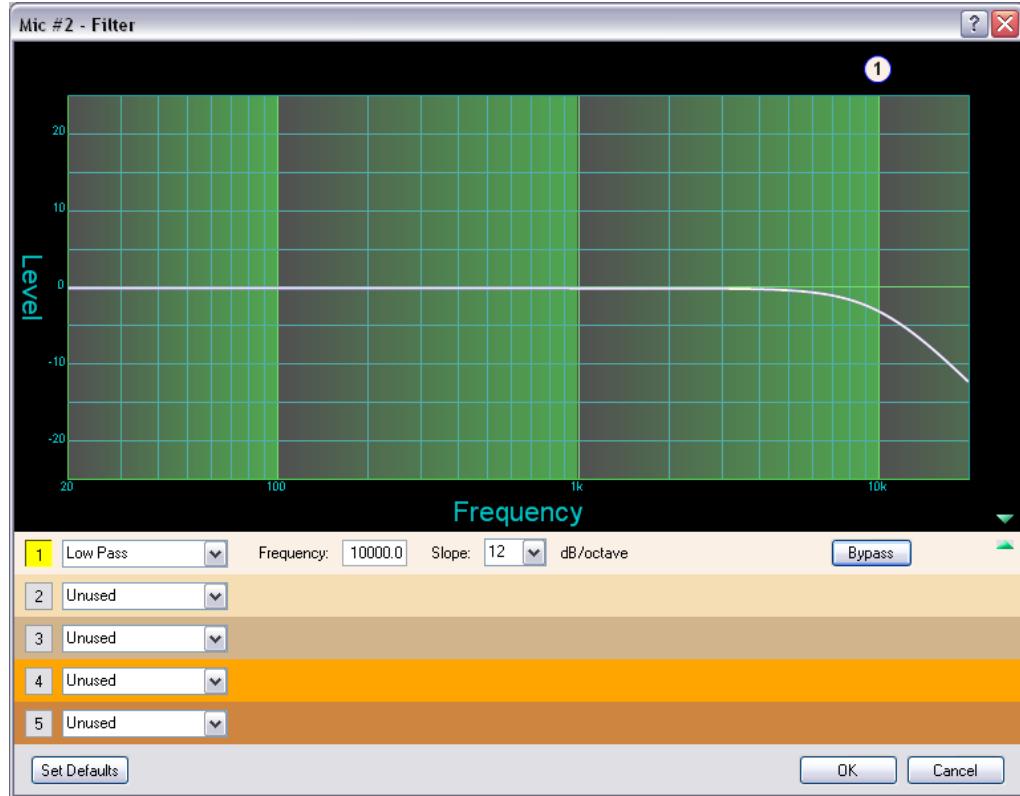


Figure 27. Low Pass Filter Response Curve

Here, the frequencies higher than the specified frequency, 10 kHz, are attenuated leaving the lower frequency response flat.

Bass and Treble Shelving

Bass and treble shelving can be added to the filter. Also known as shelving or tone controls, the separate bass and treble filters provide the ability to cut or boost gain linearly above or below a selected frequency, with the end-band shape giving the visual appearance of a shelf.

Adding the filter automatically inserts both a bass and a treble control row. If only a bass or only a treble filter is required, either bypass the unneeded control or set it to **Unused** in the selection box (see row 3, row 4, and row 5 in figure 28).

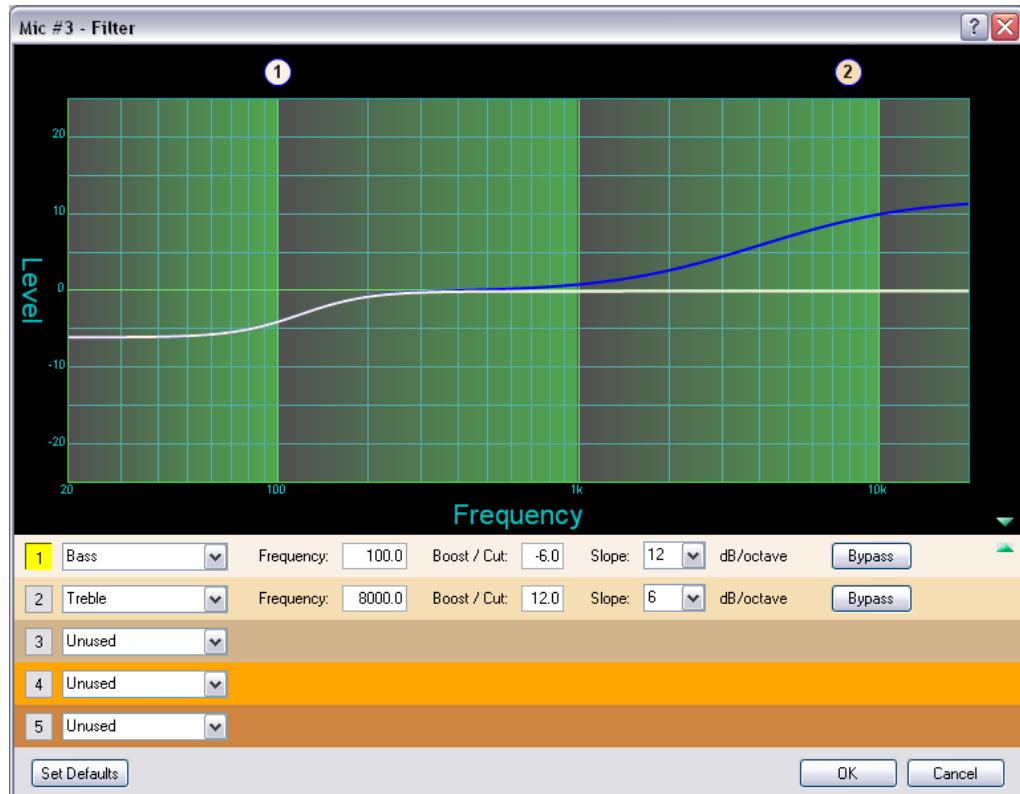


Figure 28. Bass and Treble Shelving

The corner frequency of the controls can be selected to 0.1 Hz accuracy. Two slopes, 6 and 12 dB/octave are selectable along with the ability to boost or cut the signal up to 24 dB.

Parametric (Equalizer)

The parametric filter is a frequency equalizer that offers control of all parameters, including amplitude (the amount of gain [boost], or gain reduction [cut] applied), center frequency (frequency), and range of affected frequencies (Q) around the center frequency.

Up to five parametric filters can be placed in the filter box at one time. Each can be set to a different frequency creating a five band parametric equalizer. The control boosts or cuts the center frequency. By changing the Q value, the range of affected frequencies is widened or narrowed around the center frequency. In general, a higher Q value results in a narrower affected bandwidth.

To demonstrate how Q affects the filter, see the following filter block (see figure 29) containing five parametric filters centered at different frequencies but with the same Q of 1.0. The filter in focus (③) has a center frequency of 1000 Hz boosting that frequency +12 dB over a Q of 1.0. Note the markers on either side of the peak frequency are at 300 Hz on the left and 3000 Hz on the right, a bandwidth of about 2700 Hz.



Figure 29. Parametric Filter Dialog Box, 1000 Hz

The dialog box (see figure 29) shows the frequency curve for the single active filter. To add its effect to the overall frequency response, remove the bypass on the other filters.

By increasing the Q to 10.000, the center frequency remains the same. The markers show the bandwidth of the filter narrowed to between 900 Hz and 1200 Hz, or about 300 Hz (see figure 30). Parametric filters can be used to notch out a very narrow, or very wide range of frequencies using the Q setting.

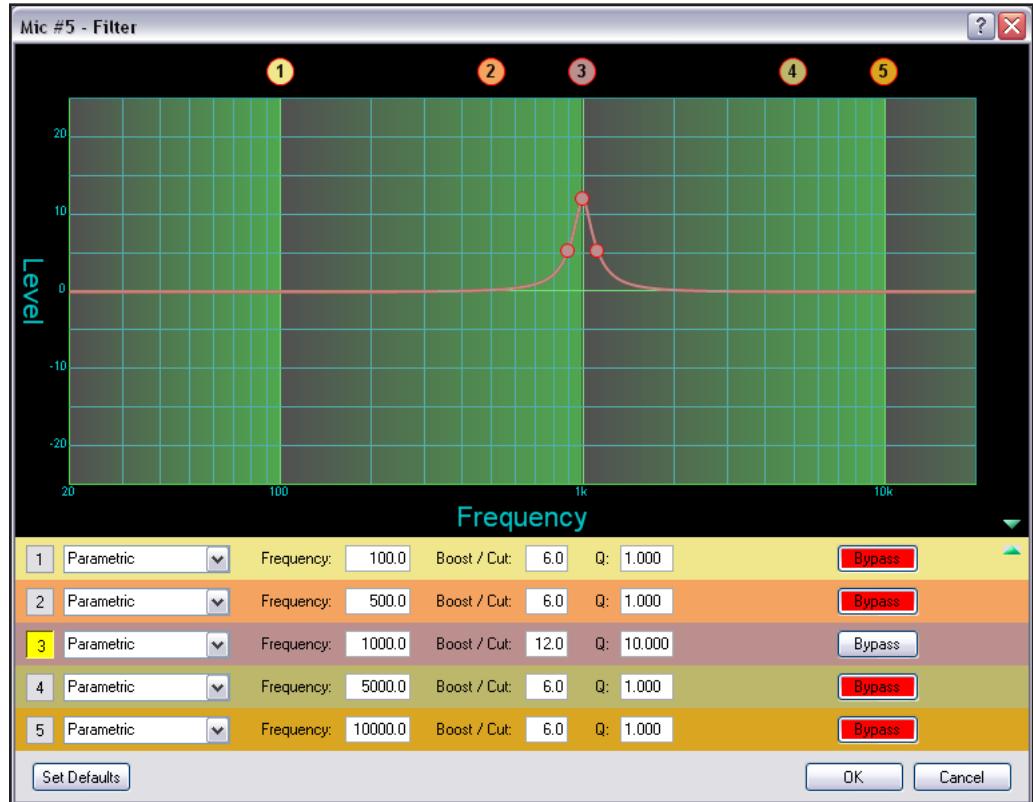


Figure 30. Parametric Filter at 1000 Hz, Q: 10.000

The dialog window above shows the frequency curve for a single active filter. To add its effect to the overall frequency response, remove the bypass on the other filters.

The overall frequency response is now shown as a solid red line with the filter in focus (located in row 3 in figure 31) shown in the color of its table row.



Figure 31. All Parametric Filters Active

The parametric filter allows frequency selection accurate to 0.1 Hz and either 6 or 12 dB of slope. The 3 dB down point remains constant regardless of the slope setting. Only the steepness of the frequency attenuation curve changes.

Feedback Suppressor (FBS)

The Feedback Suppressor is used in live situations when there is an indication of feedback during live operation. Dynamic filters automatically detect feedback on a live microphone channel, and engage a set of up to 5 fixed and 15 dynamic filters to counteract frequency peaks at the detected feedback frequency. Up to 15 separate filters can be employed at any time. The 15 filters act in a FIFO (first in, first out) rotation. If all 15 filters are employed when an additional feedback frequency is detected, it overwrites the first detected feedback frequency and so on.

To avoid a new feedback frequency overwriting a previously detected one, up to five of the dynamic feedback frequencies can be placed into fixed filters. Once written into the fixed filters, the feedback frequency can only be overwritten by the user manually writing a new frequency to the filter.

The FBS dialog box has three tabs; **Settings**, **Dynamic Filters** and **Fixed Filters**. Global settings and view options are controlled from the **Settings** tab. Dynamic to fixed filter allocations are handled from the **Dynamic** filters tab. Filter parameters can be modified from the **Fixed Filters** tab.

The FBS Dialog box provides the following global buttons:

- **Clear All** — Clears all dynamic filter settings.
- **Lock** — Locks the dynamic filters to the current settings, preventing automatic updates. This temporary mode is useful while testing the system, or during the time when dynamic filters are being converted to fixed filters. When the FBS display window is closed, lock mode is automatically disengaged.
- **Bypass FBS** — Turns off feedback detection when engaged (button is red). Only the dynamic filters are bypassed. Fixed filters remain active.
- **Set Defaults** — Click once to return the FBS to default settings.

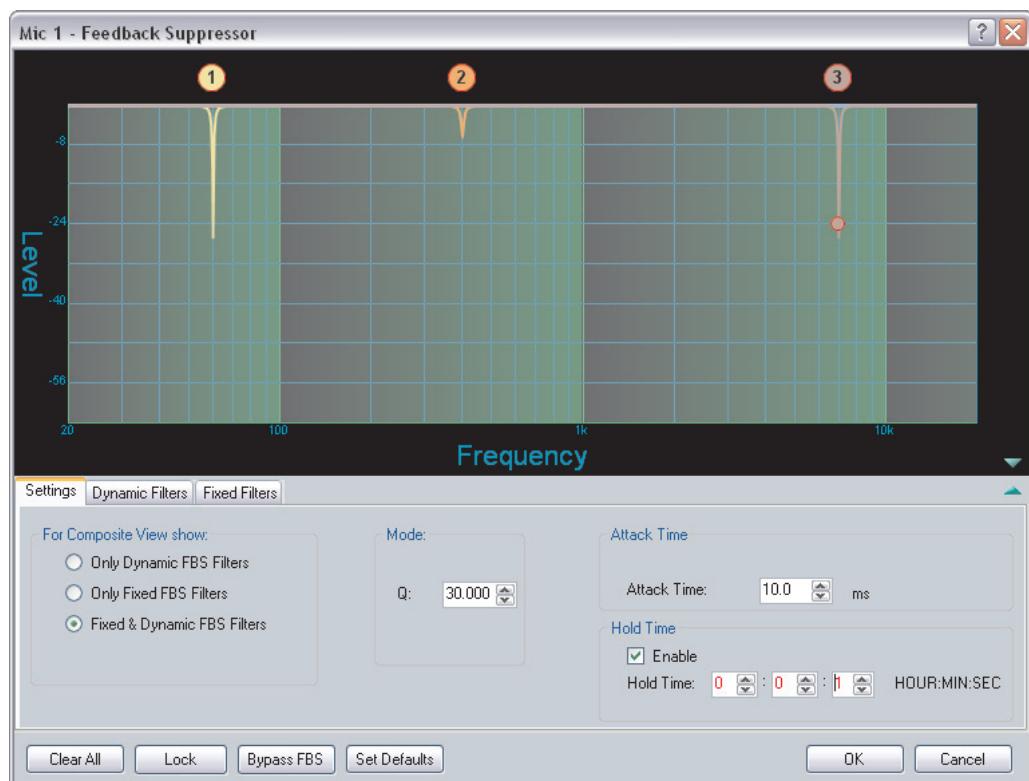


Figure 32. Feedback Suppressor

FBS Settings

The **Settings** tab enables selection of the feedback suppressor parameters.

- **For Composite View show:** — The graph view is set by one of three buttons:
 - **Only Dynamic FBS Filters**
 - **Only Fixed FBS Filters**
 - **Dynamic & Fixed FBS Filters** (default)
- **Mode: Q** — Adjusts the notch filter Q used by dynamic filters. Similar to the parametric filter Q, it changes the bandwidth of the filter. The default setting can be modified in **Tools>Options**. The range is from 5 to 65. Larger values provide less change to the audio frequency response while lower values can provide greater feedback suppression but with more impact to the tonal response of the source audio.

Suggested values for specific applications are:

Q Value	Application
7	Voice with considerable feedback potential
30	Voice with less feedback potential
65	Music with minimal feedback potential

- **Attack Time** — Sets the time at which dynamic filters are generated after feedback detection. A longer attack time (greater than 200 ms) reduces the chance that music or audio content will trigger the dynamic filters to respond. A shorter attack time (less than 2 ms) reduces the time between when feedback is detected and suppressed.
- **Hold Time** — Expressed in **hours:minutes:seconds** up to 9 hours. Hold time sets the time a dynamic filter setting persists before the filter is cleared. When hold time is disabled, dynamic filters persist indefinitely unless cleared manually or the device is power cycled.

Hold time reverts to 00:00:00 when disabled (**Enable** unchecked).

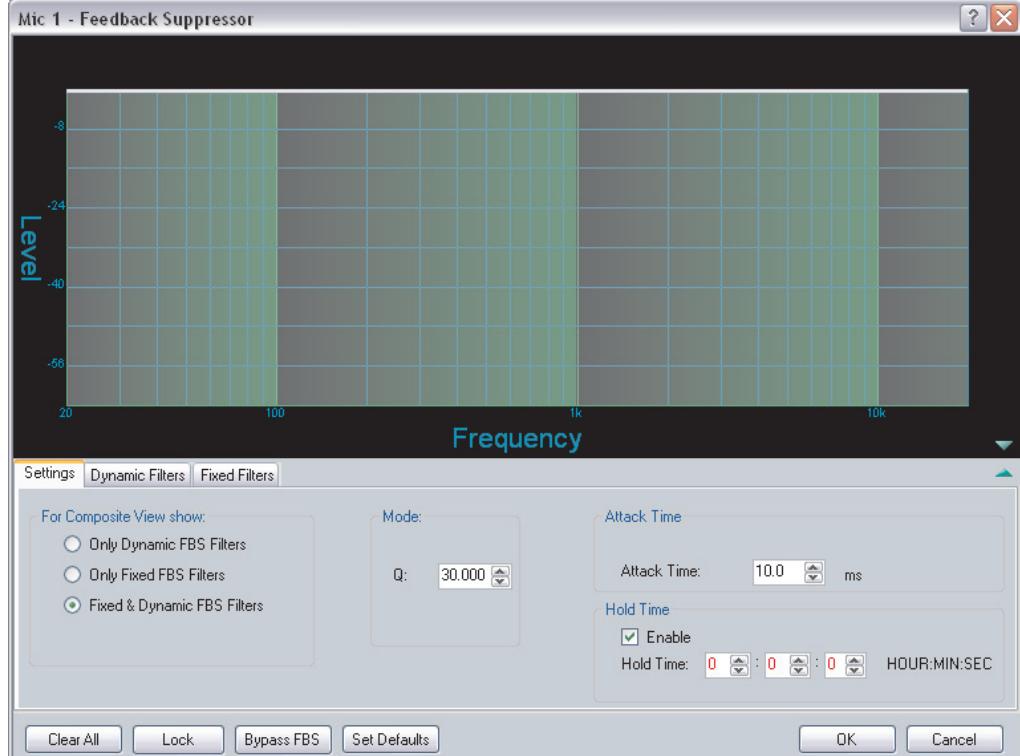


Figure 33. FBS Settings Tab

FBS Dynamic Filters

This tab contains the fifteen dynamic filters, with a scroll bar to display filters hidden due to dialog box size.

Dynamic filters are notch filters that are cut only, providing attenuation up to 30 dB at the specified Q. The default Q is set in the **Tools>Options** menu, but can be changed on the settings tab prior to engaging the FBS dynamic filters. Changing the Q setting after dynamic filters have been generated clears all dynamic filters.

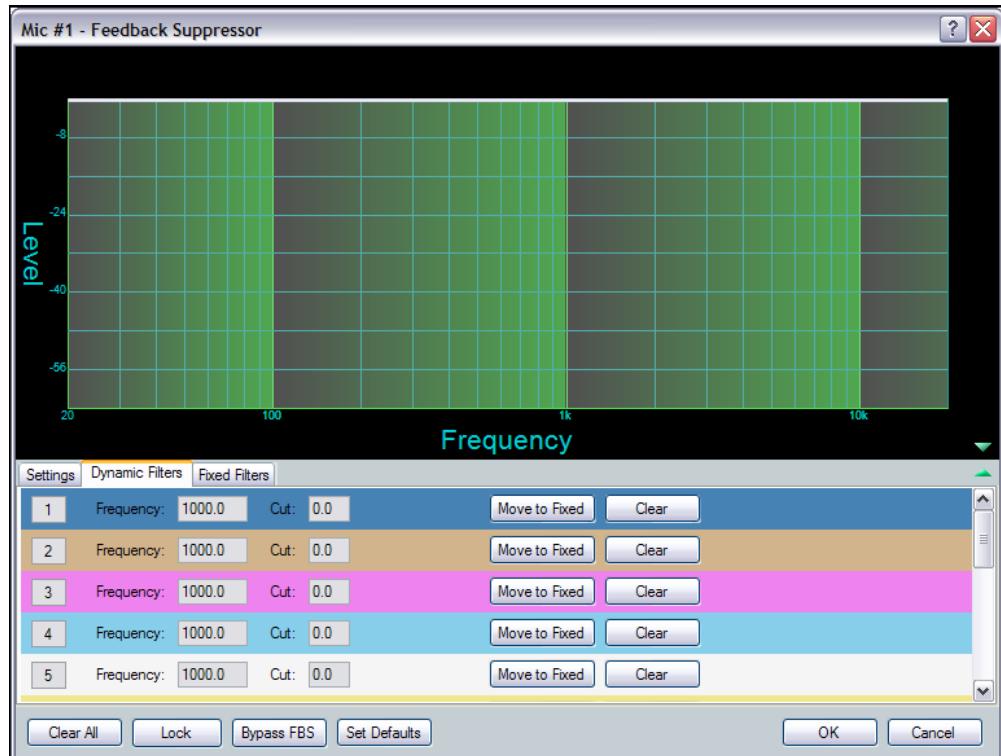


Figure 34. FBS Dynamic Filters Tab

Frequency and cut values are read only. Dynamic filters are in auto-detect mode when the FBS block is active (when **Bypass FBS** is off). If testing reaches a point where no further changes are desired, the lock button can be engaged. The lock mode of operation is temporary, and is intended to be used during setup of the FBS. When the FBS dialog box is closed, lock mode is automatically disengaged.

If there are specific dynamic filters the user wants to assure are not overwritten, press the **Move to Fixed** button to write the designated filter settings to the first available filter in the Fixed Filter tab.

NOTE: When a dynamic filter setting is moved to the fixed filter, it automatically clears that frequency from the dynamic filter.

The **Clear** button removes a detected frequency from the corresponding dynamic filter. A cleared filter reverts to auto-detect mode unless **Lock** mode is engaged.

FBS Fixed Filters

Fixed filters are notch filters with an adjustable center frequency and Q, and up to 30 dB of cut. The fixed filters are typically set by converting dynamic filters to fixed, however adjustments to filter parameters can be made manually from the **Fixed Filters** tab.

Fixed Filters are inactive and the filter type is set to **Unused** by default.

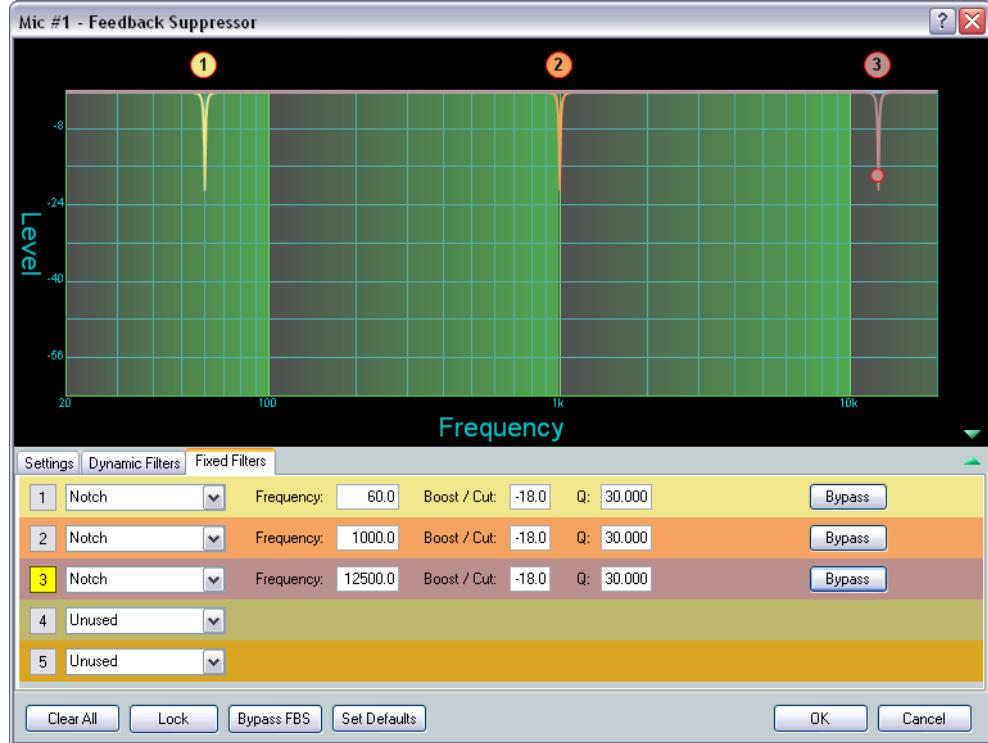


Figure 35. FBS Fixed Filters Tab

No filter parameters are displayed when the filter type is set to **Unused**. As a filter is moved to the fixed filter tab from a dynamic filter, the filter becomes active and displays **Notch** as the filter type. The parameters copied from the dynamic filter are displayed in the same line. Once a fixed filter is active, settings can be modified or adjusted if needed. Fixed filters can also be individually bypassed by clicking the **Bypass** button.

FBS Settings Ranges and Fixed Filter Defaults

FBS Parameter	Settings Range	Default Setting
Frequency	20 Hz to 20 kHz	N/A
Q	5.000 to 65.000	30.000
Attack Time	0.0 ms to 1000.0 ms	10.0 ms
Filter Hold Time	0 seconds to 9 hours	00:00:00; Disabled

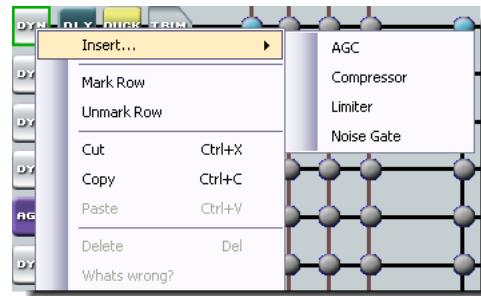
Fixed Filter Parameter	Settings Range	Default Setting
Frequency	20 Hz to 20 kHz	1000.0 Hz
Q	1.000 to 65.000	30.000
Cut	Up to 30 dB cut	0.0 dB

Dynamics (DYN)

A dynamics processor alters the dynamic range, the difference between the loudest to the quietest portions, of an audio signal. Each input channel provides two dynamics processor blocks that, when inserted, provide one of four types; AGC, Compressor, Limiter, or a Noise Gate processor.

To insert a processor into an empty block, select **Insert** from the processor menu.

The menu appears when the block is double-clicked, or is accessed from a context menu that appears when the block is right-clicked.



Once the processor has been inserted, individual processor parameters can be changed in the dialog box, accessed by double-clicking the processor block. For comparison, the block can be bypassed by clicking a **Bypass** button.

All parameters are displayed in a text box with a resolution of 0.1 (dB or ms). Parameters are set by direct entry in the text box to replace existing text, then pressing <Enter>, <Tab>, or clicking to another area. Threshold, gain/attenuation, target, and ratio parameters have adjustment points on the graph display. Use the mouse to **click + drag** the graph point to the desired destination or value. All time values have a horizontal slider allowing adjustment in 1 ms increments by either a **click + drag** of the slider handle, or focusing on the slider, then using <Left> or <Right> arrow keys (<Page Up> and <Page Down> keys adjust in increments of 10 ms).

The table below lists factory default settings for each dynamics processor type and parameter.

Parameter	AGC	Compressor	Limiter	Gate
Threshold	-40.0 dB	-30.0 dB	-10.0 dB	-65.0 dB
Max Gain	12.0 dB			
Target	-10.0 dB			
Window	12.0 dB			
Attack Time	500.0 ms	5.0 ms	2.0 ms	1.0 ms
Release Time	1500.0 ms	100.0 ms	50.0 ms	1000.0 ms
Ratio		2.0 :1		20.0 :1
Hold Time	0.0 ms	100.0 ms	50.0 ms	300.0 ms
Max. Attenuation				25.0 dB
Soft Knee		Off	Off	

Details of the individual dynamics blocks follow.

Automatic Gain Control (AGC)

AGC adjusts the gain level of a signal based upon the input strength to achieve a more consistent volume. Below the set threshold, the signal is not affected. Above the threshold, weaker signals are boosted up to the maximum gain setting to reach a user-defined target level. As the signal approaches the target level it receives less gain or no gain at all. Once the signal reaches the target level all gain is removed.

Click in each field to change the values.

Threshold — The input level where maximum gain will be applied (after the attack time is exceeded). On the graph at right, follow the red input level from the lower left to -40 dB where the first red circle is. Signal levels less than -40 dB remain at their original levels. All signal levels at or exceeding -40 dB will have up to 12 dB of gain applied (see Maximum Gain, below).

The threshold level can be adjusted from -80.0 to 0.0 dB in 0.1 dB increments.

Default is -40.0 dB.

Maximum Gain — The highest amplification applied to a signal exceeding the threshold and up to the lower limit of the window (see **Window**, below).

Maximum Gain can be set from 0.0 dB to +60 dB in 0.1 dB increments.

Default is 12.0 dB.

Target — The desired average signal level of the output when AGC is applied. AGC can vary the gain according to the input signal level, specified target level and maximum gain. As the signal approaches the target level of -10 dB, gain is reduced until at -10 dB, gain is no longer applied.

The target level can be adjusted from -40 dB to 0.0 dB in 0.1 dB increments.

Default is -10.0 dB.

Window — Indicated by the two yellow lines, is a specified range above and below the target level. Below the lower line maximum gain is always applied to the signal. When the signal reaches the window, gain control begins scaling in a linear fashion to achieve smoother results as the signal reaches the target level.

The window range can be set in 0.1 dB increments from 0.0 dB to 20.0 dB.

Default is 12.0 dB.

Attack Time — Adjusts the time delay for AGC to engage after the input signal level reaches or exceeds the threshold level.

Attack time can be adjusted from 0.0 to 3000.0 ms in 0.1 ms increments.

Default is 500.0 ms.

Hold Time — Adjusts how long AGC continues after the input signal drops below the threshold and before release time begins.

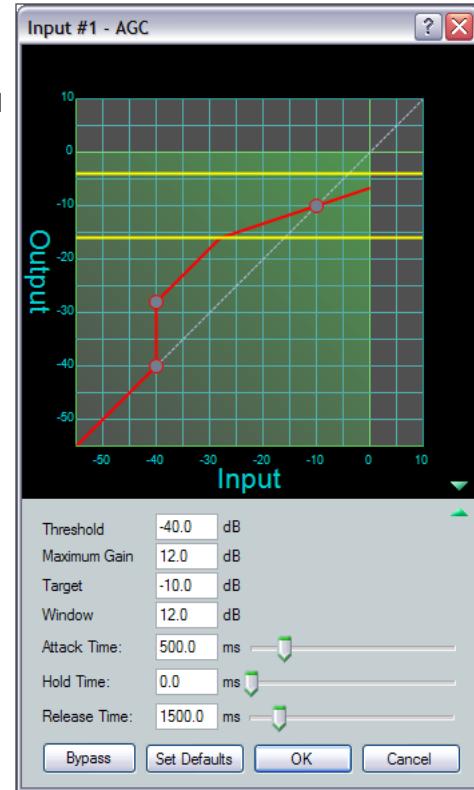
Hold time can be adjusted from 0.0 to 3000.0 ms in 0.1 ms increments.

Default is 0.0 ms.

Release Time — Adjusts the time it takes to return the signal to normal (unprocessed) levels after the signal no longer exceeds the threshold level setting. Release time begins only after hold time is reached.

Release time can be adjusted from 10.0 to 10000.0 ms in 0.1 ms increments.

Default is 1500.0 ms.



Compressor

The compressor regulates signal level by reducing (compressing) the dynamic range of the input signal above a specified threshold. The input level to output level ratio determines the reduction in the dynamic range beyond the threshold setting. For example, with a ratio setting of 2:1, for every 2 dB of input above the threshold, the compressor outputs 1 dB.

Compression is commonly used to contain mic levels within an acceptable range for maximum vocal clarity. A compressor can also make softer sounds louder in one of two ways. The dynamic range can be reduced by compressing the signal above the threshold while raising the post-compressor gain/trim (referred to as "make-up gain"). Alternately, the input signal can be increased while the compression ratio above the threshold is increased correspondingly to prevent clipping. Both techniques have the effect of making louder portions of a signal softer while at the same time increasing softer signals to raise them further above the noise floor.

Compression can also be used to protect a system or a signal chain from overload similar to a limiter. Click in each field to change the value.

Threshold — The input signal level above which compression begins (subject to attack time) and below which compression stops (subject to hold and release time).

The threshold level can be adjusted from -80.0 to 0.0 dB in 0.1 dB increments.
Default is -30.0 dB.

Ratio — The input signal level reduction when compression is engaged.

Ratio can be adjusted from 1.0 to 100.0 in 0.1 increments.
Default is 2.0:1.

Attack Time — Adjusts the time delay for compression to engage after the input signal level reaches or exceeds the threshold level.

Attack time can be adjusted from 0.0 to 200.0 ms in 0.1 ms increments.
Default is 5.0 ms.

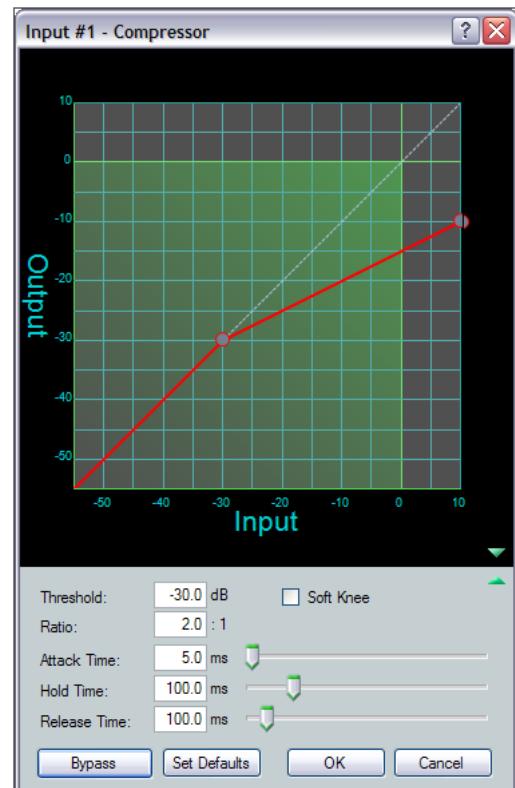
Hold Time — Adjusts how long compression continues after the input signal drops below the threshold and before release time begins.

Hold time can be adjusted from 0.0 to 500.0 ms in 0.1 ms increments.
Default is 100.0 ms.

Release Time — Adjusts the time it takes to return the signal to normal (unprocessed) levels after the signal no longer exceeds the threshold level setting. Release time begins only after hold time is reached.

Release time can be adjusted from 10 to 1000.0 ms in 0.1 ms increments.
Default is 100.0 ms.

Soft Knee — Select the **Soft Knee** checkbox to smooth and soften the transition from uncompressed to compressed output levels. There are no adjustments.



Limiter

The limiter restricts the input signal level by compressing its dynamic range above a specified threshold. The limiter is most commonly used to prevent clipping, protecting a system against component or speaker damage. While the limiter is closely related to the compressor, it applies a much higher compression ratio of $\infty:1$ above the threshold. The ratio is fixed and cannot be changed. Click in each field to change the value.

Threshold — The input signal level above which limiting begins (subject to attack time) and below which compression stops (subject to hold and release time).

Threshold level can be adjusted from -80.0 to 0.0 dB in 0.1 dB increments.

Default is -10.0 dB.

Attack Time — Adjusts the time delay for limiting to engage after the input signal level reaches or exceeds the threshold level.

Attack time can be adjusted from 0.0 to 200.0 ms in 0.1 ms increments.

Default is 2.0 ms.

Hold Time — Adjusts how long limiting continues after the input signal drops below the threshold and before release time begins.

Hold time can be adjusted from 0.0 to 500.0 ms in 0.1 ms increments.

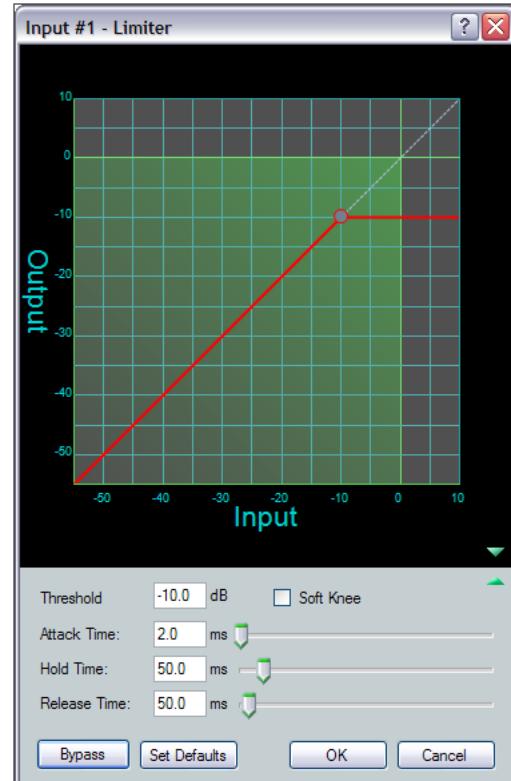
Default is 50.0 ms.

Release Time — Adjusts the time it takes to return the signal to normal (unprocessed) levels after the signal no longer exceeds the threshold level setting. Release time begins only after hold time is reached.

Release time can be adjusted from 10 to 1000.0 ms in 0.1 ms increments.

Default is 50.0 ms.

Soft Knee — Select the **Soft Knee** checkbox to smooth and soften the transition from uncompressed to compressed output levels. There are no adjustments.



Noise Gate

The noise gate allows an input signal to pass only when it exceeds a specified threshold level. Above the threshold level, the signal passes unprocessed. Below the threshold the signal is attenuated at the rate set by the ratio adjustment. The typical setting of the noise gate threshold is just above any noise level in the environment or source equipment. That allows signals that are above the noise to pass, and attenuates the noise when there is no signal, eliminating background noise.

Threshold — The input signal level below which attenuation (gating) begins (subject to attack time) and above which gating stops (subject to hold and release time).

The threshold level can be adjusted from -80.0 to 0.0 dB in 0.1 dB increments. Default is -65.0 dB.

Max Attenuation — The maximum attenuation of the signal when it drops below the threshold.

Maximum attenuation can be adjusted from 0.0 to 80.0 dB in 0.1 dB increments. Default is 25.0 dB.

Ratio — The input signal level reduction when gating is engaged.

The ratio can be adjusted from 1.0 to 100.0 in 0.1 increments. Default is 20.0:1.

Attack Time — Adjusts the time delay for gating to engage after the input signal level drops below the threshold level.

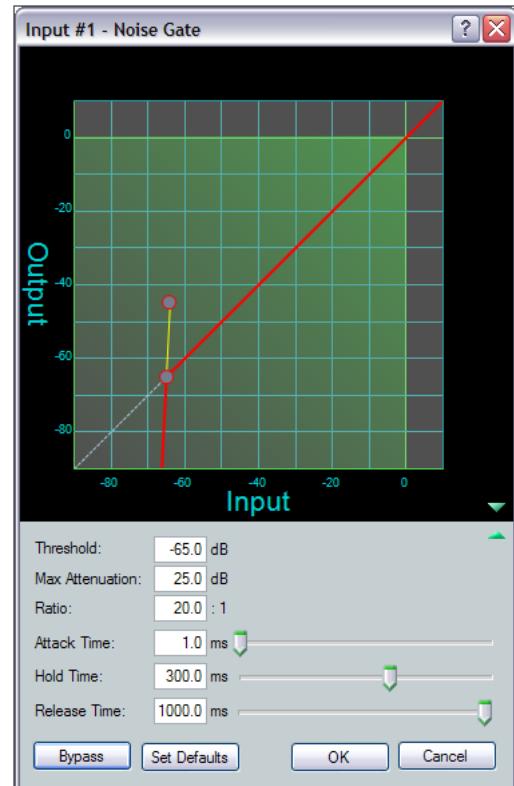
Attack time can be adjusted from 0.0 to 200.0 ms in 0.1 ms increments. Default is 1.0 ms.

Hold Time — Adjusts how long gating continues once the input signal rises above the threshold. If the signal is still above the threshold when hold time ends, release time begins.

Hold time can be adjusted from 0.0 to 500.0 ms in 0.1 ms increments. Default is 300.0 ms.

Release Time — Adjusts the time it takes to return the signal to normal (unprocessed) levels once the signal has increased above the threshold level setting. Release time begins only after hold time is reached.

Release time can be adjusted from 10.0 to 1000.0 ms in 0.1 ms increments. Default is 1000.0 ms.



Delay (DLY)

The delay processor, when inserted, provides a means to delay the audio signal. Audio delay syncs audio to video or can time-align speakers placed at different distances from the listener. The DMP 64 can set delay by either of two criteria: time or distance (feet or meters).

The default units setting is time with a range of 0.0 ms to 200.0 ms adjustable in 0.1 ms steps. Default is 100.0 ms.

Settings are controlled with a vertical slider and indicated with a value readout field. The value can be changed by clicking within the readout field, changing the number, then either pressing <Enter>, <Tab>, or clicking away from the field.

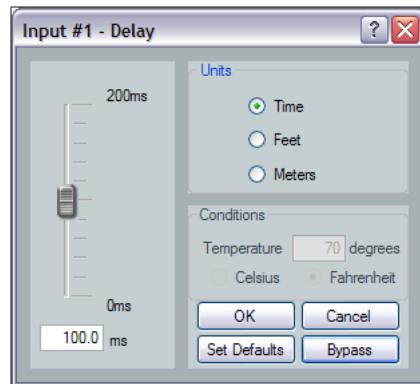


Figure 36. Delay Dialog

Slider adjustments made in feet or meters correspond incrementally to the distance required to make 1 ms or 5 ms adjustments (detailed in the table below). If more precision is required, enter time in 0.1 ms increments directly into the readout field.

Method	Time	Feet	Meters
Click + drag	1 ms	~1.1 feet	~0.3 m
Focus + arrow	1 ms	~1.1 feet	~0.3 m
Focus + Page Up/Down	5 ms	~5.6 feet	~1.7 m

When distance (feet or meters) is chosen, the conditions (temperature) field becomes available and can be set either by degrees Fahrenheit or Celsius. When entering a distance, time delay compensation is automatically modified based on differences in the speed of sound due to air temperature.

Default is 70 degrees Fahrenheit.

NOTE: When using distance (feet or meters), set a temperature value first, then set the distance.

Ducking (DUCK)

Ducking provides a means to duck, or lower, the level of one or more input signals when a specified source must take precedence. The ducking processor block, when inserted, provides a means to duck one or more mics and program material (ducking targets) when the processor detects a signal from the ducking source. Ducking lasts for the duration of the interrupting signal (ducking source) determined by the threshold setting (plus hold and release time) and restores the original levels of the ducked inputs once the other signal has ceased.



Ducking can be useful when:

- Program material needs to be attenuated in order to more clearly hear a narrator voice.
- One microphone, such as one used by a master of ceremonies, needs to have priority over other mics, program material, or both.
- A paging mic needs to attenuate all other signals.

All ducking processor blocks are controlled by a common dialog box that opens when any of the ducking blocks are selected (see figure 37). All empty ducking processor blocks have no ducking source or target settings by default.

When the first ducking processor is inserted, that input is automatically set to **Enable Mic/Line Source**. All inactive ducking processor blocks have **Enable Mic/Line Source** unchecked by default.

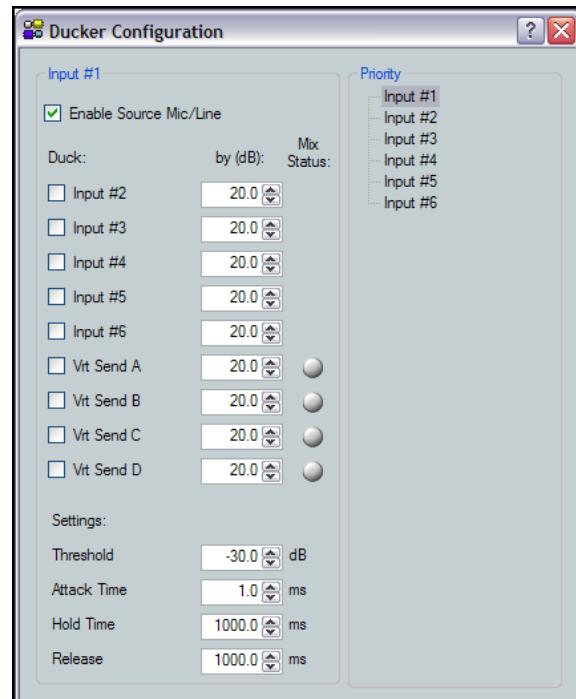


Figure 37. Ducker Configuration Dialog

Any of the six inputs can be ducking sources. Any or all of the remaining inputs and virtual sends can be targets.

Ducking Configuration

Ducking is configured in a dialog box that opens when an active ducking processor block is double-clicked (see figure 37 on the previous page).

① Current source indicator

Shows the input selected as the ducking source. Ducker settings affect the input channel shown here. When a ducker dialog is opened for a channel, the current source defaults to that channel. The current source can also be selected by the priority readout/source selector (see below).

② Enable mic/line source checkbox

When checked, ducking is enabled for the current source and the ducker processor block is lit. When unchecked, ducking is disabled for the current source. The ducker processor block is unlit.

③ Duck Targets:

Shows all potential input targets. Only inputs that are checked are ducked. The current source is not available as a target (a source cannot duck itself). If the current source is designated as a target of another input channel, that input channel is not available (a target cannot be the source).

④ Settings:

Used to configure the parameter settings for the ducker source. When a ducker block is copied, these settings are transferred.

Threshold — Sets the input signal level, in dB, the ducking source must exceed before ducking begins. If ducking does not occur quickly enough to avoid loss of speech or program material from the ducking source, decrease this setting. If ducking occurs too soon, allowing background noise to trigger ducking, increase the setting.

The range is -60 to 0 dB in 1 dB increments.

Default is -30 dB.

Attack Time — Adjusts the time to duck the targets once the threshold is exceeded. The range is 0 to 3000 milliseconds in 1 millisecond increments. Default is 1 millisecond.

Hold Time — Determines the time, in milliseconds, after a ducking source signal drops below the threshold before ducking ceases.

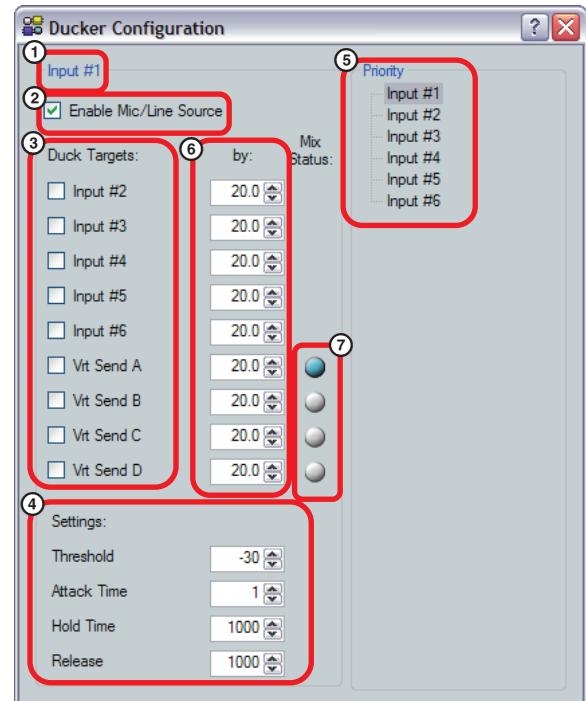
The range is 0 to 10000 milliseconds in 1 millisecond increments.

Default is 1000 milliseconds (1 second).

Release — Determines how long, in milliseconds, after the ducking source level is below the threshold and the hold time is met, the ducking targets take to restore signal levels.

The range is 10 to 10000 milliseconds in 1 millisecond increments.

Default is 1000 milliseconds (1 second).



⑤ Priority

Displays the hierarchy of ducking source to duck targets (see Ducking Priority below). Priority levels are displayed in tree fashion. Click an input channel to select that channel as the current source. The current source indicator (O) reflects the selected input channel.

⑥ By (dB): (Target gain reduction amount)

Individual attenuation settings for each duck target in dB. If additional attenuation of a target is required, increase this value.
The attenuation range is 80.0 to 0.0 dB in 0.1 dB increments.
Default is 20.0 dB.

⑦ Mix Status (for virtual returns):

Indicates whether the source is being routed to the virtual sends. This is a readout value only, shown for convenience during ducking configuration.

Ducking Priority

Multiple levels of ducking can be required enabling an input source to take precedence over all but one other input. In this example, inputs 2 through 6 are set to duck when Input #1 has a signal above the ducking threshold. Input #2 is set to duck inputs 5 and 6. Since Input #1 has previously been set to duck Input #2, Input #1 is disabled to prevent contradictory priorities.

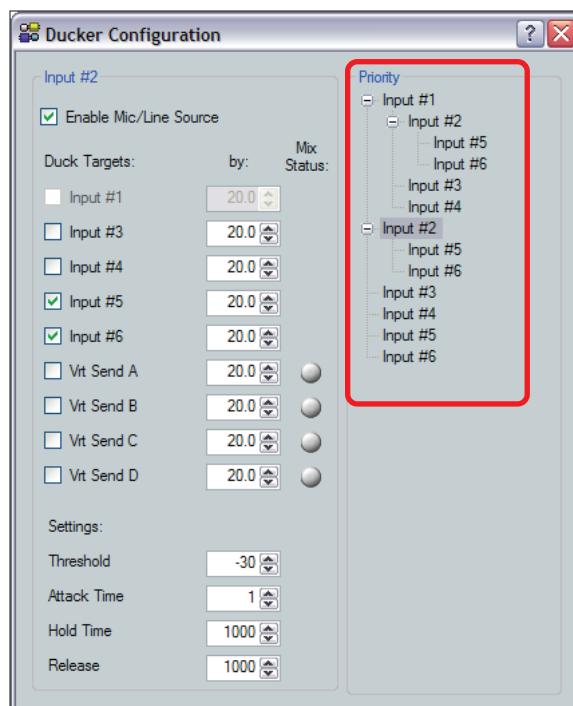


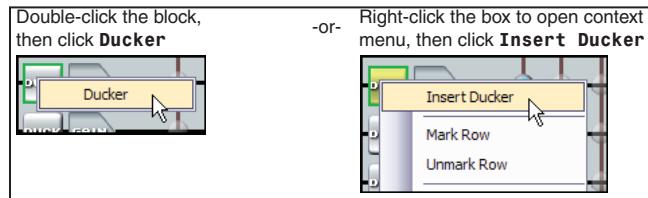
Figure 38. Ducker Configuration, Input Priority

The priority tree outlined on the right side of figure 38 shows the inputs arranged by their priority status. Input #1 has all other ducked inputs under it, therefore if a signal is detected, it will trigger Inputs 2 through 6 to duck. If Input #2 detects a signal and there is no signal on Input #1, Input #2 will trigger inputs 5 and 6 to duck. However, if the Input #1 signal exceeds the threshold, it will then duck all inputs including Input #2.

NOTE: Ducking attenuation is not additive. When an input target is ducked, regardless of how far down the priority list it is, the maximum attenuation is what is set in the “**by (dB):**” column near the center of the dialog box.

Ducker Tutorials

The examples below are based on different input configurations. Insert a ducker from a ducker processor block using one of the following methods:



Once inserted, double-click on the ducker block to open the ducker configuration dialog box. The **Enable Mic/Line Source** box is checked.

Ducking and Priority Ducking

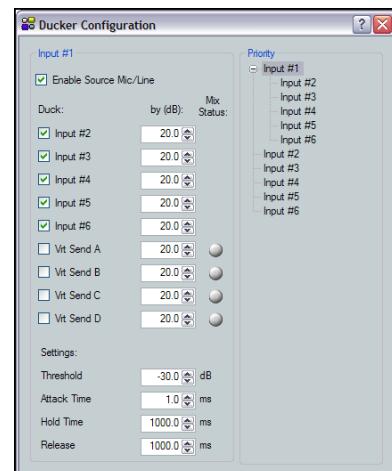
The first inserted channel ducks all selected targets.

To set a ducking source:

1. Insert a ducking processor to input #1.
2. Open the ducker configuration box and select the desired duck targets. In this example inputs #2 to #6 are the ducking targets.

A signal on input #1 that exceeds the ducking threshold now ducks inputs #2 to #6.

The ducking processor also provides a means to have an additional input duck other targets using the priority feature. The second input ducks its selected duck targets, and can also be ducked by the first ducking source.



To set an additional ducking source:

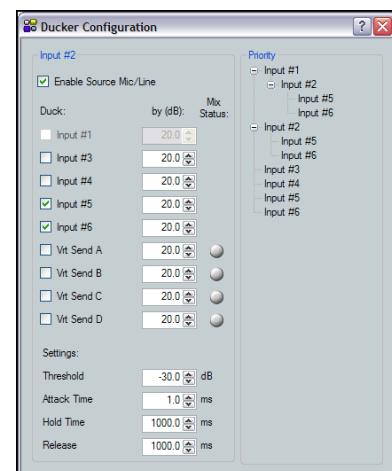
1. Insert a ducking processor on the additional ducking source. In this example, input #2 is the second ducking source, with input #1, as shown above, as the first source.

NOTE: Since it was previously selected as a ducking target, Input #1 is not available as a target of input #2.

2. Open the ducking dialog window for the input and select the desired duck targets. In this example inputs #5 and #6 are the ducking targets of input #2.

Any signal on input #2 that exceeds the ducking threshold now ducks inputs #5 and #6. The ducking targets can be changed at any time by double-clicking the input #2 ducking processor block.

If a signal on input #1 exceeds the ducking threshold, inputs #2 to #6 are still ducked regardless of whether the signal on input #2 exceeds its ducking threshold.



NOTE: No input is ducked more than the level set in the **by (dB)** : box.

Pre-mixer Gain (GAIN)

The post-input processing gain control (also called the pre-mixer gain) provides gain or attenuation post-processing gain block. It includes a mono long-throw fader with a -100.0 to +12.0 dB gain range, and a current level setting readout below the fader. Fader adjustments are in 1 dB increments, while adjustments can be entered manually to 0.1 dB resolution.

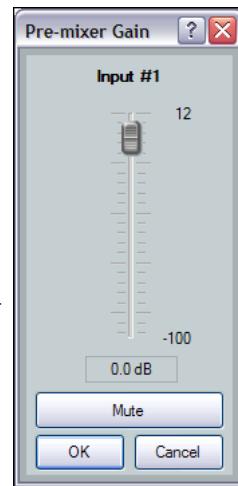
Default is unmuted at unity (0.0 dB) gain.

Selecting the fader handle with the mouse or clicking within the fader area brings focus to the fader. The input signal level can be adjusted using any of the following methods:

- Select and hold the fader handle, then drag it to desired level in 1.0 dB steps.
- Select or tab to the fader handle, then use the up/down arrow to set the desired level in 1 dB steps.

<Page Up> and <Page Down> increases and decreases level in 5 dB steps.

Click in or tab to the level readout field. Type a new value, then press <Enter> or <Tab> to another area. .



Line Output Channels

There are four mono line output channels. Controls and processing blocks, identical for each output channel, are described in the following sections.



Loudness (LOUD)

The loudness processor, when inserted, applies a filter compensation curve to the signal in an inverse relationship to the output volume control setting. The higher the gain setting, the less loudness compensation is applied. Generally, as volume is lowered, perception of certain frequencies is progressively diminished, returning to a more flat response as volume is increased. Loudness boosts those diminished frequencies to the highest degree at low volume levels, decreasing the boost as volume increases.

The bypass button is red when engaged (loudness control defeated), and gray when disengaged (loudness control active).



When bypassed, the graph displays the current filter curve as a dotted line. When bypass is disengaged, the current filter curve is displayed as a solid line.

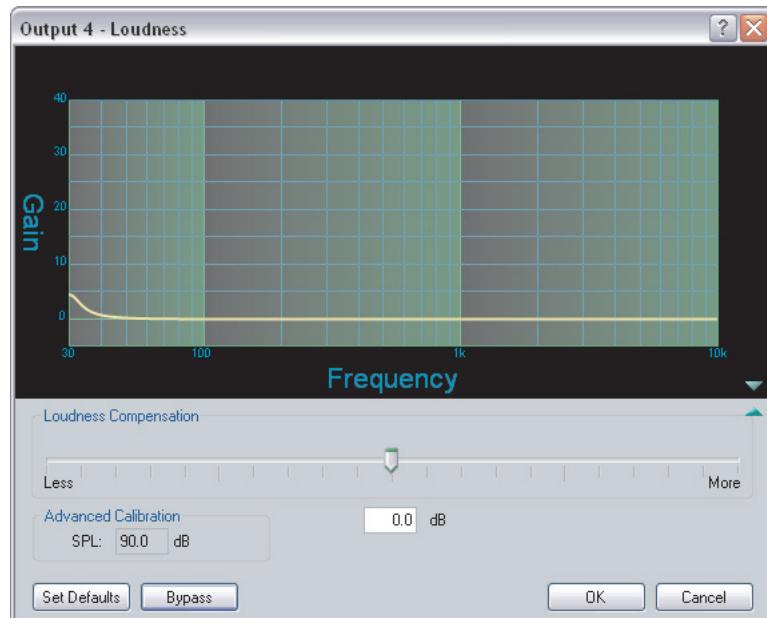


Figure 39. Loudness Dialog Window

The Loudness dialog window contains the following elements:

- **Graph** — Displays the compensation curve applied to the signal. These curves are read-only, and are not adjustable from the graph.
- **Loudness Compensation slider** — From a center zero-point, the user can slide to the left for less loudness compensation (filter curve is reduced), or to the right for more (filter curve is increased). The slider position is translated into a dB value, displayed in the compensation readout box contained in the Advanced Calibration section. The slider has a 48 dB (± 24 dB) range.
- **Advanced Calibration** — The advanced calibration box provides a value that corresponds to the position of the compensation adjustment slider. The SPL box displays the summed value of the slider and the preceding trim control.

Calibrating Loudness

The user can fine-tune the amount of loudness compensation using the compensation adjustment slider and adjusting "by ear," or by measuring SPL levels in a particular room, then using the slider to adjust the loudness filter relative to the SPL of the room and system gain structure.

Before calibrating loudness, set up the system gain structure (see [Optimizing Audio Levels](#) on page 88). A pre-recorded track of pink noise or pink noise from a signal generator is preferable for this purpose. Program material can also be used (using familiar material is recommended).

If using a signal generator, set it to output -10 dBu. Then set the input gain of the DSP Configurator so the input meter reads -20 dBFS. If using a recorded source, the pink noise should be recorded at -20 dBFS and the player output level setting control set to maximum, or 0 dB of attenuation. For program material, set the input level to meter at approximately -15 dBFS, with peaks safely below 0 dBFS.

Unmute the mix-point from the pink noise source to the output connected to the room amplifier being calibrated. With the basic gain structure previously set up, loudness can be calibrated using an SPL meter or by ear. Loudness can also be set using an SPL meter, then fine-tuned by ear.

To calibrate loudness, use a sound pressure level meter set to "C" weighting:

1. Set the Loudness processor to **Bypass** (**Bypass** button red).
2. Place the meter in an average (but somewhat prominent) listening location.
3. Generate pink noise, or start the program material playback.
4. Measure the SPL in the room.
5. In the loudness dialog, adjust the slider until the value in the **SPL** readout box matches the reading on the SPL meter.

NOTE: Theoretically, calibration can be performed with the output channel volume and post-mixer gain level set to any comfortable listening level. But a relatively loud volume (well above the ambient noise in the room) that can be easily measured is preferred.

Loudness is now calibrated. Disengage **Bypass** to hear the compensation.

Alternate method to calibrate loudness:

1. Set up the procedure using step 1 through 3 of the previous procedure.
2. Set the compensation adjustment slider to its default center position
3. Set the output channel volume fader to 0 dB (100% volume),
4. Adjust the amplifier until the SPL meter reads 90 dB.

Loudness is now calibrated. This method works if 90 dB is an acceptable volume limit for the room.

Setting Loudness "By Ear"

When setting loudness by ear, it is essential the system gain structure be set up first. Sit in an average (but somewhat prominent) listening location.

1. Set the loudness processor to **Bypass**.
2. Set the output volume fader in the DSP Configurator to a relatively quiet listening level. Filter compensation from the loudness processor is most prominent at low listening levels. Use familiar program material set to the levels described earlier.

3. Set the **Calibrate** slider to 0, the center point. Disengage the loudness **Bypass**. The result is a moderate enhancement to the program material, with more accentuated bass frequencies (below 500Hz), and more brightness in the high frequencies that carry harmonic content (above 7kHz). Engage and disengage the **Bypass** switch in order to “A/B” the difference between loudness off and on, respectively.
4. To experiment with loudness compensation, move the loudness compensation slider to the left (less) or to the right (more).
5. Adjustments made to the loudness compensation slider carry through to all listening levels. Set the output volume fader in the DSP Configurator to a relatively loud listening level.
6. Engage and disengage the **Bypass** switch in order to compare the difference between loudness off and on. At a loud listening level, the difference should be minimal or barely perceivable.

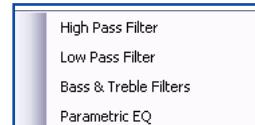
Delay Block (DLY)

The delay processor block, when inserted, provides a means to delay the audio signal to compensate for loudspeaker placement in situations where speakers delivering the same signal are much farther away than others. The delay processor block is identical to the delay processor available on the input and described in [Delay \(DLY\)](#) on page 48. Typically the near speakers would be delayed so that audio delivery time matches the speakers further away.



Filter Block (FILT)

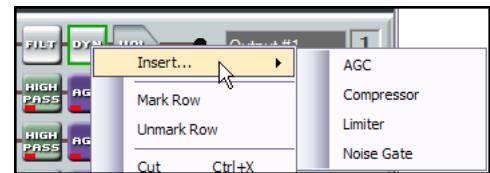
The filter processor block, when first inserted, provides one of four filter selections: High Pass, Low Pass, Bass & Treble filters and Parametric EQ. Up to nine filters can be added to each filter block. The output filter block is identical to the input filter processor block except that up to nine filters total can be selected [Filter \(FILT\)](#) on page 29.



NOTE: Selecting the **Bass & Treble Filter** inserts two separate filters.

Dynamics Block (DYN)

A dynamics processor block, when inserted, provides one of four dynamics processors: AGC, Compressor, Limiter, and Noise Gate. The available processors are identical to the processors available on the input dynamics processor block and described in [Dynamics \(DYN\)](#) on page 43.

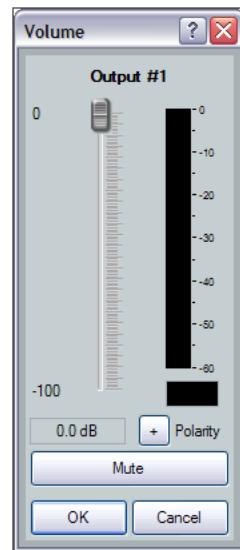


Volume Control (VOL)

Each output channel volume block provides a mono long-throw fader with a range of 0 to 100 dB of attenuation, and a volume setting readout (in dB) below the fader. Volume level is adjustable with the slider or by entering the desired level directly into the volume setting readout in 0.1 dB increments.

Clicking the fader handle or clicking within the fader area brings focus to the fader. The input signal level can be adjusted using any of the following methods:

- Click and hold the fader handle, then drag it to desired level in 1.0 dB steps.
- Click or tab to the fader handle, then use the <up> and <down> arrow keys to change the desired level in 1 dB steps. The keyboard <Page Up> and <Page Down> keys increase or decrease the level in 5 dB steps.
- Click in or tab to the level readout field. Type a new value, then press <Enter> or <Tab> to another area.



Output polarity switching is also provided with a button that toggles polarity.

The default setting is unmuted, at 0 dB attenuation. A peak meter displays the real-time audio level from – 60 to 0 dBFS.

The **OK** button accepts settings and closes the dialog with a click. The **Cancel** button ignores changes and closes the dialog.

The output volume control provides level control for each output. The output control is a trim control adjustable from – 100.0 to 0 dB. The default setting is unity gain (0.0 dB).

The **Polarity** button, accessible in the dialog box, allows the polarity of the wires connected to the audio connectors (+/tip and -/ring) to be flipped in order to easily correct for miswired connectors.

The **Mute** button, accessible in the dialog box, allows the post-meter audio output to be silenced. When the audio output is muted, the mute button lights red, and red indicators in the block turn on. If the output has been grouped with other inputs or outputs, the group number will be indicated on the right side of this button.

Virtual Bus Returns

There are four mono virtual bus return inputs, fed by the virtual bus sends. Channel controls and processing blocks described in the sub-sections that follow are identical for each virtual bus return channel, A through D.



The virtual bus is used when additional processing of an input signal is required. It is also useful to apply identical filtering, dynamics processing, loudness compensation, or signal gain/attenuation to multiple inputs.

Filter (FILT)

Filter function and interface is identical to the mic/line input channel Filter block with the exception that only three filters are allowed (see [Filter \(FILT\)](#) on page 31, for additional information).

Dynamics (DYN)

There is one dynamics processor block available on each virtual path. Dynamics function and interface is identical to the mic/line input channel Dynamics block (see [Dynamics \(DYN\)](#) on page 45 for additional information).

Loudness (LOUD)

There is one loudness processor available on each virtual path. The loudness function and interface is identical to the output channel Loudness block (see [Loudness \(LOUD\)](#) on page 56).

Gain (GAIN)

Each virtual input channel gain block provides a mono long-throw fader with a –100.0 to +12.0 dB gain range, and a level setting readout below the fader. Fader behavior is identical to the pre-mix-point gain block, described in the mic/line input section (see [Pre-mixer gain \(GAIN\)](#) on page 55). Fader adjustments are in 1 dB increments, while adjustments can be entered manually to 0.1 dB resolution.
Default is unmuted at unity gain (0.0 dB).

Primary Mix Matrix

The DSP architecture contains a primary mix matrix that connects the mic/line inputs and virtual bus returns to the line outputs. The DSP Configurator user interface provides control of the primary mix matrix, used to set mix levels from the post processing inputs and post processing virtual returns to each line output bus. Each mic/line input and virtual bus return is connected to a mix-point for each of the four line outputs. In general, mix levels are set relative to each other, achieving a desired blend of input signals at an optimal output level, close to, but not exceeding 0 dBFS at the line output volume block level meter (while accounting for processing that can occur in the line output signal chain).

Shown below is a block diagram of the DMP 64 represented in the DSP Configurator, with a red box indicating the primary mix matrix.

NOTE: Although the virtual bus send and return lines, A to D, are shown as end points in the block diagram below, they are connected A to A, B to B, C to C and D to D.

From the primary mix matrix, any or all of the six inputs can be routed to any or all of the four outputs. Any or all of the six inputs can also be routed to the secondary mix matrix.

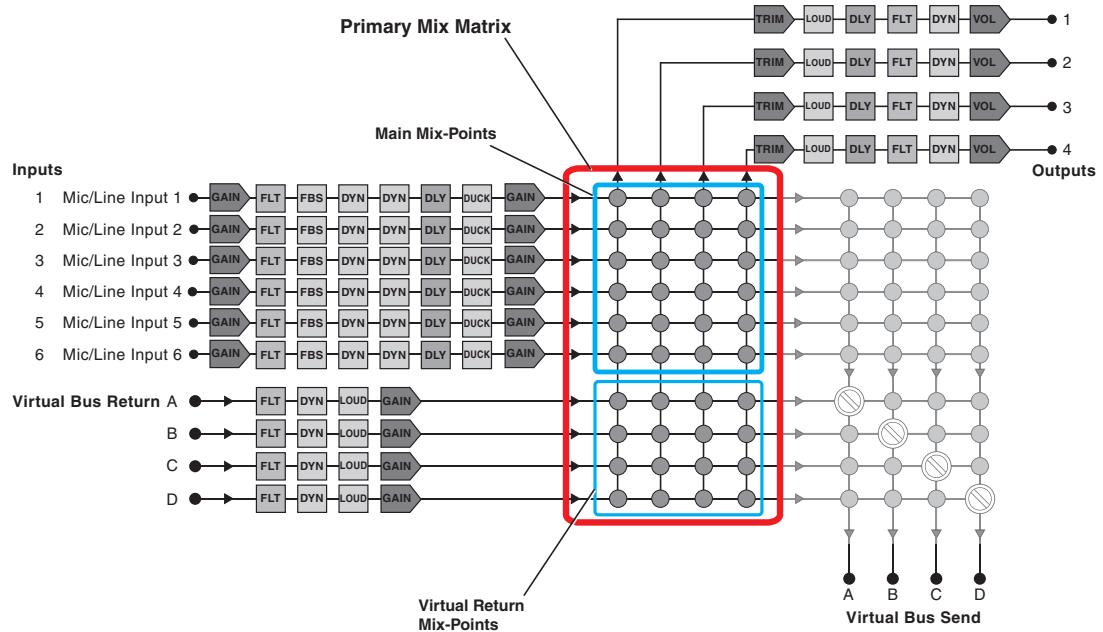
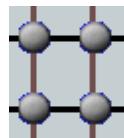


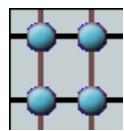
Figure 40. Primary Mix Matrix (outlined in red)

Mix-point Behavior:

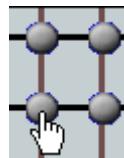
No mix information — A faint gray circle on the mix-point indicates it is muted (contains no mix information).



Mix information — A solid teal-colored circle indicates the mix-point contains mix information (the mix-point is unmuted).



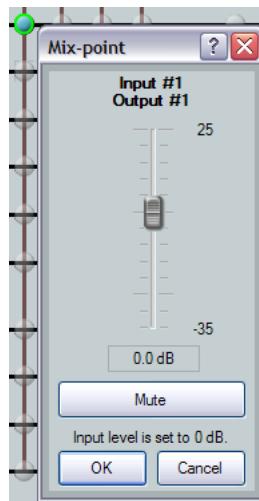
Mouse-over — The cursor changes to a hand when a mouse-over occurs at a mix-point, whether the mix-point contains mix information or not.



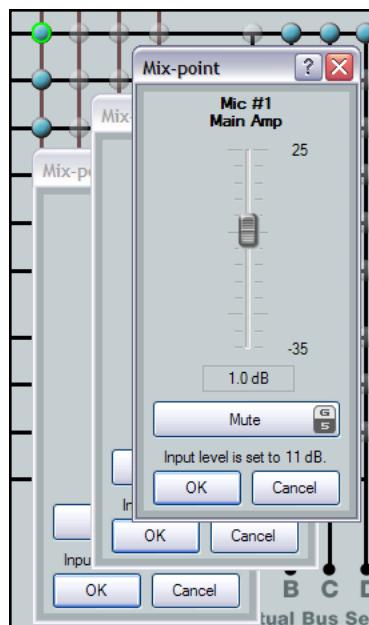
Single-click — A single-click brings focus to (selects) the mix-point, indicated by a dark green outline around circle.



Double-click — Opens the mix-point dialog box. The focus circle turns light green in color to indicate the open dialog box. If the mix-point is muted, the circle is gray. If unmuted, the circle is teal.



Multiple open dialog boxes — When multiple mix-point dialog boxes are open, the mix-point for the most recently opened dialog box receives the light green focus circle, while previously opened dialog boxes relinquish their focus. Focus can be returned by either clicking on a previously opened dialog box, or by double-clicking on a mix-point.



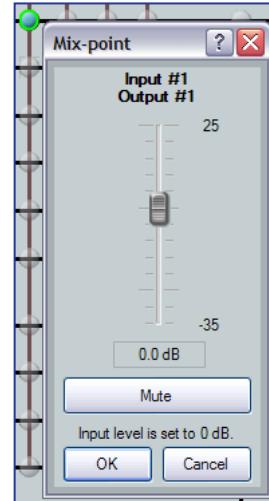
Clicking a mix-point brings focus to that mix-point. A circle appears around the teal mix-point which remains transparent. Double-clicking a mix-point opens a configuration dialog box with the following components:

- **Mono Fader** — Sets the signal level from the selected input to the output bus. Gain range is -35 dB to +25 dB. Fader behavior is identical to the input channel gain block described in the mic/line input section with the exception that coarse adjustment (keyboard <Page Up> and <Page Down>) increases or decreases in 5 dB increments.
- **Mute** — Mutes and unmutes the signal to the output bus. The mix-point ball is transparent when muted (Mute button red) and solid when unmuted.
- **OK/Cancel** — click **OK** to accept changes and close the box. **Cancel** ignores changes and closes the dialog box.

The title above the fader reflects the input and output channel names for the mix-point. The example on the right is the **Input #1** to **Output #1** mix-point set to 0.0 dB.

The input level text below the mute button indicates the input level setting for the input gain control of the selected input signal path, in this example 0 db.

Only when the mix-point is unmuted does the circle become solid.



Mix-point Examples

In order to better understand how mix-points work, the following diagrams provide examples of different mix setups.

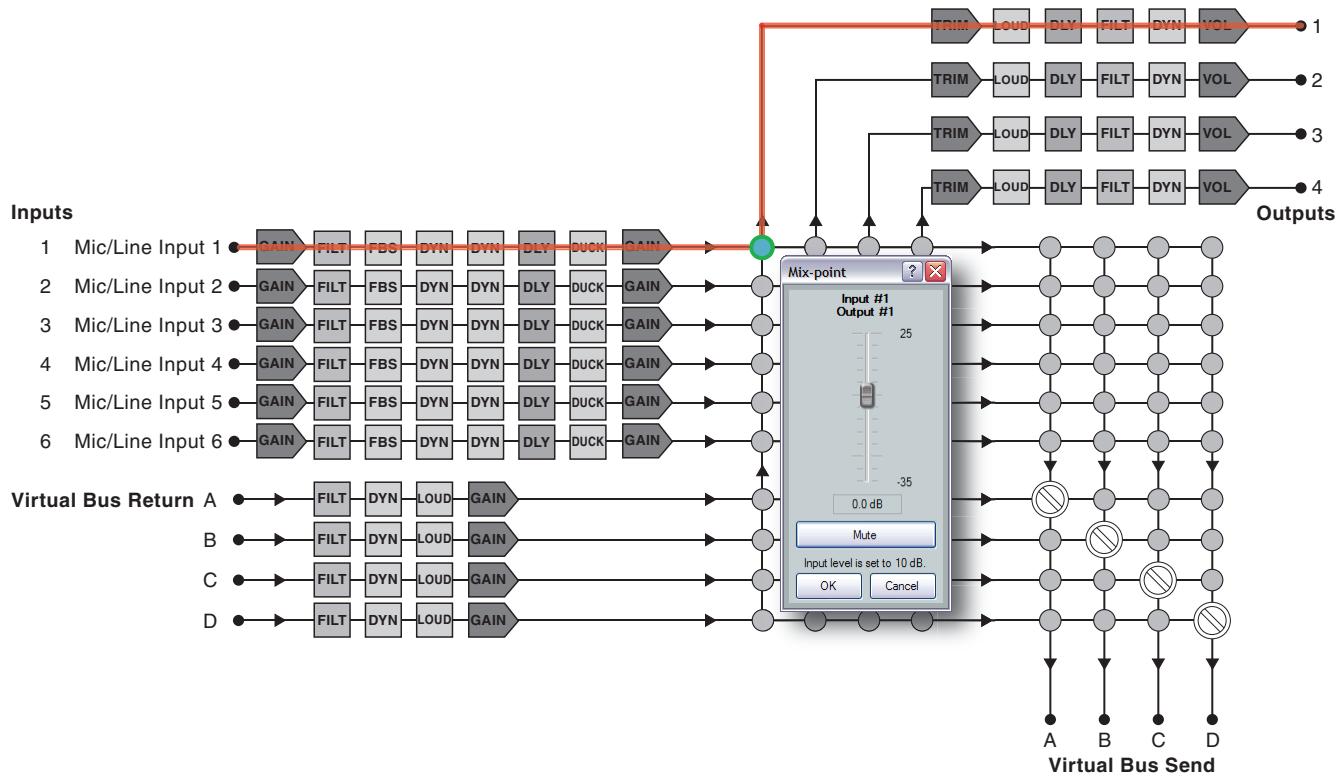


Figure 41. Input 1 to Output 1

In the first example (see figure 41), input audio from Mic/Line Input 1 is processed and arrives at the output mix-point. Double-click on the mix-point to open the dialog box. The mix-point opens muted (mute box is red). When the mute button is released, the mix-point turns teal with a light green circle to indicate the open mix-point dialog box is the focus, and the signal is routed to output 1.

The mix level can be adjusted using the slider or by direct input of a value between -35 and 25.0 dB into the dialog box below the slider.

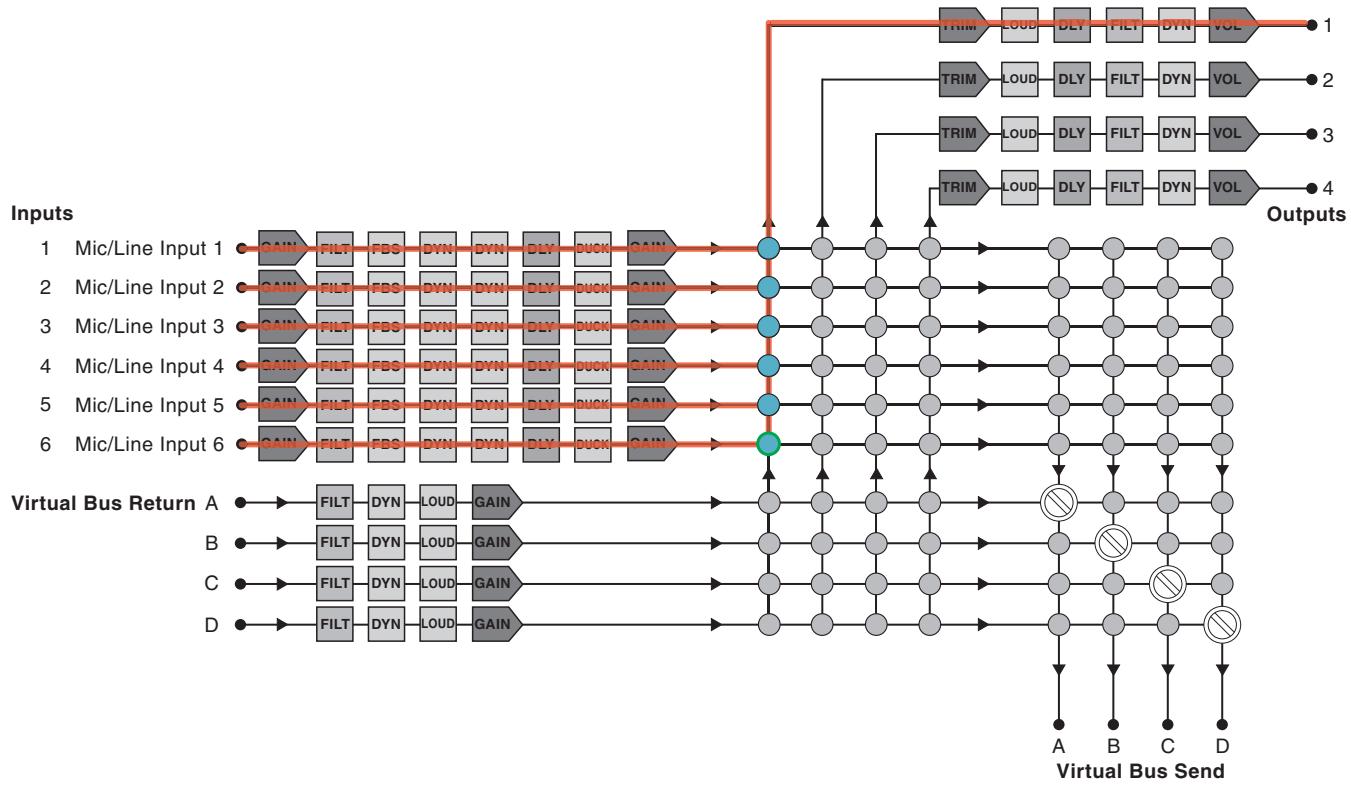


Figure 42. All Inputs to Output 1

In the next example (see figure 42), input audio from all six mic/line inputs are processed individually and arrive at the primary mix-points. When the individual mix-point mute buttons are released, the primary mix-point junctions turn teal, and all six signals are routed to output 1. Since all six inputs are now on the output 1 signal line, open the individual mix-point dialog boxes to adjust signal levels for the desired balance.

In this manner, any single input, or any number of inputs can be routed to any single output or any number of outputs.

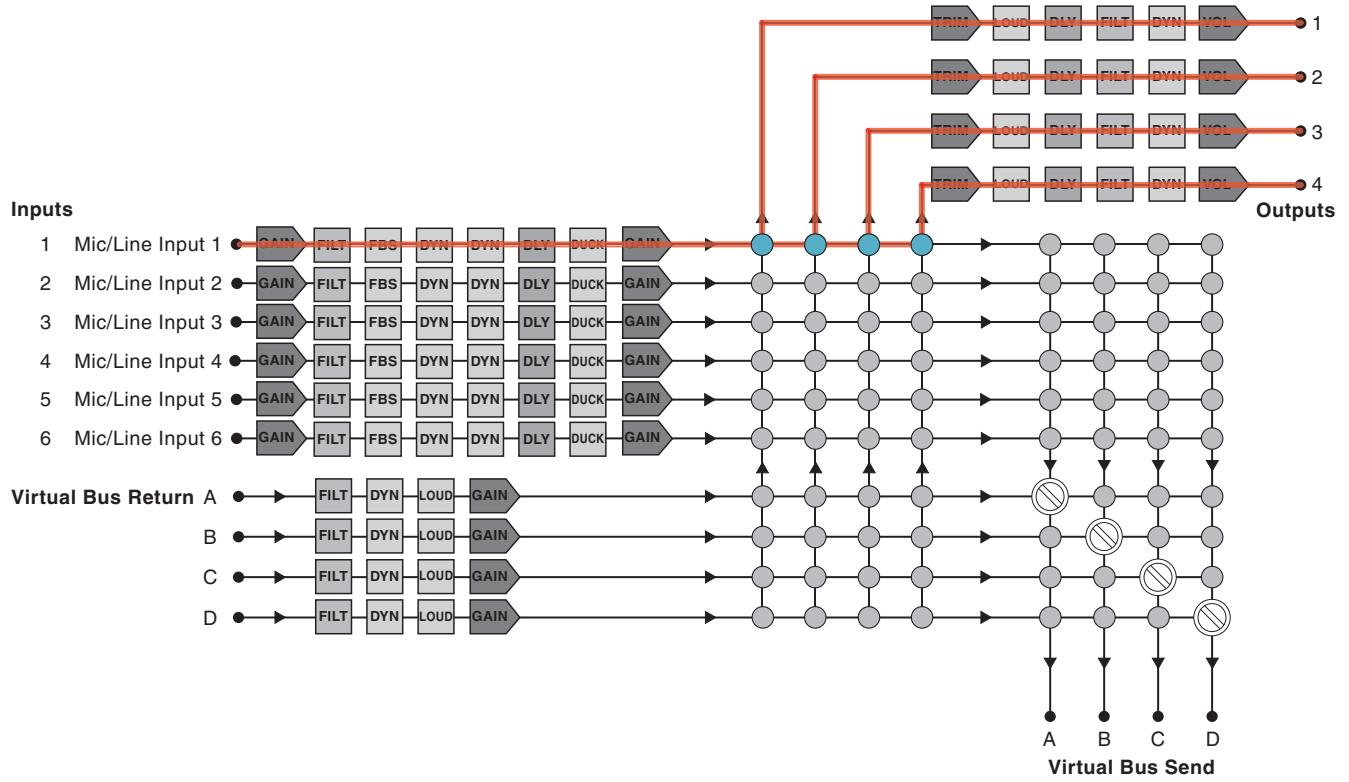


Figure 43. Input 1 to All Outputs

In this example (see figure 43), input 1 has been routed to all four outputs by unmuting the mix-point for mic/line input 1 on each output (1 to 4) bus. Again, the primary mix-points are teal to indicate the active routing.

Secondary Mix Matrix

The DSP architecture contains a secondary mix matrix that connects the mic/line inputs and virtual bus return signals to the virtual bus sends. The DSP Configurator user interface provides control of the secondary mix matrix, used to set levels from the post-processing input line and virtual bus return signals to the virtual bus sends. Each of the six mic/line and four virtual return inputs connect to a mix-point for virtual bus A through D. Each mix-point is muted and set to 0.0 dB (unity gain) by default. In general, mix levels are set relative to each other, achieving a desired blend of input signals at an optimal level close to, but not exceeding 0 dBFS at the output volume level meter.

The secondary mix matrix contains a section (see figure 44) allowing virtual bus returns to be routed back to the secondary matrix to allow further processing using an additional virtual bus processing block. To prevent feedback loops, a virtual channel is prevented from being routed back to itself by eliminating the mix-point that would allow that to occur.

In situations requiring extra processing, the virtual bus return output is routed back to the secondary mix matrix, virtual bus send, which then routes the signal back to a processing signal chain other than the one it was routed from.

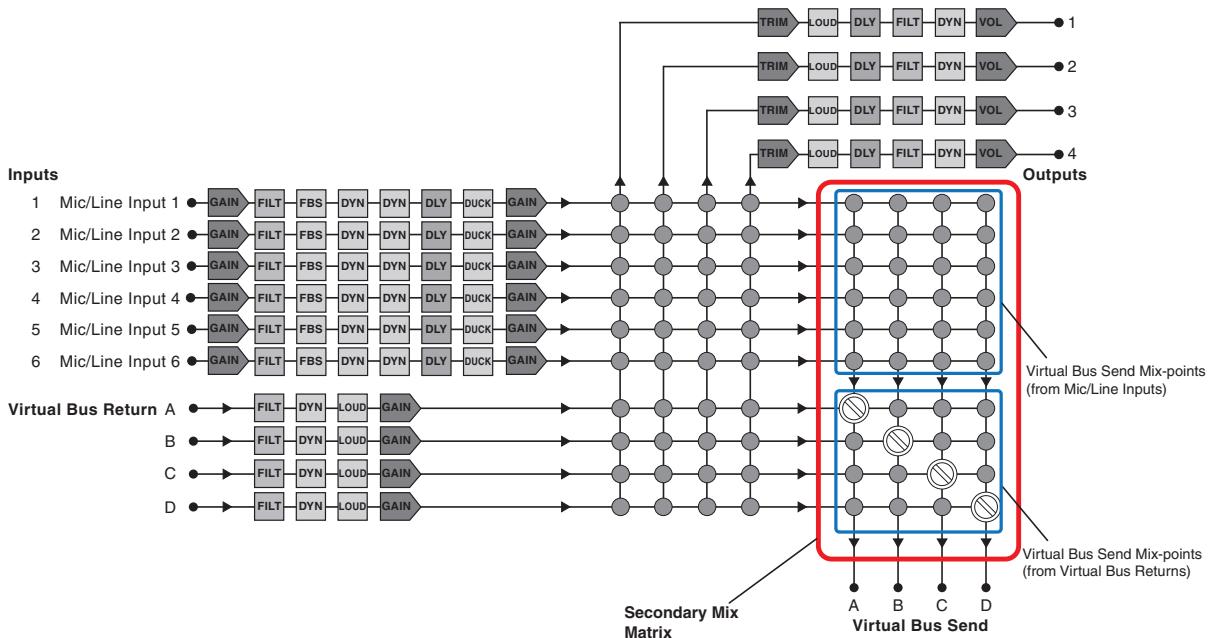


Figure 44. Secondary Mix Matrix

In the example below (see figure 45), input 1 is sent to the virtual bus send by muting all four signals on the input 1 primary mix-points. The virtual bus now serves as additional signal processing for the input. The signal routes over virtual bus A and through the signal chain before being sent to the virtual bus return mix-point and output 1.

This configuration is useful when more than one input requires identical processing. For example if all inputs were normalized but required a uniform gain to bring them up to adequate output levels, rather than changing each pre-mix gain control by a similar amount, all six inputs can be routed to virtual bus A. Then, using the virtual bus A return gain control, a single adjustment can apply the same gain to all six inputs before sending the signal to the desired output line.

In other cases, if multiple microphone inputs are being mixed with program material, only the program material might require loudness contouring. The microphones can be routed directly to the output while the program material input is routed to the virtual bus return where loudness contouring can be applied. The program material is then routed to the same output as the microphones.

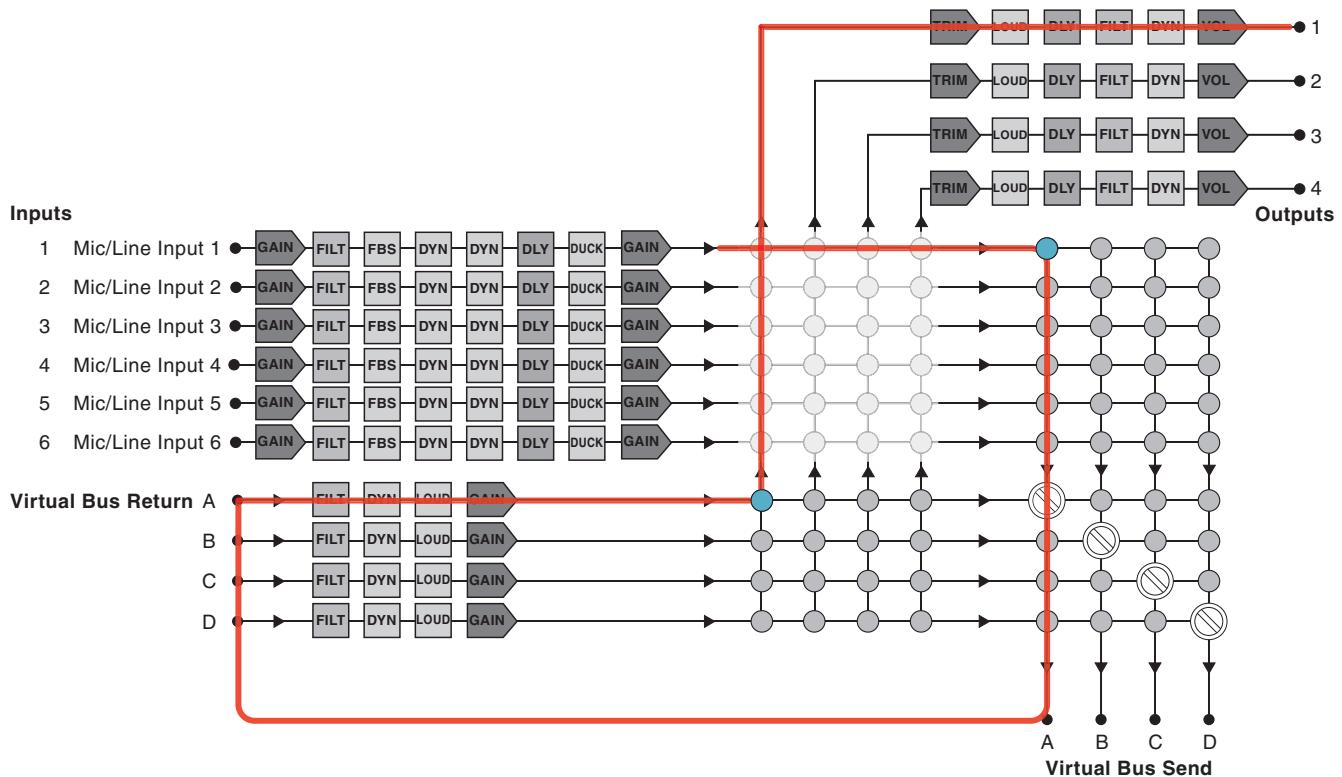


Figure 45. Input 1 to Virtual Bus A

Group Masters

There are 32 Group Masters that can each be configured to simultaneously control up to 16 group members. Group masters are configured in the DSP Configurator program and saved in the device. Working in emulate mode, group masters can be saved in a configuration file and pushed to the device upon connection.

A group master can either be a gain control or a mute control. Only one control type can be selected as group members for control by a group master. For example, a group master can be configured to control post-matrix gain levels, but not post-matrix gains plus input gain block. A group member can, however, be controlled by multiple group masters. It is recommended this feature be used cautiously, as overlapping membership can quickly become unmanageable.

Group master gain controls can send specific values, such as those sent by a fader control. Group master gain can also be set by increment/decrement. For information on using increment/decrement controls within the DSP Configurator software (see **Tools** on page 73).

Group Members

Once a group has been created, the group members, the individual controls that comprise the group, update to indicate they are now part of a group. Group members can be controlled individually, allowing for relative levels between group members to be fine-tuned. Group member levels can also be set by a preset recall.

Grouped Controls

Grouping is convenient when multiple controls require muting at the same time or when multiple signal levels need to be increased or decreased simultaneously. For example, in a system with several audio outputs dedicated to a single room, the operator may want all outputs to change at the same rate and at the same time. The output 1 through 4 volume controls can be grouped into a master that controls the volume throughout the room.

For further flexibility, individual volume controls in the group can be set for an output level based on its use. When the group fader is moved, all four output control faders move in tandem while retaining their levels relative to each other.

Grouped faders move together at relative levels to the top or bottom of their travel (see Figure 46). If one fader reaches the limit of its travel first, it retains that position while the other faders continue to travel. When the grouped faders travel in the reverse direction, the fader that was at its limit reverts to its position relative to the other faders.

NOTE: If a block was previously muted when the group mute is activated, that block remains muted when the group mute is released.

TIP: When including a control in multiple groups, do so with care. Overlapping group membership can quickly become unmanageable. Use presets to set individual faders to known levels.

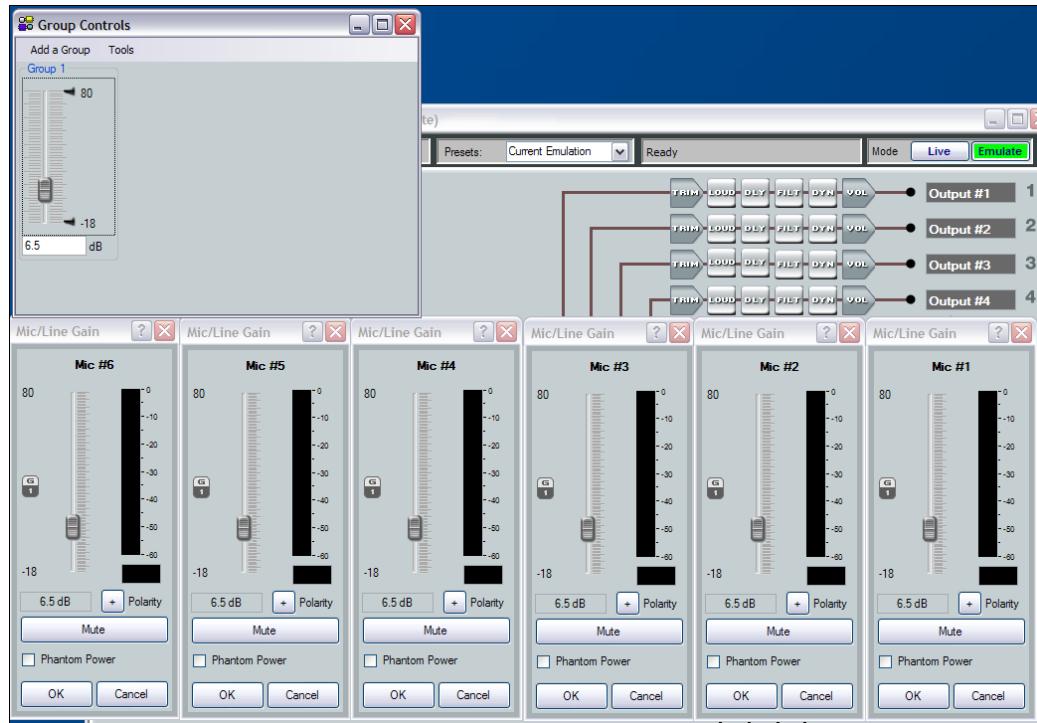


Figure 46. Sample Fader Group Master and Associated Gain Controls

Mute controls within the blocks can also be grouped (see figure 47).

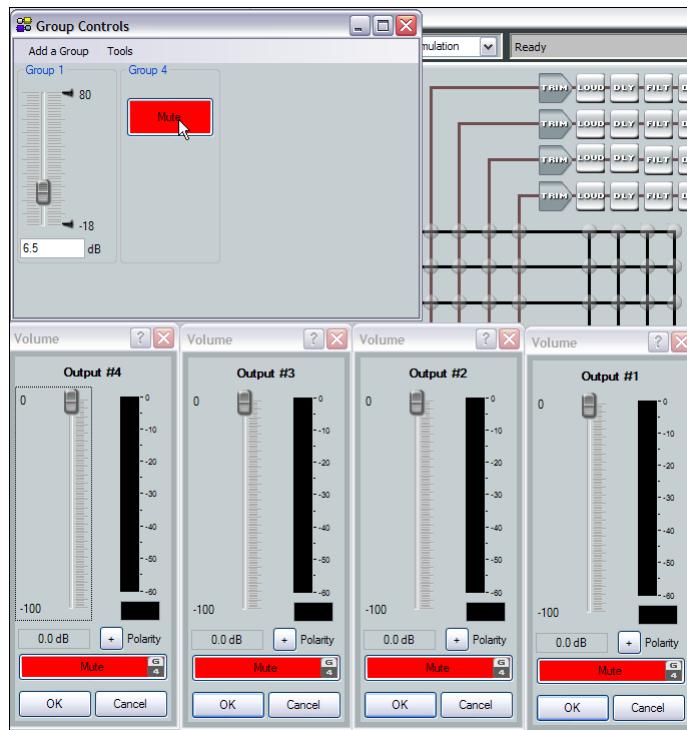


Figure 47. Sample Mute Group Master and Muted Outputs

Configuring a Group Master

Configure a group as follows:

1. Click **Tools > Configure Groups** to open the Configure Groups dialog box, or click **View > Group Controls** and then click the **Add a Group** menu selection.
2. In the **Select Group** drop-down box, click a group to select it (see figure 48). The list defaults to the first empty group. Select an empty group to begin a new group, or select an existing group to modify.

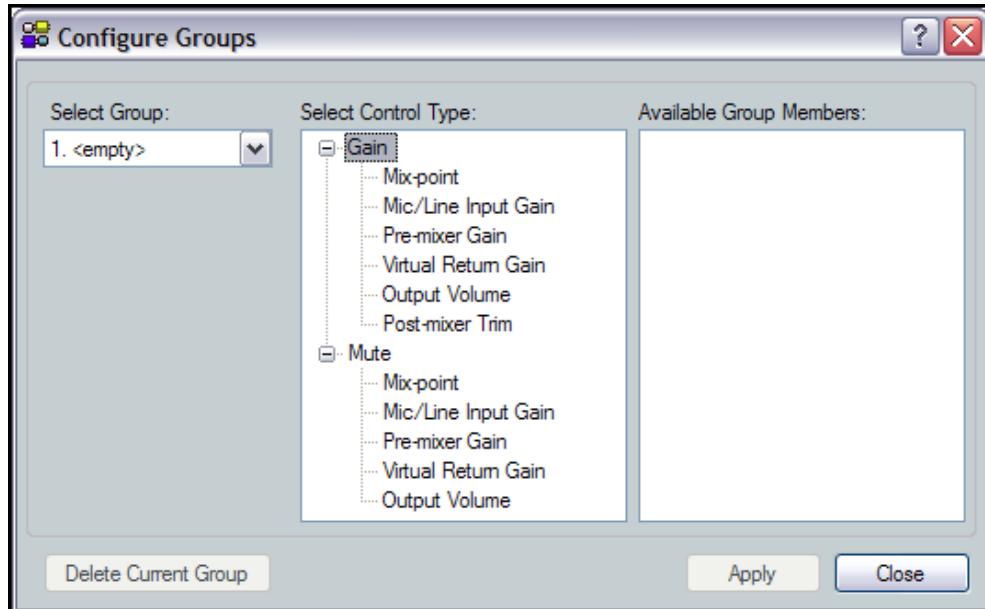


Figure 48. Configure Groups Add Group Dialog Box

NOTE: <empty> groups have no group members assigned. Numbered groups (such as <Group #1>) have controls assigned that can be overwritten if selected.

3. In the **Select Control Type** section, expand the tree for the type of control, **Gain** or **Mute**, then select the desired control type. When a selection is made in the **Select Control Types** section, the **Available Group Members** section populates with all possible members for the selected control type.

NOTE: Potential group members in step 3 that are assigned to a different group are displayed in blue.

4. In the **Available Group Members** section, make appropriate selections by clicking the checkbox. When a + sign exists, click to expand the tree and select individual controls. Up to 16 group members can be added.
5. Click the **Apply** button to create or configure the group.
6. Repeat steps **2** through **5** to create or configure up to 32 groups.
7. Click the **Close** button to exit the configure groups dialog box.

Deleting a Group Master

To delete a group:

1. Click **Tools > Configure Groups** to open the configure groups dialog box or click **View > Group Controls** and then click **Add a Group**.
2. In the **Select Group** drop-down box, click a numbered group (such as "Group #1") to select it.
3. Click the **Delete Current Group** button in the lower left area.
4. Click **Yes** in the **Confirm Deletion** dialog box.

Viewing and Using a Group Master

Click **View > Group Controls** to open the group controls dialog box (see figure 49). The group controls dialog contains two menu items:

- **Add a Group** allows you to add additional groups.
- **Tools** enable you to perform various functions from the group controls window.

In addition, once groups are created, a single mute button or a group fader plus the current setting readout and any soft limits that have been set are visible.

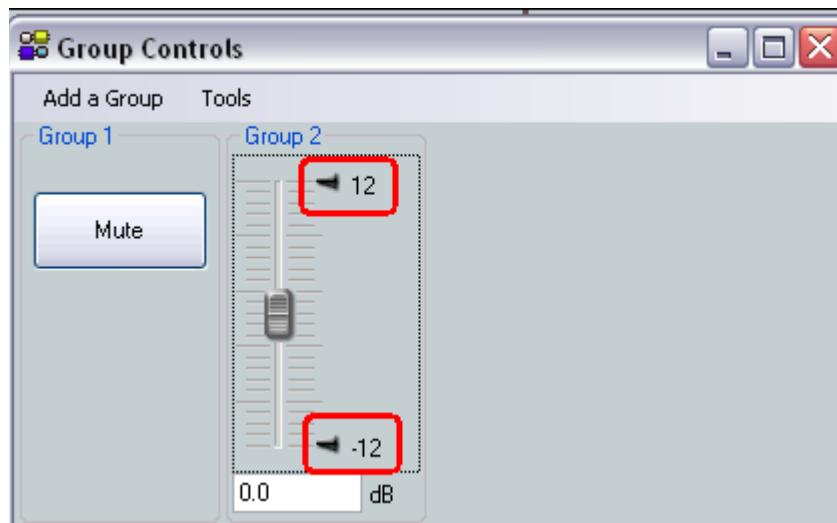


Figure 49. Group Controls Dialog Box

The group fader controls function as follows:

- Slide a group fader up and down to adjust all gain controls in the group.
- Click and drag a soft limit (■) to set the ceiling and floor for the group.

NOTE: Soft limits cannot be dragged beyond the current setting of the group fader.

Add a Group

To launch the configure groups dialog from the group controls window, click **Add a Group**. When a new group is added and the **Add New Group** dialog is closed, the group controls window refreshes to display the added control.

NOTE: If a block is muted when the group mute is selected, that block remains muted when the group mute is released.

Tools

The Tools menu contains three selections:

- **Clear All Groups** - clears all group members and group master parameters.
- **Increment/Decrement Simulator** - allows the user to test increment/decrement values, see below for more information.
- **Group Details Report** - generates a report, listing all group masters and membership.

Clear All Groups

Click **Tools > Clear All Groups** to delete all groups and reset all group memberships. Soft limits are also cleared.

Increment/Decrement Simulator

The Increment/Decrement Simulator provides a control for increment and decrement, with the ability to set increment and decrement values. This control is temporary, since the value is not remembered in the device.

To use the Increment/Decrement Simulator:

1. Select **Tools > Increment/Decrement Simulator**.
2. Select the group to be controlled from the **Select Group** drop-down list. The following dialog box appears:

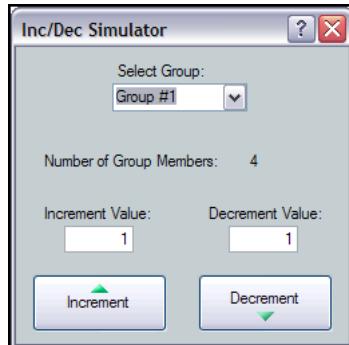


Figure 50. Increment/Decrement Simulator Dialog Box

NOTE: The **Number of Group Members:** readout indicates the number of controls affected.

3. Enter an increment value and a decrement value. The default value is 1.

NOTE: The size of the increment can be changed by typing a value in the **Increment Value** or **Decrement Value** field. Values can be as large as the maximum range of the control or as fine as 0.1 dB. For groups controlling mute, 1 is the only valid value.

4. Click the **Increment** and **Decrement** buttons as needed. The group master control increases or decreases by the set value to the top or bottom of its soft limit range.

NOTE: When set, soft limits cannot be exceeded.

Group Details Report

Select **Tools > Group Details Report** to create a Microsoft Word file that details all created groups (see figure 51).

GROUP DETAILS REPORT

Group #1

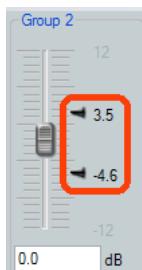
Processor Type: Output Volume
Current Mute status: Unmuted
Current Group Members:
Main Amp (Output#1) Left Channel
Stage Mixer (Output#2) Right Channel
House Video (Output#3) Left Channel
Prgm Record (Output#4) Right Channel

Group #2

Processor Type: Pre-mixer Trim
Current Gain value: 2 dB
Current Group Members:
Mic #1 (Input#1)
Mic #2 (Input#2)
Mic #3 (Input#3)
Mic #4 (Input#4)
Mic #5 (Input#5)
Mic #6 (Input#6)

Figure 51. Sample Group Details Report

Soft Limits



Each gain type control provides upper and lower soft limits that can limit the range of the group master control. **Soft limits** (), shown at left, prevent group controls from exceeding an upper limit or going below a lower limit. They are easily adjustable and provide the ability to set a ceiling and floor for the group. When a group master is created, the soft limits default to the hard limits (maximum and minimum) of that group of controls.

Soft Limits can be defined using the mouse by dragging the **soft limit** icon. The resolution is 0.1 dB. For more precise setting use the keyboard as follows:

Click within the group master fader to bring focus, then use the following key combinations:

To move the upper limit:

- <Shift + Up/Down Arrow> key moves in 0.1 dB increments.
- <Shift + Page Up/Page Down> key moves in 10 dB increments.
- <Shift + Home> moves limit to upper default.
- <Shift + End> moves limit to the current fader position.

To move the lower limit:

- <Ctrl + Up/Down Arrow> key moves in 0.1 dB increments.
- <Ctrl + Page Up/Page Down> key moves in 10 dB increments.
- <Ctrl + End> moves limit to lower default.
- <Ctrl + Home> moves limit to the current fader position.

Digital I/O Ports

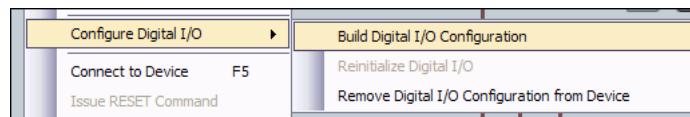
The DMP 64 provides six digital I/O ports that can trigger external events from DMP 64 actions, or allow external events to trigger DMP 64 actions. The DSP Configurator software provides pre-configured scripts with a fixed set of common trigger and event combinations. When selected, the script is compiled and placed onto the File Management system of the device. For more advanced or custom scripts, contact an Extron Electronics Applications Engineer.

When no scripts are active, the digital I/O ports default to DI (digital input) and inactive ('Logic Hi' ≈ +5 VDC). The DI detects a Logic Hi as +5 VDC and Logic Low (active) as less than +1 VDC.

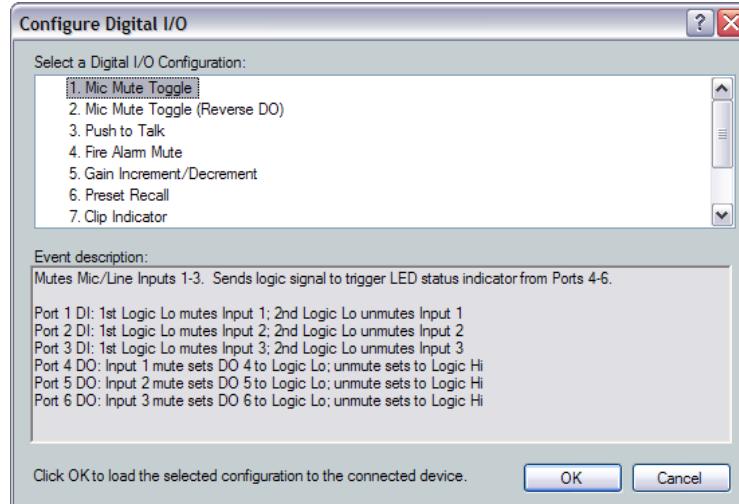
A DO (digital output) sends a Logic Lo as less than +1 VDC and a Logic Hi as +5 VDC. For every script that involves a DO, two versions are available to provide either a Logic Hi or a Logic Lo response to any action. The alternate script is designated as Reverse DO.

To build a script and place it into the DMP 64 File Management system:

- From the tools menu, click **Configure Digital I/O>Build Digital I/O Configuration**.



- This brings up a dialog that allows selection from a list of pre-configured scripts.
- Select a script from the **Select a Digital I/O Configuration** section. The event description section describes the script and how the digital I/O ports act while the script is running. Highlight the desired script, then click **OK**.



- A dialog box appears, verifying the file has been successfully uploaded to the device.

NOTE: When performing this procedure in Emulate mode, the connection dialog appears between step 3 and step 4. DSP Configurator connects and then disconnects during the procedure, returning to Emulate mode when the Digital I/O has been configured.

Reinitialize Digital I/O

Should the script stop running for any reason, select **Tools > Configure Digital I/O**, then select **Reinitialize Digital I/O**. This option is only available in **Live** mode.

To remove a digital I/O script from the DMP 64:

Only one digital I/O configuration can be active at a time. If the I/O activity needs to be modified, remove the current configuration by:

1. From the **Tools** menu, click **Configure Digital I/O**, then select **Remove Digital I/O Configuration from the Device** and click **OK**.
2. If the DSP Configurator is connected to a device, the I/O configuration is removed. If it is not connected, a connection dialog box appears.
3. Make certain the connection information is correct, then press **OK**. The I/O configuration script is removed and a confirmation dialog box will appear.

Emulate Mode and Live Mode

The DSP Configurator program has two operational modes, **Live** and **Emulate**. In **Live** mode, the program has established a connection and is synced with the DMP 64. Changes affect the device in real-time and changes in the current state of the device are reflected in the DSP Configurator. In contrast, **Emulate** mode allows the user to work offline, creating or editing configurations that do not immediately affect DMP 64 operation.

The DSP Configurator program always starts in **Emulate** mode. All functions of the DSP Configurator program are available without connecting to the DMP 64. The user can build a configuration from the blank screen, or open an existing file that contains the last configuration displayed plus saved presets. All settings and adjustments are saved to a configuration file on the PC. When the saved file is opened in the DSP Configurator program, the program restores all settings as the current configuration (emulated if in **Emulate** mode or live if in **Live** mode).

Live mode can be entered at any time after program launch, either with a blank configuration, after creating a configuration, or after loading a previously saved configuration file.

In **Emulate** mode, the current state is titled **Current Emulation**. In **Live** mode, the current state is titled, **Current State**.

Synchronizing: Pull From or Push To the DMP 64

When switching to **Live** mode after making changes to the current configuration in **Emulate** mode, either:

- **Pull** data from the device and update the DSP Configurator program configuration. This option downloads device settings from the DMP 64 and synchronizes it with the DSP Configurator program overwriting the current DSP Configurator settings, or
- **Push** data from the DSP Configurator program to the device, overwriting settings in the DMP 64.

Live mode can also be used to tailor audio settings in real time while listening to the audio output.

Selecting Live Mode and Pushing or Pulling Data

To switch from Emulate mode to Live mode:

1. Select the desired connection to the DMP 64 and make the proper connections.

NOTE: Extron recommends connection with the Ethernet LAN port when using DSP Configurator.

2. Click the Mode **Live** button, (see figure 52, ②). The **Communication Type** selection window appears.

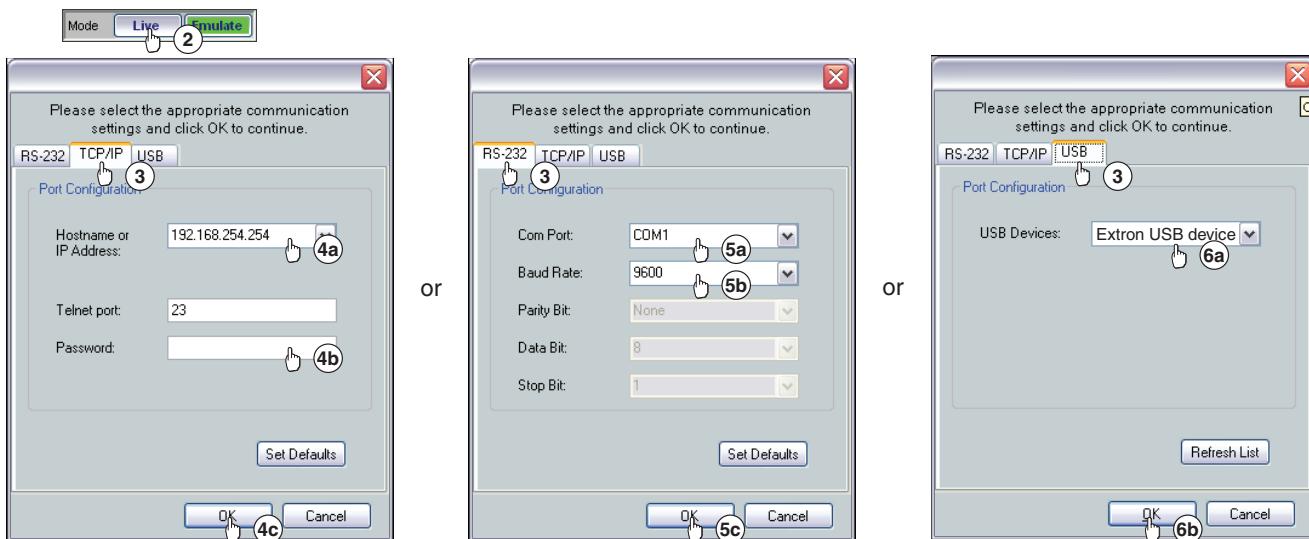


Figure 52. Selecting Live Mode

3. Click one of the following tabs:

- **TCP / IP** for connection using the LAN port (**preferred**) — proceed to step 4,
- **RS - 232** for connection using either of the rear panel RS-232 ports — proceed to step 5, or
- **USB** for connection using the front panel configuration port — proceed to step 6.

4. If TCP/IP is selected in step 3:

- a. Observe the **IP Address** field in the IP connection window. The field displays the last IP address entered.
 - If the IP address is **correct**, proceed to step 4b.
 - If the IP address is not **correct**, either click in the **IP Address** field and enter the IP address or click the drop-down arrow to open a list and select from a recently used address. Proceed to step 4b.

NOTE: If the local system administrators have not changed the address, the factory-specified default, 192.168.254.254, is the correct value for this field.

- b. If the device is password protected, click in the **Password** field and enter the appropriate administrator password.

- c. Click **OK**.

The **Synchronize with Device** dialog box (see figure 53 on page 79) opens. Proceed to step 7.

5. If RS-232 is selected in step 3:

- a. Click the **com port** drop-down list and select the PC comm port connected to the rear panel RS-232 port.
- b. Check the baud rate displayed in the Com Port selection window. If the baud rate does not match the device's rate, click the **Baud Rate** drop-down menu and select the desired baud rate. The default is 38400.
- c. Click **OK**.

The **Synchronize with Device** dialog box (see figure 53 on page 79) appears. Proceed to step 7.

6. If USB is selected in step 3:

- a. Click the **USB Device** drop-down menu and select **DMP 64** (or **Extron USB device**, if **DMP 64** is not available),
- b. Click **OK**.

The **Synchronize with Device** dialog box (see figure 53) appears. Proceed to step 7.

7. Click either the:

- a. **Pull** radio button to configure the DSP Configurator program to match the device — proceed to step 9

-Or-

- b. **Push** radio button to configure the device to match the DSP Configurator program — proceed to step 8

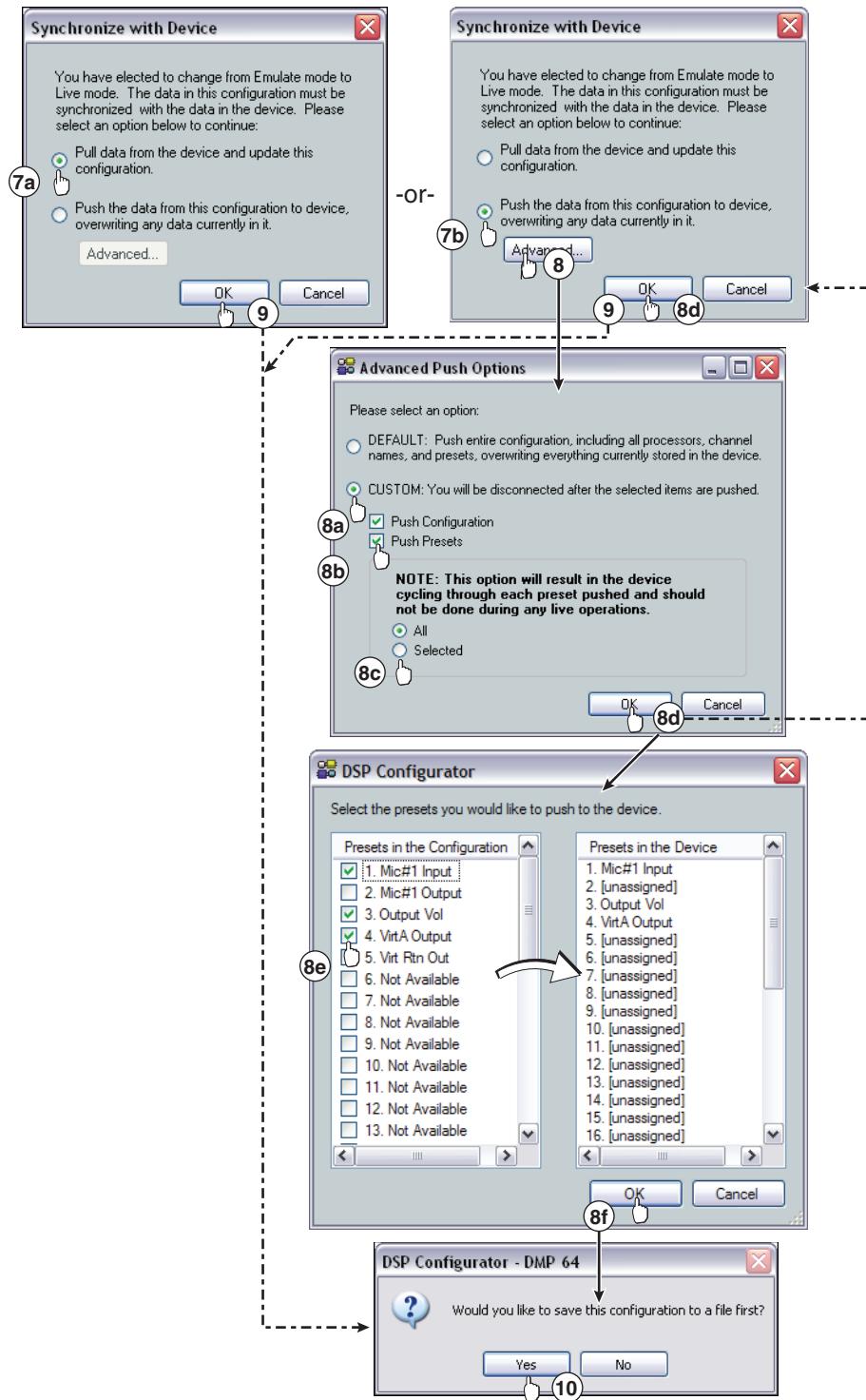


Figure 53. Selecting Live Mode, continued

- 8.** To push all of the DSP Configurator gain and processor block adjustments (configuration), and all presets to the DMP 64, proceed to step 9.

To tailor the push (push only the configuration, only the presets, or the configuration and selected presets), click the **Advanced** button and proceed to step 8a.

- a.** Select the **Custom** radio button.
- b.** Select the desired checkboxes: **Push Configuration**, **Push Presets** or both. If **Push Configuration** is the only box checked, click **OK** and proceed to step 9.

NOTE: **Push Configuration** includes all mix-point, gain, and processor block settings. It does not include partial presets.

- c.** If **Push Presets** was clicked in step 8b, click **All** to select all presets or **Selected** to choose specific presets.
 - If **Selected** was clicked, click **OK** and proceed to step 8d.
 - If **All** was clicked (equivalent to a standard push), click **OK** and proceed to step 9.
- d.** If **Selected** was clicked in step 8c, the Synchronize with Device dialog box (7b) reappears. Click **OK**. The presets dialog box appears.
- e.** Select the desired partial presets to push by clicking the appropriate checkbox.
- f.** Click **OK**. — Proceed to step 10.
- 9.** Click **OK**. The DSP Configurator program is connected live to the device, the processors and presets are pushed or pulled as selected, and the selection of **Live** mode is complete.
- 10.** If changes were made to the DSP parameters (including mix-point, gain or processor blocks) since the last file save, the DSP Configurator prompts to save the file. Click **Yes** or **No**.

If a password is required and not entered or if an incorrect password is entered, the program prompts for the password.

The configuration and presets are uploaded to the DMP 64.

Presets

Presets recall a group of frequently used settings. Presets created by DSP Configurator can contain all elements (gain blocks, processor blocks, and mix-points) or a portion of the elements available within the program. In **Emulate** mode, up to 32 partial presets can be created, uploaded as a set, and stored to the device or a disk as a configuration file. In **Live** mode, presets can be created one at a time from the current state. They can then be saved to a chosen preset number in the device, with the option to name or rename, or save it to disk.

When recalled, a preset overwrites only elements contained in the preset. Presets are useful when settings for a particular room or only certain elements of a configuration need to be changed regularly.

Presets can be created in **Live** or **Emulate** modes. In **Emulate** mode, the presets are created, saved to a file, then pushed to the DMP 64 when connecting in **Live** mode.

When a **pull data** synchronization is performed, preset data remains in the DMP 64, with only the list of preset names pulled from the device. Presets in this state are marked with an asterisk until that preset is recalled (which pulls the preset data from the device), or until a backup is performed (see **Backup** on page 26). Presets pulled from the device cannot be saved to disk until they have been recalled, at which time the preset data is pulled into the DSP Configurator. Presets with no asterisk can be saved to disk.

Saved presets can be recalled via the DSP Configurator, or a control system sending an SIS preset recall command. Presets can also be saved and recalled via the embedded web page. Presets saved using the web page interface contain input gain, output volume, and the primary mix-point settings.

Previewing and Recalling a Preset

A preset can be previewed in either **Live** or **Emulate** mode by selecting the preset from the preset drop-down list.

The program indicates a view-only preset configuration by displaying each preset element with a translucent green mask over the block.

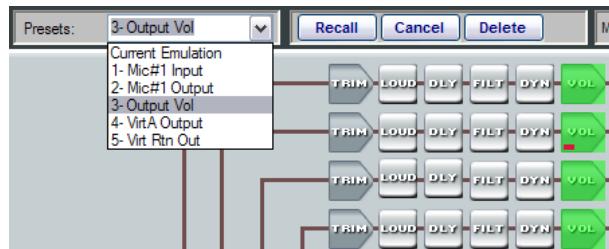


Figure 54. Preset Preview

Behavior for previewing and applying presets is as follows:

- **Live Mode.** After selecting a preset, the DSP Configurator displays the preset elements that will be affected by a preset recall with a translucent mask over the element. It leaves all other DSP Configurator elements unaltered. Elements without a translucent mask represent elements in the current state that will be unaffected by a preset recall. Real-time changes to the current state are not reflected onscreen while previewing a preset, and the user cannot alter those elements. To apply the preset, click **Recall**. The preset reverts to **Current State**.
- **Emulate Mode.** After selecting a preset from the list the DSP Configurator displays the elements that will be affected by a preset recall with a green translucent mask, leaving all other elements (which represent the current emulation) unaltered. Click **Recall** to apply the viewed preset to the current emulation. The preset number reverts to **Current Emulation**.

Building a Preset

Only elements of the preset highlighted (given focus) are saved as a preset. <Ctrl + A> highlights all elements within the DSP Configurator.

To build a preset, highlight the desired DSP Configurator elements (gain and processor blocks, and mix-points) using standard keyboard and mouse actions as follows:

1. Click the desired block to select a single block,
2. <Ctrl + click> to select multiple blocks that are not adjacent,
3. With the first block selected, hold <Shift>, then click the last block in either a vertical column or horizontal row to select multiple blocks,
4. Click and drag a selection rectangle to select multiple adjacent blocks in either the vertical or horizontal direction,
5. Go to **Tools>Presets>Mark All Items** or press <Ctrl + A>. This marks all elements within the DSP Configurator.
6. To save the selections see **Save Preset** below.

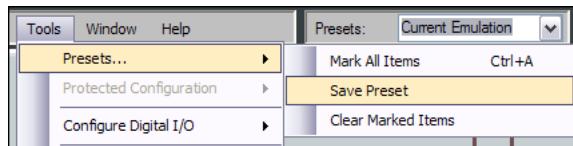
Save Preset

A preset can be saved in either **Emulate** or **Live** mode.

Saving a preset in **Emulate** mode stores it in the currently open file. The DSP Configurator file must then be saved to disk using **File menu>Save** (recommended), or pushed to the device after a connection is established. This differs from **Live** mode where a created preset is saved in real-time to the device and becomes part of the configuration file.

To save a preset use the following instructions:

1. Highlight the desired preset blocks using the previous instructions.
2. Select **Tools>Presets>Save Preset** in the main structural menu.



3. Select a preset number. In the **Preset Name** field, unused presets are named **unassigned**. To create a new preset, select an unused preset number and type a preset name. If no name is entered, a default name will be assigned. To overwrite an existing preset, select a preset with a name other than **unassigned**.

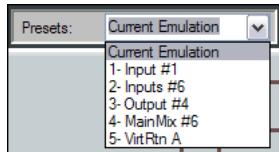


Figure 55. Save Preset

4. Click **OK** to save the preset, or **Cancel** to stop the save preset operation.

Managing Presets

Once a preset is created (whether or not the DSP Configurator file is saved), it appears in the preset list, available from the DSP Configurator user interface.



In **Live** and **Emulate** mode, after a preset is selected from the list, action buttons become available next to the presets bar.



The user can either **Recall** (make the preset active), **Cancel** (return to the current emulation or state), or **Delete** the preset.

In **Live** mode, selecting **Recall** applies the currently displayed preset elements ("marked" elements) from the stored preset, overwrites that portion of the current state, and then switches the drop-down list to read **Current State**.

In **Emulate** mode, the **Recall** action button applies the currently displayed preset elements ("marked" elements) from the file, overwrite the information contained in the DSP Configurator as the current emulation, then switch the drop-down list to read **Current Emulation**.

When a preset is previewed in either **Live** or **Emulate** mode, the **Delete** button is available. When pressed in live mode, the preset is deleted from the hardware, which is reflected in software (it is removed from the preset list). After disconnecting from the device and before exiting the program, the file must then be saved to retain this change. If pressed in emulate mode, the preset is deleted from the file in software, which must then be saved (before exiting) to retain this change. In either mode, the **Cancel** button defeats the preview action and returns the user to current state or current emulation, respectively.

Presets: Pull vs. Push or Create Live

When a preset is pulled from the device, the preset data remains in the device until the preset is recalled. The DSP Configurator pulls only the names of the presets. These presets cannot be saved to disk until they have been recalled.

An asterisk next to the preset name indicates that only the preset name has been pulled from the device, and the preset data exists only in the device (it has not been recalled). Presets pushed to the device or created in **Live** mode have no asterisk. Presets with no asterisk can be saved to disk.

Protected Configuration

A protected configuration is secured with PIN protection. The protected configuration can be recalled by any user, but can only be written or overwritten using the assigned 4-digit PIN. Utilities for Save, Recall, and Change PIN, (separate from preset save), are accessed from the **Tools** menu as three sub-menus under a protected configuration menu item.

Protected configuration menu items are only available in live mode from the **Tools > Protected Configuration** menu. These functions can only be performed in **Live** mode:

- **Save**
- **Recall**
- **Change Password**

Save Protected Configuration

The default PIN is **0000**. The user can enter the default PIN or use the Change PIN (see below) dialog box to create a new one.

Recall Protected Configuration

The dialog box prompts the user to continue. Click **OK** to continue or **Cancel** to cancel the operation.

Change PIN

The change PIN utility allows the user to change a current protected configuration PIN. The current PIN must be entered before changes are allowed.

DSP Configurator Windows Menus

The DSP Configurator program is fully navigable using the computer keyboard. Some keyboard navigation behavior matches Windows standards, while other behaviors are specific to DSP Configurator.

Keyboard Navigation

When the program starts, the cursor focus defaults to the mic/line input gain block (figure 56, ①). The input 1 gain block is highlighted green [④]. The <Tab> key toggles to the various sections outlined in red in figure 56.

Within the sections, the <navigation arrows> can be used to move one block right, left, up, or down within these sections.

NOTE: The callout numbers for figure 56 are not the same as figure 17.

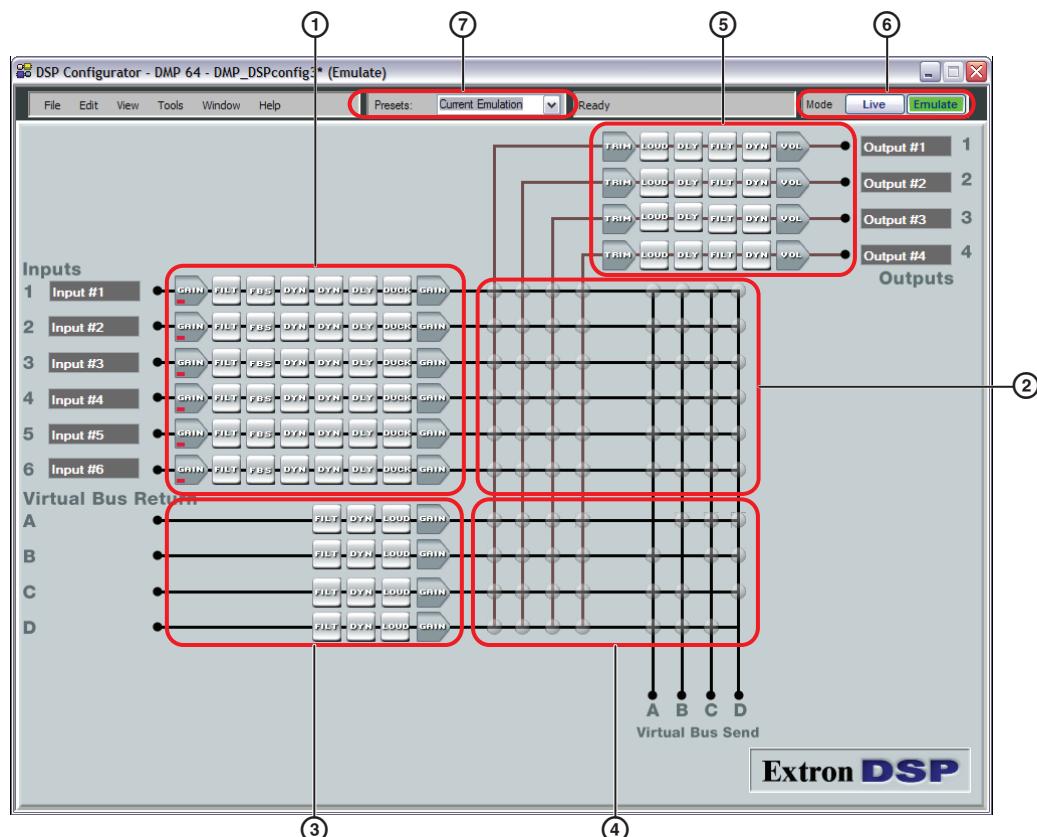


Figure 56. DSP Configurator Program Window

Standard Windows Navigation

The keyboard keys navigate and function as follows:

- **Tab key —**

Sequentially jump among major segments of the DSP Configurator program. From the audio input chains (①), sequential jumps are in the following order:

- ② Main mix-points
- ③ Virtual return signal path
- ④ Virtual return mix-points
- ⑤ Output signal chain
- ⑥ Mode (toggles to **Live**, then **Emulate**)
- ⑦ Presets (<Down> arrow can select presets)

- **Shift+Tab key combination** — Reverses the direction of the <Tab> key function.
- **Arrow (↑, ↓, ←, and →) Keys** — Navigate up, down, left, and right within any of the areas outlined in figure 56.
- **Enter Key** — Performs the same action as a mouse double-click. For example, it can open the context menu from which a processor type can be selected, or open a dialog box when applicable. When an action button is highlighted, <Enter> executes the button action and toggles the button when applicable.
- **Control key** — The <Ctrl> key can be used in the following shortcuts.
 - <Ctrl+X> — Cut the selected elements.
 - <Ctrl+C> — Copy the selected elements.
 - <Ctrl+V> — Paste the selected elements from a previous cut or copy.
 - <Ctrl+A> — The first press of the <Ctrl+A> combination highlights all matrix block nodes.
- **Alt key** — <Alt> is used with specific letter keys to open and navigate task bar menus. When the <Alt> key is pressed and released, the **File** menu opens. When the <Alt> key is pressed and held, the first letters in the menu titles (**File**, **Edit**, **View**, **Tools**, **Window**, or **Help**) become underlined. Press the underlined letter key to open that menu.
- Once a task bar menu opens, use the <Up> and <Down> arrow keys to move up and down in the menu or submenu, use the <Right> arrow key to open a submenu (if applicable), and use the <Esc> key to back out of an active menu or submenu.

DSP Configurator-unique Navigation

Highlighting and marking items, cutting or copying, saving a preset:

When an item within the program is selected, it is highlighted by a green boundary box. One or more highlighted items can be cut, copied, pasted, or saved as a preset. The cut, copy, and paste functions can be performed using the task bar menus (see the <Alt> key, above) or the shortcuts described on the previous page.

NOTE: When an item is cut, it is not removed from its original location until it has been pasted in its new location.

Highlight multiple elements for cut, copy, paste, or a preset as follows:

1. Use the <Arrow> (↑, ↓, ←, and →) keys to move to the first block to be highlighted.
2. To highlight a block:
 - a. **Press and hold** the <Shift> key, then use the <Arrow> keys to navigate away from the selected block.
 - b. To highlight additional sequential blocks, **continue to hold** the <Shift> key, then use the <Arrow> keys to navigate away from the selected block. Additional blocks will be highlighted as long as <Shift> is pressed. When the last element is highlighted, move the highlight box one additional block, then release the <Shift> key.
3. To move away from the highlighted block or set of sequential blocks, or to highlight non-sequential blocks:
 - a. After highlighting blocks in step 2, **press and hold** <Ctrl>, then use the <Arrow> keys to navigate to the next desired element. As long as the <Ctrl> key is held down, the block moved away from will not be highlighted. If the block is highlighted, it will be unhighlighted.
 - b. Release the <Ctrl> key, but do not press any <Arrow> keys.

4. To highlight another element or group of elements, repeat steps 2 and 3 as required.
 5. **To cut or copy**, press the <Ctrl+X> or <Ctrl+C> key combination.
 6. **To save a preset**, press <Alt +T>, <Right Arrow>, <Down Arrow>, then <Enter>, (see figure 57 below).
 7. The **Save a Preset** dialog box appears.
- a. <Tab> to highlight the **Preset Number** field and type a specific number.
 - b. <Tab> to highlight the **Preset Name** field and type a name.

NOTE: Unless entering a specific number and name, DSP Configurator enters the next sequential unused preset number.

- c. <Tab> to highlight the **OK** button and press the <Enter> key.

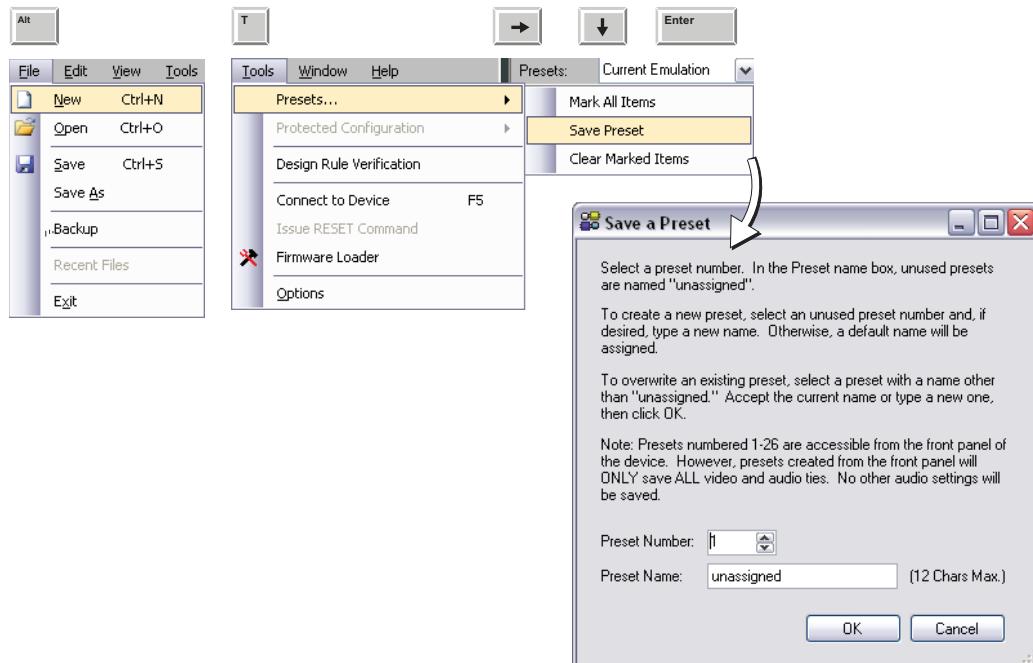


Figure 57. Saving a Preset Using Keyboard Navigation

Optimizing Audio Levels

The DMP 64 uses floating point DSP technology, processing data using a combination of 32- and 64-bit algorithms. The analog to digital converters (ADC) and digital to analog converters (DAC) sample at 48 kHz with 24-bit resolution.

With floating point DSP it is extremely difficult to clip the audio signal within the DSP audio signal chain, after the ADC input and before the DAC output. That means the audio signal must not be clipped at the input ADC. Clipping gives audibly undesirable results and once the audio is clipped at the input there is no way to correct it further down the signal chain. If audio clipping occurs at the output DAC that is not a result of clipping at the input ADC, there are ways to address it within the DSP audio signal chain.

The meters in the DSP Configurator indicate clipping at a user-definable point, with the default setting at -1 dB. This means the meter indicates clipping when it reaches -1 dBFS, or 1 dB below actual clipping (0 dBFS). Setting the clipping meter below clipping provides a “safety net”, allowing the user to reduce input gain before clipping actually occurs. This safety net can be increased or decreased by selecting **Tools>Options>Processor Defaults>Defaults>Meter Clipping**, and setting the **Clip Threshold** to a number between 0 (dB) and -20 (dB).

NOTE: When the **Clip Threshold** is set to 0 (dB), clipping is indicated only when clipping occurs.

Meters within the DSP Configurator are peak-type meters, referenced to full scale, or 0 dBFS. For the DMP 64 outputs, 0 dBFS corresponds to +21 dBu, the maximum output level of the device. Maximum input level is +24 dBu. Gain from -3 dB to +80 dB is applied in the analog domain, while attenuation from -3 dB to -18 dB is applied in the digital domain. The input meters are post-ADC, while the output meters are pre-DAC.

The remainder of this section will reference the gain, trim and volume controls outlined in figure 58 below.

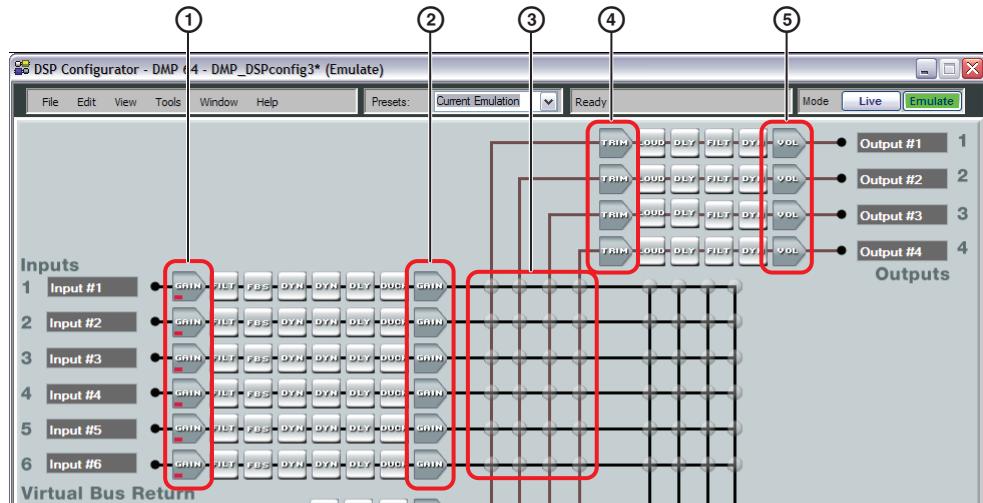


Figure 58. Gain, Trim and Volume Controls

- ① Mic/Line input gain
- ② Pre-mixer gain
- ③ Mix-point gain
- ④ Post-mixer trim
- ⑤ Output volume

About Setting Gain Structure

There are two approaches the system designer can take in setting up gain structure, depending upon where output volume is to be controlled. The output volume of the DMP 64 can be controlled by either of the following two gain blocks (see figure 58 on the previous page):

- Volume (⑤) and
- Pre-mixer gain (②).

NOTE: While the pre-mixer gain control is not in the output signal chain, it can be used to control program level independent of mix-point levels.

In the following instructions, setup is described for output volume or pre-mixer gain when appropriate.

Setting Input Gain

Floating point DSP technology is internally more flexible than fixed point. However, the input ADCs and output DACs always run as fixed point, so it is important to optimize the audio by setting the input level as close to 0 dBFS as possible. This maintains the resolution at 24-bit. Within the DSP it is not critical to maintain audio levels at 0 dBFS in order to secure the resolution at 24-bit.

Input gain can be set using the intended input source device and typical source material for the installation. When source material is not available, it can be set using pink noise obtained either from a pre-recorded track on a DVD or CD, or a signal generator.

For program material, set the input gain level (see figure 58, ①) so the meters reach –15 to –12 dBFS, with peaks at approximately –5 to –3 dBFS. This setting provides enough headroom to accommodate transients or unanticipated loud events in the program material to avoid clipping.

When using pink noise, it should be recorded at –20 dBFS. If the player has an output level setting control, set the output of the player to its maximum, or 0 dB of attenuation.

NOTE: If the maximum output setting provides gain, back the control down slightly from the maximum setting.

When using a signal generator, set the output at –10 dBu.

Whichever pink noise source is used, set the input gain in the DSP Configurator so the input meter reads –20 dBFS.

Setting a Nominal Output Level

In order to set up a gain structure to include signal processing, listening to the audio can be advantageous. Route the audio that carries program material from the source to the speakers in the room being set up. With the output volume control (see figure 58, ⑤) set to –20 dB, set the external amplifier so the source material plays at a volume level that is reasonably loud, but tolerable.

NOTE: When using the volume control for this purpose, set post-mixer trim (④) to 0 dB. If using the post-mixer trim for this purpose, set volume to 0 dB (100%).

Verify the amplifier is not clipping by observing the amplifier clip indicator. This sets the amplification (volume) nominal level of the system, and if desired, allows listening while making adjustments. Adjust or mute the volume control as necessary (see [Setting Volume Control for the Amplifier Stage](#) on page 91).

Adjusting Pre-mixer Gain

After setting input gain, add desired processors into the input signal chain (see figure 58 on page 88). The pre-mixer gain control (②) is used to compensate for level changes due to processing. Adding a compressor generally reduces the signal level, while a filter can boost or cut the overall signal level. When changes are made to filter settings after setting dynamics processors, re-check the levels in the dynamics processors to make certain they are still valid.

NOTE: This procedure is valid only when there is no active processing in the output signal path and if the post-matrix trim value is set to 0 dB, unity gain. If processors are inserted in the output signal path, engage **Bypass** to temporarily remove them.

To adjust pre-mixer gain:

1. Open the line input gain (①), output volume (③), and pre-mixer gain (②) dialog boxes.
2. Connect program material (or pink noise) at the input.
3. Set the output volume to 100% (mute if necessary).
4. Adjust the pre-mixer gain (②) so the meter level on the input gain dialog matches the meter level of the output volume dialog. This maintains the audio at an optimal level in the input signal chain.

This sets a good starting point. After setting up the microphone input gain and mix-point levels, output processing, and trim levels, if more headroom is required to prevent clipping at the outputs, return to the pre-mixer gain controls (②) and lower each one by specific amounts. Further minor adjustments to the pre-mixer gain controls help to balance perceived audio levels of the different inputs.

When using the pre-mixer gain for output volume control, the procedure can be reversed. Set pre-mixer gain to 0 dB. With program material (or pink noise) present at the input, adjust the output volume until the meter level in the output volume dialog box is below clipping (or ideally, matches the level at the input gain meter).

Setting Output Gain Structure

Add all desired processors into the output signal chain. Keep in mind that a filter can boost or cut the overall signal level and adding a compressor generally reduces the signal level. Inserting either or both can require resetting of the output volume.

Since a limiter is the most likely choice for output processing and can only reduce the signal to prevent overload, a reduction of output level does not have to be considered. Loudness boosts the overall signal level, but only at lower volume settings.

After adding processors to the output signal chain, the output volume level can clip when set to 100% (or less). Floating point DSP allows clipping to be overcome by lowering the output volume (③) setting. However, unless a user is prevented from changing the volume setting to 100% (or to any position where clipping occurs), it is best to adjust the pre-mixer gain (②) or post-mixer trim (④) control to prevent any possible clipping.

Alternately, use the post-mixer trim controls to adjust output volume. Post-mixer trim controls provide 12 dB of gain, so use a group master with soft limits to control levels, setting an upper limit of 0 dB or less. Mic levels also contribute to clipping at the outputs, and can need lowered to maintain the balance between program material (line outputs) and voice.

Setting Mic/Line Input and Mix Levels

In this example, the mic/line input 1 signal is sent to output 1 (see figure 58 on page 88).

To set the mic/line input and mix levels:

1. Connect a microphone to input #1.
2. Double-click the mix-point (⑧) for mic/line 1 to output 1 to open the dialog for that mix-point, then unmute the mix-point to place the signal into the mix. The default level for the mix-point is 0 dB, or unity gain.
3. Open the Input 1 **Gain** (①) dialog and set gain to 0 dB (turn on phantom power if the mic requires it), then unmute the channel.
4. While testing the mic, raise the fader level until the mic is clearly audible. The amount of gain and the meter level vary at this point, but as a general guideline the input gain level should be at 40 to 50 dB, with the meter averaging somewhere around -20 dBFS.

Ideally, audio will be optimized here, but voice levels at microphone inputs can vary greatly. Having the meters average around -20 dBFS allows enough headroom to accommodate sudden changes to voice levels. Further adjustment can be necessary.

Adjusting Trim

This is where setting gain structure becomes a balancing act. The following sections provide guidelines, but it can take a bit of going back and forth to correctly set levels for the installation. For example, output level can be controlled and kept below clipping using a compressor or limiter in the output dynamics block. However, adjusting the post-matrix trim will affect how the compressor or limiter works.

1. Apply program material (or pink noise) at the input to be adjusted.
2. Open the output **Volume** (⑤) and post-matrix **Trim** (④) dialog boxes.
3. Set output volume to 100% (mute if necessary).
4. Adjust the post-matrix trim until the meter level in the output volume dialog is below clipping (or ideally, matches the level at the input gain meter).

This maintains the audio at an optimal level in the output signal chain while preventing clipping at the output.

Setting Volume Control for the Amplifier Stage

The maximum output of the DMP 64 is +21 dBu. As an example, assume the maximum input level of a power amp is +4 dBu with its input attenuator fully open. If using the output **Volume** control (⑥) of the DMP 64 to control volume levels, turn down the input attenuator of the power amp the equivalent of 17 dB ($21 - 4 = 17$) to ensure clipping does not occur at the amplifier. That puts the amplifiers input level at -13 dB ($+4 - 17 = -13$). If the amplifier setting (when the output volume controls of the DMP 64 are at maximum) is too loud for the room, it can need to be reduced further. If it is not loud enough for the room, a more powerful amplifier can be required.

Extron recommends using the output volume or post-mixer trim control on the DMP 64 for controlling output volume. When using loudness processing on the unit, it only works in conjunction with these controls.

When using the power amplifier input attenuation to control volume (using the same power amp maximum input level), set the output volume or post-mixer trim control of the DMP 64 to -17 dB. This is another way that clip points of the two devices are matched. Verify the amplifier is not clipping by observing the amplifier clip indicator.

NOTE: Using the amplifier input attenuation to control volume compromises the signal-to-noise ratio of the DMP 64 and is not recommended.

SIS Programming and Control

This section describes SIS programming and control of the DMP 64, including:

- [Connection Options](#)
- [Host-to-device Communications](#)
- [Command and Response Table for Basic SIS Commands](#)
- [Command and Response Tables for DSP SIS Commands](#)
- [Special Characters](#)

Connection Options

The DMP 64 Digital Matrix Processor can be remotely connected via a host computer or other device (such as a control system) attached to the rear panel RS-232 port or LAN port, or the front panel USB Config port.

The DMP 64 can be set up and controlled using SIS commands, embedded web pages, or DSP Configurator software. See [Installation](#) on page 5 for pin assignments and details on the configuration and control port connections. For information on DSP Configurator, see [DMP Software](#) on page 14, and for the embedded web pages, see [HTML Operation](#) on page 120.

SIS commands can be executed using the Extron Electronics DataViewer program, found on the Software Products DVD included with the product.

DMP 64 RS-232 protocol:

- 38400 baud
- no parity
- 1 stop bit
- 8 data bits
- no flow control

NOTE: Both rear panel configuration ports require 38400 baud communication. This is a higher speed than many other Extron products use. When using HyperTerminal or a similar application, make sure the PC or control system connected to these ports is set for 38400 baud.

For additional details on connecting the RS-232 port, see [RS-232 Ports](#) on page 93.

USB port details:

The Extron USB driver must be installed before the USB port can be used (see [Install the USB Driver](#) on page 17).

LAN port defaults:

DMP 64 IP address: 192.168.254.254

gateway IP address: 0.0.0.0

subnet mask: 255.255.0.0

DHCP: off

RS-232 Ports

The DMP 64 has two serial ports that can be connected to a host device such as a computer running the HyperTerminal utility or DataViewer. The ports make serial control of the switcher possible. Use the protocol information listed to make the connection. For SIS programming details once the connection is made, see [Host-to-device Communications](#) on page 95.

USB Port (Front Panel)

The DMP 64 has a front panel USB port that can be connected to a host device such as a computer running the HyperTerminal utility, or the DataViewer utility. The port makes serial control of the switcher possible. For SIS programming details once the connection is made, see [Host-to-device Communications](#) on page 95.

Ethernet (LAN) Port

The rear panel LAN connector on the device can be connected to an Ethernet LAN or WAN. Communication between the device and the controlling device is via Telnet (a TCP socket using port 23). The Telnet port can be changed, if necessary, via SIS. This connection makes SIS control of the device possible using a computer connected to the same LAN or WAN. The SIS commands and behavior of the product are identical to the commands and behavior the product exhibits when communicating via a serial port or USB.

Ethernet Connection

The Ethernet cable can be terminated as a straight-through cable or a crossover cable and must be properly terminated for your application (see figure 59).

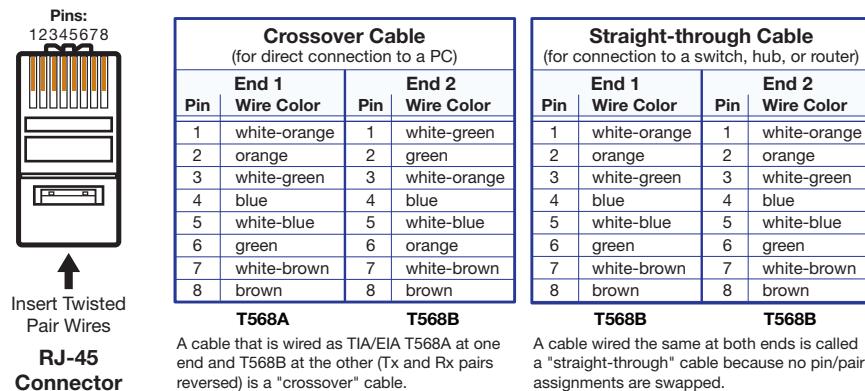


Figure 59. RJ-45 Ethernet Connector Pin Assignments

To establish a network connection to the DMP 64:

1. Open a TCP socket to port 23 using the mixer IP address.

NOTE: If the local system administrators have not changed the value, the factory-specified default, 192.168.254.254, is the correct value for this field.
2. The DMP 64 responds with a copyright message including the date, the name of the product, firmware version, part number, and the current date and time.
 - a. If the DMP 64 is not password-protected, the device is ready to accept SIS commands immediately after it sends the copyright message.
 - b. If the DMP 64 is password-protected, a **password** prompt appears below the copyright message. Proceed to step 3.
3. If the device is password protected, enter the appropriate administrator or user password.
 - a. If the password is accepted, the device responds with **Login User** or **Login Administrator**.
 - b. If the password is not accepted, the **Password** prompt reappears. Repeat step 3.

Connection Timeouts

The Ethernet link times out after a designated period of time of no communications. By default, this timeout value is set to 5 minutes but the value can be changed. See the **Configure Port Timeout** commands in the “[Command and Response Table for Basic SIS Commands](#)” beginning on page 98.

NOTE: Extron recommends leaving the default timeout at 5 minutes and periodically issuing the **Query (Q)** command to keep the connection active. If there are long idle periods, disconnect the socket and reopen the connection when another command must be sent.

Verbose Mode

Telnet connections can be used to monitor for changes that occur, such as SIS commands from other Telnet sockets or a serial port. For a Telnet session to receive change notices, the Telnet session must be in verbose mode 1 or 3. In verbose mode 1 or 3, the Telnet socket reports changes in messages that resemble SIS command responses.

Host-to-device Communications

The commands listed in the following tables perform the same functions, but are encoded differently to accommodate the requirements of each port (Telnet or browser).

DMP 64-initiated Messages

The DMP 64 initiates messages under specific conditions. No response is required from the host. The DMP 64-initiated messages are listed here (underlined).

© Copyright 2009, Extron Electronics, DMP 64, Vn.nn, 60-1054-01
Day, DD MMM YYYY HH:MM:SS

Vn.nn is the firmware version number.

The DMP 64 sends the boot and copyright messages under the following circumstances:

- If the DMP 64 is off and an RS-232 connection is set up (the PC is cabled to the DMP 64 and a serial communication program such as HyperTerminal is open), the connected unit sends these messages by RS-232 when first powered on.
- If the DMP 64 is on, it sends the boot and copyright messages when a Telnet connection to the DMP 64 is first opened. The day of the week, date, and time are shown when the DMP 64 is connected by Telnet, but not by RS-232. If using a Telnet connection, the copyright message, date, and time are followed by a password prompt.

Password Information

The **←Password:** prompt requires a password (administrator level or user level) followed by a carriage return. The prompt is repeated if the correct password is not entered.

If the correct password is entered, the unit responds with **←Login Administrator ←** or **←Login User←**, depending on the password entered. If passwords are the same for both administrator and user, the unit will default to administrator privileges.

SIS commands consist of a string (one or more characters per command field). No special characters are required to begin or end a command sequence. When the DMP 64 determines a command is valid, it executes the command and sends a response to the host device. All responses end with a carriage return and a line feed (CR/LF = **←**), signaling the end of the response character string.

When programming, certain characters are more conveniently represented by their hexadecimal rather than ASCII values. The table below shows the hexadecimal equivalent of each ASCII character:

ASCII to HEX Conversion Table							
Space	20	!	21	"	22	#	23
(28)	29	*	2A	+	2B
0	30	1	31	2	32	3	33
8	38	9	39	:	3A	;	3B
@	40	A	41	B	42	C	43
H	48	I	49	J	4A	K	4B
P	50	Q	51	R	52	S	53
X	58	Y	59	Z	5A	[5B
\`	60	a	61	b	62	c	63
h	68	i	69	j	6A	k	6B
p	70	q	71	r	72	s	73
x	78	y	79	z	7A	{	7B
							7C
						}	7D
						~	7E
							DEL
							7F

Figure 60. ASCII to Hex Conversion Table

The Command and Response tables list valid ASCII (for Telnet or RS-232) command codes, the corresponding URL (Uniform Resource Locator) encoded (for Web browsers) command codes, the DMP 64 responses to the host, and a description of the command function or the results of executing the command.

Symbol definitions

←	=	CR/LF (carriage return/line feed) (hex 0D 0A)
←	=	Carriage return (no line feed, hex 0D) (for URL-encoded commands, use the pipe character, , instead)
•	=	Space character (%20 for web browser)
	=	Pipe (vertical bar) character
*	=	Asterisk character (which is a command character, not a variable)
Esc	=	Escape key (hex 1B) (use W instead of Esc for Web browsers)

NOTE: For Web encoding only: data is directed to the specified port and must be encoded (URL encoding) if it is non-alphanumeric. Change any non-alphanumeric character (% , +, |, ←, and so on) within the data section into the corresponding hexadecimal equivalent, %xx, where xx represents the two-character hex byte. For example, a space (hex: 20) would be encoded as %20 and a plus sign (hex: 2B) would be encoded as %2B.

Error Responses

When the DMP 64 is unable to execute the command, it returns an error response to the host. The error response codes and their descriptions are as follows:

E01 - Invalid input number (number is too large)	E23 - Checksum error (for file uploads)
E12 - Invalid port number	E24 - Privilege violation
E13 - Invalid parameter (number is out of range)	E25 - Device is not present
E14 - Not valid for this configuration	E26 - Maximum connections exceeded
E17 - System timed out	E27 - Invalid event number
E22 - Busy	E28 - Bad filename or file not found

Simple Control Port Commands - Telnet and Web-browser Accessible

Upper and lower case text can be used interchangeably except where noted. Port 23 is default for Telnet. Port 80 is default for Web browser. They both can be mapped to different ports.

The following commands are for either a Telnet (port 23) or web browser (port 80) connection. There are minor differences when implementing these commands via Telnet or via URL encoding using a Web browser. All commands listed will work using either connection method but due to some limitations of the Web browser, the encapsulation characters must be modified to be certain the Web browser will properly handle them. All examples are shown in a proper implementation of a Telnet or web browser session.

NOTE: When using Web browsers, some non-alpha numeric characters must be represented as their hex equivalent such as %xx where xx equal the two character representation of the hex byte that needs to be sent (for example, a comma ',' would be represented as %2C). Characters such as '%' (percent), '+' (plus) and ' ' (space) should also be encoded in Hex.

Telnet

Escape (Hex 1B)

Carriage Return (Hex 0D)

Web Browser

W [must not be encoded]

Pipe Character (>) [must not be encoded]

When describing the use of SIS commands via a web browser, the [URL] reference is used to shorten the examples. [URL] would be the full URL of the control interface and web page reference including all path information (for example, <http://192.168.254.254/mypage.HTML>).

To send commands using a web browser, prefix them with the full URL followed by ?cmd= (for example, <http://192.168.254.254/mypage.html?cmd=WSF>>).

	ASCII	Hex	Unit response
Control Command (via Telnet)	Esc X3 X2 Command ← X2 Data	1B X3 X2 Command 0D X2 Data	response from command ←
Example:	Esc 03 RS ← 1*2!	1B 30 33 52 53 0D 31 2A 32 21	OUT 02•IN 01• ALL ←

	ASCII	URL Encoded (Web)	Unit response
Control Command (via WEB)	URL?cmd=W X3 X2 Command > X2 Data	URL?cmd=W X3 X2 Command > X3 X2 Data	response from command ←
Example:	URL?cmd=W 03 RS>1*2!	URL?cmd=W 03 RS > 1%2A%21	OUT 02•IN 01• ALL ←

NOTES: **X2** = Input number, 1 through 6
X3 = Output number 1 through 4

Although the DMP 64 uses the same structure for SIS commands, there are two variations. One is the global command structure noted above and documented in the [Command and Response Table for Basic SIS Commands](#) on page 98.

The second set of tables uses the command structure outline beginning with [Command and Response Tables for DSP SIS Commands](#) on page 103. While using the same structure of basic SIS commands, they differ in how the software addresses the individual processor blocks within the DMP 64.

Generally, the basic SIS commands are used for global configuration such as setting IP addresses, date/time, while the DSP SIS commands allow functionality of the audio signal chain.

Command and Response Table for Basic SIS Commands

Command	ASCII command (host to device)	URL Encoded (web)	Response (device to host)
Information requests			
Firmware Version	Q	*Q	X11←
Firmware and build version	*Q	*Q	X11←
Kernel firmware and build	**Q	**Q	X11←
Verbose version info	ØQ	ØQ	Sum of 2Q-3Q-4Q←
Firmware version	1Q	1Q	X11←
Bootstrap Version	2Q	2Q	X11←
Factory Firmware Version	3Q	3Q	X11 plus web ver.-desc-UL date/time←
Updated firmware version	4Q	4Q	X11 plus web ver.-desc-UL date/time←
NOTE: An asterisk (*) after the version number indicates the currently running version. Question marks (?.??) indicate that only factory firmware is loaded. A caret (^) indicates the firmware version that should be running, but a Mode 1 reset was executed and the default factory firmware is running. An exclamation point (!) indicates corrupted firmware.			
Query part number	N	N	60-1054-01←
Query model name	I	I	V00x00·A06x04←
Query model name	1I	1I	DMP•64←
Query model description	2I	2I	Digital•Matrix•Processor←
Query system memory usage	3I	3I	#Bytes used out of #KBytes←
Query user-memory usage	4I	4I	#Bytes used out of #KBytes←

NOTE: X11 = Version number
Firmware version number to second decimal place (x.xx)
Version and Build number adds four digits (x.xx.xxxx)
to the Version number

Command and Response table for basic SIS commands (continued)

Command	ASCII command (host to device)	Response (device to host)	Additional description
IP Setup Commands			
Set unit name	[Esc]X12 CN←	Ipn• X12 ←	
View unit name	[Esc] CN←	X12 ←	
Set name to factory default	[Esc]• CN←	Ipn• X49 ←	
Set time and date	[Esc]X13 CT←	Ipt• X13 ←	
View time and date	[Esc] CT←	X13 ←	
Set GMT offset	[Esc]X3 CZ←	Ipz X3 ←	
View GMT offset	[Esc] CZ←	X3 ←	
Set Daylight Saving Time	[Esc]X34 CX←	Ipx X34 ←	
Read Daylight Saving Time	[Esc] CX←	X34 ←	
Set IP address	[Esc]X14 CI←	Ipi X14 ←	
Read IP address	[Esc] CI←	X14 ←	
Read hardware address (MAC)	[Esc] CH←	X18 ←	
Set subnet mask	[Esc]X19 CS←	Ips X19 ←	
Read subnet mask	[Esc] CS←	X19 ←	
Set gateway IP address	[Esc]X14 CG←	Ipg X14 ←	
View gateway IP address	[Esc] CG←	X14 ←	
Set DHCP on	[Esc]1DH ←	Idh1←	
Set DHCP off	[Esc]ØDH ←	IdhØ←	
NOTE: Changing DHCP from On to Off resets the IP address to the factory default (192.168.254.254)			
View DHCP status	[Esc] DH←	X5 ←	
Set verbose mode	[Esc]X22 CV←	Vrb X22 ←	
View verbose mode	[Esc] CV←	X22 ←	
Get connection listing	[Esc] CC←	[number of connections]←	
NOTES: X3 = Greenwich Mean Time offset X5 = On/Off status X12 = Unit name X13 = Local date/time X14 = IP Address X18 = Hardware MAC address X19 = Subnet mask X22 = Verbose/Response mode X34 = Daylight Saving time X49 = Alpha-numeric unit name			
<p>X3 = Greenwich Mean Time offset X5 = On/Off status X12 = Unit name X13 = Local date/time X14 = IP Address X18 = Hardware MAC address X19 = Subnet mask X22 = Verbose/Response mode X34 = Daylight Saving time X49 = Alpha-numeric unit name</p> <p>GMT offset value (-12:00 to 14:00) representing hours and minutes (HH:MM) local time is offset from GMT time Ø=off/disable 1=on/enable Alpha-numeric up to 24 characters. No special characters except hyphen (-) No upper/lower case distinction, no blanks or spaces, first character must be alpha, last character cannot be hyphen. Set: MM/DD/YY-HH:MM:SS Read: day of week, date, month, year HH:MM:SS (Example: Fri, 21 Jun 2002 10:54:00) default 192.168.254.254 Ø-Ø -A6 -xx -xx -xx Default 255.255.Ø .Ø Ø=clear 1=verbose 2=tagged responses, 3=verbose + tagged responses Ø=off/ignore ; 1= USA (begins first Sunday in April/ends last Sunday in October); 2= Europe (begins last Sunday in March/ends last Sunday in October); 3= Brazil (begins third Sunday in October/ends third Saturday in March). combination of unit name and last three pairs of MAC address </p>			

Command and Response table for basic SIS commands (continued)

Command	ASCII command (host to device)	Response (device to host)	Additional description
Password and Security Settings			
Set administrator password	[Esc][X33]CA←	Ipa•[X41]←	
View administrator password	[Esc]CA←	[X41]←	
Reset (clear) administrator password	[Esc]•CA←	Ipa•←	
Set user password	[Esc][X33]CU←	Ipu•[X41]←	
View user password	[Esc]CU←	[X41]←	
Reset (clear) user password	[Esc]•CU←	Ipu•←	
Query session security level	[Esc]CK←	[X52]←	
Ethernet Data Port			
Set current port timeout	[Esc]0*[X69]TC←	Pti0*[X69]←	
View current port timeout	[Esc]0TC←	[X69]←	
Set global IP port timeout	[Esc]1*[X69]TC←	Pti1*[X69]←	
View global IP port timeout	[Esc]1TC←	[X69]←	
File Commands			
Erase user-supplied web page file	[Esc] filename EF ←	Del•filename ←	
Erase current directory	[Esc]/EF ←	Ddl←	Also deletes files inside directory
Erase current directory and sub-directories	[Esc]//EF ←	Ddl←	filename x•date/time•length
List files from current directory	[Esc]DF ←	filename x•date/time•length filename x•date/time•length filename x•date/time•length ... space_remaining•Bytes Left←	
List files from current directory and below	[Esc]LF ←	filename x•date/time•length filename x•date/time•length filename x•date/time•length filename x•date/time•length ... space_remaining•Bytes Left←	
NOTE: LF has the same response from unit as DF command, except the directory path will precede filenames for files from directories below the current directory.			
NOTES: [X33] = 12 alpha-numeric characters [X41] = alpha-numeric password returns four **** to mask password [X52] = Security level of connection 0=anonymous, 11=user, 12=administrator [X69] = IP connection timeout 1 - 65000 steps, (1 step=10 seconds)			

Command and Response table for basic SIS commands (continued)

Command	ASCII command (host to device)	Response (device to host)	Additional description
Serial Port			
Send Data String	[Esc][X1][X17][X20]*[X21]RS ← [X2]	response ←	
Configure parameters	[Esc][X1][X25][X26][X27][X28]CP ←	Cpn[X1]•Ccp[X25][X26][X27][X28] ←	
View serial port parameters	[Esc][X1]CP ←	[X25][X26][X27][X28] ←	
Configure rcv timeout	[Esc][X1][X17][X20]*[X23]*[X21]CE ←	Cpn[X1]•Cce[X17][X20][X23][X21] ←	
View receive timeout	[Esc][X1]CE ←	[X17][X20][X23][X21] ←	

NOTES:	[X1] = Port Number	[0]1-99 represented by 2 Bytes (ASCII).
	[X2] = Command data section	
	[X17] = Command string wait time	0-32767 in tens of milliseconds
	[X20] = Character wait time	0-32767 in tens of milliseconds
	[X21] = Length of stream or delimiter	L=Byte Count (00 – 32767) D=decimal value for ASCII character (0 - 00255)
	[X23] = Priority status for receiving timeouts (Default=0)	0=Send data string command parameters if they exist 1=Configure receive timeout command parameters instead.
	[X25] = Baud Rate (Default=9600)	300, 600, 1200, 1800, 2400, 3600, 4800, 7200, 9600, 14400, 19200, 38400, 57600, 115200
	[X26] = Parity (Default=N=none)	O=odd E=even N=none M=mark S=Space
	[X27] = Data bits (Default=8)	7, 8
	[X28] = Stop bits (Default=1)	1,2

Command	ASCII command (host to device)	Response (device to host)	Additional description
Event Control			
Read event buffer memory	[Esc][X35][X36][X37][X38]E ←	[X54] ←	
Write event buffer memory	[Esc][X35][X36][X39][X38]E ←	Evt[X35][X36][X37][X39] ←	
NOTE:	Response to Write Event is padded with leading zeros for [X35] & [X37].		
Read string from event buffer	[Esc][X35][X36][X37][X44]FE	{string} ←	
Write string to event buffer	[Esc]{string}*[X35][X36][X37]FE		
NOTE:	'F' must be capitalized to read and write strings to event buffer memory. Response to Write Event is padded with leading zeros for [X35] & [X37].		
Start events	[Esc]1AE ←	Ego ←	
Stop events	[Esc]0AE ←	Est ←	
Query # of running events	[Esc]AE ←	##### ← (5 digit number)	

NOTES:	[X35] = Event number	range 00-99
	[X36] = Event buffer	0=receive 1=Unified 2=data 3=NVRAM
	[X37] = Event buffer offset	range: 0 to Max buffer size
	[X38] = Event data size (case sensitive)	b=bit, B=Byte (8-bit), S=short (16-bit), L=long ((32-bit))
	[X39] = Event data to write	
	[X44] = number of Bytes to read	range 1-127
	[X54] = Data element read	ASCII digit(s) representing numeric value of data elements read from buffer (leading zeros suppressed)

Command and Response table for basic SIS commands (continued)

Command	ASCII command (host to device)	Response (device to host)	Additional description
Presets, I/O Names			
Write preset name	[Esc][X10][X11]NG ←	Nmg[X10][X11]←	
Example:	[Esc]1,Security1NG ←	Nmg01,Security1←	Name preset 1 "Security 1"
Read preset name	[Esc][X10]NG ←	[X11]←	
Example:	[Esc]2NG ←	Security2←	
Recall a preset	[X10].	Rpr[X10]←	Command character is a period.
Example	5.	Rpr[X10]←	Recall preset 5, which becomes the current configuration.
Write input name	[Esc][X3][X11]NI ←	Nmi[X3][X11]←	
Example:	[Esc]9,Podium cam1NI ←	Nmi09,Podium cam1←	Name input 9 "Podium cam1"
Read input name	[Esc][X3]NI ←	[X11]←	
Write output name	[Esc][X2][X11]NO ←	Nmo[X2][X11]←	
Example:	[Esc]1,Main PJ1NO ←	Nmo01,Main PJ1←	Name output 1 "Main PJ1"
Read output name	[Esc][X2]NO ←	[X11]←	
Resets			
Reset presets and names	[Esc]ZG ←	Zpg←	Clear all presets and their names.
Reset an individual preset	[Esc][X10]ZG ←	Zpg[X10]←	Clear preset [X10].
Reset a group	[Esc][X20]GRPM ←	Grpm[X20]←	Delete all members from group [X20], reset parameters and soft limits.
NOTE: See Group Masters on page 69, for more information about audio group masters.			
Reset flash	[Esc]ZFFF ←	Zpf←	Reset flash memory (erase user-supplied files).
System Reset (factory defaults)	[Esc]ZXXX ←	Zpx←	Resets all processors, level controls and mixers to default.
Reset all device settings and delete files	[Esc]ZY ←	Zpy←	
NOTE: This reset excludes IP settings such as IP address, subnet mask, gateway IP address, unit name, DHCP setting and port mapping (telnet/web/direct access) in order to preserve communication with the device. This reset is recommended after a firmware update.			
Absolute reset	[Esc]ZQQQ ←	Zpq←	Similar to System Reset , plus sets the IP address to 192.168.254.254 and the subnet mask to 255.255.0.0.
NOTES: X3 = Input number X2 = Output number X10 = Preset # X11 = Name X20 = Group master group number			
01 – 06 01 – 04 32 maximum (0 = current configuration) 12 characters maximum 01 – 32			

Command and Response Tables for DSP SIS Commands

Many digital signal processor (DSP) functions; gain, mute, group masters, and a protected configuration can be controlled using SIS commands. These commands follow the same general rules as basic SIS commands, but the variables (Xn) tend to be more complex. Also, a comprehensive understanding of the audio signal flow is helpful to understanding the commands. Figure 61 shows specific DSP functions available using SIS commands.

NOTE: The entire signal flow is described in more detail in the DSP Configurator program section (see [Windows-based Program Control](#) on page 15).

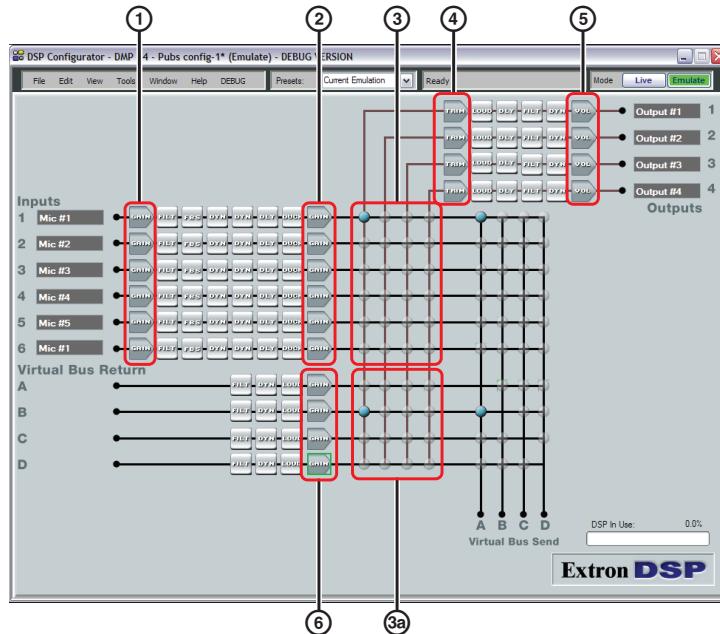


Figure 61. DSP Processors Addressable by SIS Commands

- ① Mic/line input gain block (including gain and mute)
- ② Pre-mixer gain block (including gain and mute)
- ③ Mix-points (including gain and mute)
- ④ Virtual return mix-points
- ⑤ Post-mixer trim block (gain only)
- ⑥ Output volume (including gain and mute)
- ⑦ Virtual return gain

Symbol definitions

←	= CR/LF (carriage return/line feed) (hex 0D 0A)	
←	= Carriage return (no line feed, hex 0D) (use the pipe character, , for Web browser commands)	
•	= Space character	
	= Pipe (vertical bar) character	
Esc	= Escape key (hex 1B) (use W instead of Esc for Web browsers)	
X60	= Gain and trim control or mix-point select	See the tables on page 107.
X61	= Level value; mix-point gain (③), and post-mixer trim (④)	See the table on pages 109, 110, and 111. -35 dB to + 25 dB, (1698 to 2298) in 0.1 dB increments.
		NOTE: Post-mixer, -12 dB to +12 dB (1928 to 2168) only.
X62	= Mic/line gain (①) level value	See the table on pages 112, 114, and 115. -18.0 dB to +80 dB, (1868 to 2848) in 0.1 dB increments.
X63	= Level value; pre-mixer gain (②), and output volume (⑤)	See the table on pages 116, 117, 118, and 119. -100.0 dB to +12.0 dB, (1048 to 2168) in 0.1 dB increments.
		NOTE: Output volume, -100.0 dB to +0.0 dB (1048 to 2048) only.
X64	= Mute status	0 = unmute 1 = mute
X65	= Group master group number	01 – 32
X66	= Group fader setting	dB value, in 0.1 dB increments using negative numbers but not decimal places. The valid range depends on the type of gain block that is assigned to the group number (X65) specified in the command: ① = -180 to 800 (-18.0 dB to +80.0 dB) ② = -1000 to 120 (-100.0 dB to +12.0 dB) ③ = -350 to 250 (-35.0 dB to + 25.0 dB) ④ = -120 to 120 (-12.0 dB to +12.0 dB) ⑤ = -1000 to 000 (-100.0 dB to +0.0 dB)
		NOTE: Leading zeros are ignored.
X67	= Group fader increment	dB value, in 0.1 dB increments, to raise or lower a group fader
X68	= Group fader soft limit	dB value, in 0.1 dB increments. The valid range X66 must be within the range for the gain block grouped in X65 .
X69	= Group type	6 = gain 12 = mute
X70	= Personal Identification Number (PIN)	Four numeric digits, default = 0000
X71	= Protected configuration status	0 = no protected configuration saved 1 = protected configuration saved

Special Characters

The HTML language reserves certain characters for specific functions. The device will not accept these characters as part of preset names, the device name, passwords, or locally created file names.

The DMP 64 rejects the following characters:

{space (spaces **are** OK for names)} + } ~ , @ = ' [] { } < > ' " ; : > \ ?

Command and Response Table for DSP SIS Commands

Command	ASCII command (host to device)	Response (device to host)	Additional description
Audio Level Control, and Mix-point Selection			
<p>NOTE: The command format is the same, regardless of the control or mix-point to be set; the acceptable adjustment range varies depending on the control or mix-point:</p> <ul style="list-style-type: none"> • The mic/line input gain range is –18 dB to +80 dB, in 0.1 dB increments. • The pre-mixer gain range is –100 dB to +12 dB, in 0.1 dB increments. • The main mix-points range –35 dB to +25 dB, in 0.1 dB increments. • The post-mixer trim range is –12 dB to +12 dB, in 0.1 dB increments. • The output volume range is –100 dB to 0 dB, in 0.1 dB increments. <p>All responses are shown with the DMP 64 in Verbose mode 2 or 3.</p>			
Set a trim or gain (excluding mic/line inputs)	<code>EscG[X60]*[X61]AU←</code>	DsG[X60]*[X61]←	Set trim or mix control [X60] to a value of [X61] dB.
Example 1 (pre-mixer gain):	<code>EscG40105*2040AU←</code>	DsG40105*2040←	Set the #6 pre-mixer gain to a value of -0.8 dB.
Example 2 (mix-point gain):	<code>EscG20001*2213AU←</code>	DsG20001*2213←	Mix +16.5 dB of mic 1 into output 2.
Set a mic/line gain	<code>EscG[X60]*[X62]AU←</code>	DsG[X60]*[X62]←	Set mic/line gain control [X60] to a value of [X62] dB.
Example:	<code>EscG40001*2288AU←</code>	DsG40001*2288←	Set the mic/line input 2 gain to a level of +24.0 dB.
Set Output Volume	<code>EscG[X60]*[X63]AU←</code>	DsG[X60]*[X63]←	Set output volume control [X60] to a value of [X63] dB.
Example:	<code>EscG60000*1548AU←</code>	DsG60000*1548←	Output 1 volume set to a level of -50.0 dB.
Read a trim or mix (excluding mic/line inputs)	<code>EscG[X60]AU←</code>	DsG[X60]*[X61]←	DSP trim or mix control [X60] is set to a value of [X61] dB.
Example 1 (post mixer gain control):	<code>EscG60101AU←</code>	DsG60101*2103←	Output 2, post mixer trim is set to a value of +5.5 dB.
Example 2 (mix control):	<code>EscG20203AU←</code>	DsG20203*2140←	+9.2 dB of mic 3 is mixed into output 4.
Read a mic/line gain	<code>EscG[X60]AU←</code>	DsG[X60]*[X62]←	Mic/line gain control [X60] is set to a value of [X62] dB.
Example:	<code>EscG40000AU←</code>	DsG40000*2598←	Mic/line input 1 gain is set to a value of +55.0 dB.
Audio Mute			
<p>NOTES:</p> <ul style="list-style-type: none"> • The post-mixer trim cannot be muted. • All responses are shown with the mixer device in Verbose mode 2 or 3. 			
Audio mute	<code>EscM[X60]*1AU←</code>	DsM[X60]*1←	Mute audio point [X60].
Example:	<code>EscM20301*1AU←</code>	DsM20301*1←	Mute mix-point input 4 to output 2.
Audio unmute	<code>EscM[X60]*0AU←</code>	DsM[X60]*0←	Unmute audio point [X60].
Read audio mute or level	<code>EscM[X60]AU←</code>	DsM[X60]*[X64]←	

NOTES:	[X60] = Audio level control, or mix-point select	See table 1 on page 111 and 112.
	[X61] = Level value; mix-point or post-mixer trim	See tables 2, 3, and 4 on page 113 through 115.
	[X62] = Mic/line gain level value	See table 5 on page 116 through 119.
	[X63] = Level value; pre-mixer gain and output volume	See table 6 on page 120 through 123.
	[X64] = Mute status	0 = unmute 1 = mute

Command and Response Table for DSP SIS Commands (continued)

Command	ASCII command (host to device)	Response (device to host)	Additional description
Audio group master commands			
NOTE: <ul style="list-style-type: none"> See Group Masters on page 69, for more information about audio group masters. A group must have assigned members for these commands to have an effect. For X66, a positive (+) value is assumed unless a negative (-) value is specified. If entering a X66 value outside the valid range for the group or outside the soft limits, the DMP 64 responds with an “invalid parameter” (E13) error. X66, X67, and X68 values can be sent without leading zeroes; responses are always 5 digits. 			
Set a group fader control	[Esc]D[X65]*[X66]GRPM ←	Grpm[X65]*[X66] ←	Set the group fader to a value of X66 .
Example:	[Esc]d2*-293*GRPM ←	GrpmD02*-00293 ←	Set the group 2 fader control to -29.3 dB.
Raise a group fader control	[Esc]D[X65]*[X67]+GRPM ←	Grpm[X65]*[X66] ←	Increase the level of the X65 group fader by X67 dB.
Example	[Esc]d2*30+GRPM ←	GrpmD02*-00263 ←	Raise the group 2 fader 3 dB (from -29.3 dB to -26.3 dB, starting from the level set in the “Set a group fader control” example, above).
Lower a group fader control	[Esc]D[X65]*[X67]-GRPM ←	Grpm[X65]*[X66] ←	Decrease the level of the X65 group fader by X67 dB.
View the group fader control level	[Esc]D[X65]GRPM ←	Grpm[X65]*[X66] ←	In verbose modes 1 and 2, the response is simplified to X66 ←.
Mute a group mute control	[Esc]D[X65]*1GRPM ←	GrpmD[X65]*+00001 ←	Mute all blocks in group X65 .
Clear (unmute) a group mute control	[Esc]D[X65]*0GRPM ←	GrpmD[X65]*+00000 ←	Umute all blocks in group X65 .
View a group mute control	[Esc]D[X65]GRPM ←	GrpmD[X65]*[X64] ←	For group masters, X64 is always expressed as a positive or negative 5-digit value.
Set soft limits	[Esc]L[X65]*[X68]upper*[X68]lowerGRPM ←	GrpmL[X65]*[X68]*[X68] ←	Set the groups soft limits to X68 and X68 .
Example:	[Esc]L2*+60*-60GRPM ←	GrpmL02*+00060*-00060 ←	Set the upper soft limit for the group 2 fader to +6.0 dB and the lower limit to -6.0 dB.
View soft limits	[Esc]L[X65]GRPM ←	GrpmL[X65]*[X68]*[X68] ←	In verbose modes 0 and 1, the response is simplified to X68*X68 ←.
View group type	[Esc]P[X65]GRPM ←	GrpmP[X65]*[X69] ←	Show the group type (X69) for group X65 . In verbose modes 0 and 1, the response is simplified to X69 ←.
View group members	[Esc]0[X65]GRPM ←	Grpm0[X65]*[X60]¹*[X60]²* ...*[X60] ¹⁶ ←	X60 is the control or mix-point. In verbose modes 0 and 1, the response is simplified to [X60]¹*[X60]²*...*[X60]¹⁶ ←.

NOTES: **X60** = Audio level control, or mix-point select

See Table 1 on page 111 and 112.

X65 = Group master group number

01 - 32

X66 = Group fader level

dB value, in 0.1 dB increments, using negative numbers but not decimal places. -100.0 dB to +80.0 dB is represented by -1000 to 800. The valid range depends on the type of gain or trim block assigned to the group number (**X65**).

X67 = Group fader increase/decrease

dB value, in 0.1 dB increments, to raise or lower a group fader.

X68 = Group fader soft limit

dB value, in 0.1 dB increments. The valid range must be within the range for the gain block grouped in **X65**.

X69 = Group type

6 = gain
12 = mute

Command and Response Table for DSP SIS Commands (continued)

Command	ASCII command (host to device)	Response (device to host)	Additional description
Protected configuration			
<p>NOTE: The DMP 64 can save and recall a Personal Identification Number (PIN)-protected configuration, including mic mixes, parameters, variables, and values (with the exception of the device's IP address). The protected configuration is useful to establish the DMP 64 in a known state, either as a troubleshooting tool or as a baseline configuration.</p>			
Save the configuration	[Esc] S [X70] PCFG ←	PcfgS ←	Save the configuration to the protected memory location.
Recall the configuration	[Esc] RPCFG ←	PcfgR ←	Recall the protected configuration
Change the PIN	[Esc] P [X70]^{old} [X70] new PCFG ←	PcfgP [X70]^{new} ←	Overwrite the old PIN ([X70] ^{old}) with the new one ([X70] ^{new}).
Query configuration saved status	[Esc] QPCFG ←	[X71] ←	
NOTES: [X70] = Personal Identification Number (PIN) [X71] = Protected configuration status		Four numeric digits, default = 0000 0 = no protected configuration saved 1 = protected configuration saved	

Table 1. [X60] — Level Control and Mix-point Selection

NOTE: Circled numbers refer to **DSP Processors Addressable by SIS Commands** on page 103.

① Input Gain Control	[X60]
Mic/Line Input 1	40000
Mic/Line Input 2	40001
Mic/Line Input 3	40002
Mic/Line Input 4	40003
Mic/Line Input 5	40004
Mic/Line Input 6	40005

② Pre-mixer Gain	[X60]
Mic/Line Input 1	40100
Mic/Line Input 2	40101
Mic/Line Input 3	40102
Mic/Line Input 4	40103
Mic/Line Input 5	40104
Mic/Line Input 6	40105

⑤ Volume Out Control	[X60]
Output 1	60000
Output 2	60001
Output 3	60002
Output 4	60003

⑥ Virtual Return Gain	[X60]
Output A	50000
Output B	50001
Output C	50002
Output D	50003

Command and Response table for DSP SIS commands (continued)

③ Main Mix-Point	X60	③ Main Mix-Point	X60
Input 1 to Output 1	20000	Input 2 to Output 1	20100
Input 1 to Output 2	20001	Input 2 to Output 2	20101
Input 1 to Output 3	20002	Input 2 to Output 3	20102
Input 1 to Output 4	20003	Input 2 to Output 4	20103
Input 1 to VR A	20004	Input 2 to VR A	20104
Input 1 to VR B	20005	Input 2 to VR B	20105
Input 1 to VR C	20006	Input 2 to VR C	20106
Input 1 to VR D	20007	Input 2 to VR D	20107
Input 3 to Output 1	20200	Input 4 to Output 1	20300
Input 3 to Output 2	20201	Input 4 to Output 2	20301
Input 3 to Output 3	20202	Input 4 to Output 3	20302
Input 3 to Output 4	20203	Input 4 to Output 4	20303
Input 3 to VR A	20204	Input 4 to VR A	20304
Input 3 to VR B	20205	Input 4 to VR B	20305
Input 3 to VR C	20206	Input 4 to VR C	20306
Input 3 to VR D	20207	Input 4 to VR D	20307
Input 5 to Output 1	20400	Input 6 to Output 1	20500
Input 5 to Output 2	20401	Input 6 to Output 2	20501
Input 5 to Output 3	20402	Input 6 to Output 3	20502
Input 5 to Output 4	20403	Input 6 to Output 4	20503
Input 5 to VR A	20404	Input 6 to VR A	20504
Input 5 to VR B	20405	Input 6 to VR B	20505
Input 5 to VR C	20406	Input 6 to VR C	20506
Input 5 to VR D	20407	Input 6 to VR D	20507
③a Virtual Return Mix-Point	X60	③a Virtual Return Mix-Point	X60
VR A to Output 1	20600	VR B to Output 1	20700
VR A to Output 2	20601	VR B to Output 2	20701
VR A to Output 3	20602	VR B to Output 3	20702
VR A to Output 4	20603	VR B to Output 4	20703
VR A to VR A	NA	VR B to VR A	20704
VR A to VR B	20605	VR B to VR B	NA
VR A to VR C	20606	VR B to VR C	20706
VR A to VR D	20607	VR B to VR D	20707
VR C to Output 1	20800	VR D to Output 1	20900
VR C to Output 2	20801	VR D to Output 2	20901
VR C to Output 3	20802	VR D to Output 3	20902
VR C to Output 4	20803	VR D to Output 4	20903
VR C to VR A	20804	VR D to VR A	20904
VR C to VR B	20805	VR D to VR B	20905
VR C to VR C	NA	VR D to VR C	20906
VR C to VR D	20807	VR D to VR D	NA

X61 – **Mix-point gain (③), and Post-mixer trim (④) level values**

NOTE: The maximum range of the mix-point gain is -35.0 to +25.0 dB; post-mixer trim is -12.0 to +12.0 dB.

dB value	X61																		
-34.9	1699	-34.8	1700	-34.7	1701	-34.6	1702	-34.5	1703	-34.4	1704	-34.3	1705	-34.2	1706	-34.1	1707	-34.0	1708
-33.9	1709	-33.8	1710	-33.7	1711	-33.6	1712	-33.5	1713	-33.4	1714	-33.3	1715	-33.2	1716	-33.1	1717	-33.0	1718
-32.9	1719	-32.8	1720	-32.7	1721	-32.6	1722	-32.5	1723	-32.4	1724	-32.3	1725	-32.2	1726	-32.1	1727	-32.0	1728
-31.9	1729	-31.8	1730	-31.7	1731	-31.6	1732	-31.5	1733	-31.4	1734	-31.3	1735	-31.2	1736	-31.1	1737	-31.0	1738
-30.9	1739	-30.8	1740	-30.7	1741	-30.6	1742	-30.5	1743	-30.4	1744	-30.3	1745	-30.2	1746	-30.1	1747	-30.0	1748
-29.9	1749	-29.8	1750	-29.7	1751	-29.6	1752	-29.5	1753	-29.4	1754	-29.3	1755	-29.2	1756	-29.1	1757	-29.0	1758
-28.9	1759	-28.8	1760	-28.7	1761	-28.6	1762	-28.5	1763	-28.4	1764	-28.3	1765	-28.2	1766	-28.1	1767	-28.0	1768
-27.9	1769	-27.8	1770	-27.7	1771	-27.6	1772	-27.5	1773	-27.4	1774	-27.3	1775	-27.2	1776	-27.1	1777	-27.0	1778
-26.9	1779	-26.8	1780	-26.7	1781	-26.6	1782	-26.5	1783	-26.4	1784	-26.3	1785	-26.2	1786	-26.1	1787	-26.0	1788
-25.9	1789	-25.8	1790	-25.7	1791	-25.6	1792	-25.5	1793	-25.4	1794	-25.3	1795	-25.2	1796	-25.1	1797	-25.0	1798
-24.9	1799	-24.8	1800	-24.7	1801	-24.6	1802	-24.5	1803	-24.4	1804	-24.3	1805	-24.2	1806	-24.1	1807	-24.0	1808
-23.9	1809	-23.8	1810	-23.7	1811	-23.6	1812	-23.5	1813	-23.4	1814	-23.3	1815	-23.2	1816	-23.1	1817	-23.0	1818
-22.9	1819	-22.8	1820	-22.7	1821	-22.6	1822	-22.5	1823	-22.4	1824	-22.3	1825	-22.2	1826	-22.1	1827	-22.0	1828
-21.9	1829	-21.8	1830	-21.7	1831	-21.6	1832	-21.5	1833	-21.4	1834	-21.3	1835	-21.2	1836	-21.1	1837	-21.0	1838
-20.9	1839	-20.8	1840	-20.7	1841	-20.6	1842	-20.5	1843	-20.4	1844	-20.3	1845	-20.2	1846	-20.1	1847	-20.0	1848
-19.9	1849	-19.8	1850	-19.7	1851	-19.6	1852	-19.5	1853	-19.4	1854	-19.3	1855	-19.2	1856	-19.1	1857	-19.0	1858
-18.9	1859	-18.8	1860	-18.7	1861	-18.6	1862	-18.5	1863	-18.4	1864	-18.3	1865	-18.2	1866	-18.1	1867	-18.0	1868
-17.9	1869	-17.8	1870	-17.7	1871	-17.6	1872	-17.5	1873	-17.4	1874	-17.3	1875	-17.2	1876	-17.1	1877	-17.0	1878
-16.9	1879	-16.8	1880	-16.7	1881	-16.6	1882	-16.5	1883	-16.4	1884	-16.3	1885	-16.2	1886	-16.1	1887	-16.0	1888
-15.9	1889	-15.8	1890	-15.7	1891	-15.6	1892	-15.5	1893	-15.4	1894	-15.3	1895	-15.2	1896	-15.1	1897	-15.0	1898
-14.9	1899	-14.8	1900	-14.7	1901	-14.6	1902	-14.5	1903	-14.4	1904	-14.3	1905	-14.2	1906	-14.1	1907	-14.0	1908
-13.9	1909	-13.8	1910	-13.7	1911	-13.6	1912	-13.5	1913	-13.4	1914	-13.3	1915	-13.2	1916	-13.1	1917	-13.0	1918
-12.9	1919	-12.8	1920	-12.7	1921	-12.6	1922	-12.5	1923	-12.4	1924	-12.3	1925	-12.2	1926	-12.1	1927		

Table 2. Mix-point Gain Only

[X61] – Mix-point gain (③), and Post-mixer trim (④) level values, (continued)

dB value	[X61]																		
-11.9	1929	-11.8	1930	-11.7	1931	-11.6	1932	-11.5	1933	-11.4	1934	-11.3	1935	-11.2	1936	-11.1	1937	-11.0	1938
-10.9	1939	-10.8	1940	-10.7	1941	-10.6	1942	-10.5	1943	-10.4	1944	-10.3	1945	-10.2	1946	-10.1	1947	-10.0	1948
-9.9	1949	-9.8	1950	-9.7	1951	-9.6	1952	-9.5	1953	-9.4	1954	-9.3	1955	-9.2	1956	-9.1	1957	-9.0	1958
-8.9	1959	-8.8	1960	-8.7	1961	-8.6	1962	-8.5	1963	-8.4	1964	-8.3	1965	-8.2	1966	-8.1	1967	-8.0	1968
-7.9	1969	-7.8	1970	-7.7	1971	-7.6	1972	-7.5	1973	-7.4	1974	-7.3	1975	-7.2	1976	-7.1	1977	-7.0	1978
-6.9	1979	-6.8	1980	-6.7	1981	-6.6	1982	-6.5	1983	-6.4	1984	-6.3	1985	-6.2	1986	-6.1	1987	-6.0	1988
-5.9	1989	-5.8	1990	-5.7	1991	-5.6	1992	-5.5	1993	-5.4	1994	-5.3	1995	-5.2	1996	-5.1	1997	-5.0	1998
-4.9	1999	-4.8	2000	-4.7	2001	-4.6	2002	-4.5	2003	-4.4	2004	-4.3	2005	-4.2	2006	-4.1	2007	-4.0	2008
-3.9	2009	-3.8	2010	-3.7	2011	-3.6	2012	-3.5	2013	-3.4	2014	-3.3	2015	-3.2	2016	-3.1	2017	-3.0	2018
-2.9	2019	-2.8	2020	-2.7	2021	-2.6	2022	-2.5	2123	-2.4	2024	-2.3	2025	-2.2	2026	-2.1	2027	-2.0	2028
-1.9	2029	-1.8	2030	-1.7	2031	-1.6	2032	-1.5	2033	-1.4	2034	-1.3	2035	-1.2	2036	-1.1	2037	-1.0	2038
-0.9	2039	-0.8	2040	-0.7	2041	-0.6	2042	-0.5	2043	-0.4	2044	-0.3	2045	-0.2	2046	-0.1	2047	0.0	2048
+0.1	2049	+0.2	2050	+0.3	2051	+0.4	2052	+0.5	2053	+0.6	2054	+0.7	2055	+0.8	2056	+0.9	2057	+1.0	2058
+1.1	2059	+1.2	2060	+1.3	2061	+1.4	2062	+1.5	2063	+1.6	2064	+1.7	2065	+1.8	2066	+1.9	2067	+2.0	2068
+2.1	2069	+2.2	2070	+2.3	2071	+2.4	2072	+2.5	2073	+2.6	2074	+2.7	2075	+2.8	2076	+2.9	2077	+3.0	2078
+3.1	2079	+3.2	2080	+3.3	2081	+3.4	2082	+3.5	2083	+3.6	2084	+3.7	2085	+3.8	2086	+3.9	2087	+4.0	2088
+4.1	2089	+4.2	2090	+4.3	2091	+4.4	2092	+4.5	2093	+4.6	2094	+4.7	2095	+4.8	2096	+4.9	2097	+5.0	2098
+5.1	2199	+5.2	2100	+5.3	2101	+5.4	2102	+5.5	2103	+5.6	2104	+5.7	2105	+5.8	2106	+5.9	2107	+6.0	2108
+6.1	2109	+6.2	2110	+6.3	2111	+6.4	2112	+6.5	2113	+6.6	2114	+6.7	2115	+6.8	2116	+6.9	2117	+7.0	2118
+7.1	2119	+7.2	2120	+7.3	2121	+7.4	2122	+7.5	2123	+7.6	2124	+7.7	2125	+7.8	2126	+7.9	2127	+8.0	2128
+8.1	2129	+8.2	2130	+8.3	2131	+8.4	2132	+8.5	2133	+8.6	2134	+8.7	2135	+8.8	2136	+8.9	2137	+9.0	2138
+9.1	2139	+9.2	2140	+9.3	2141	+9.4	2142	+9.5	2143	+9.6	2144	+9.7	2145	+9.8	2146	+9.9	2147	+10.0	2148
+10.1	2149	+10.2	2150	+10.3	2151	+10.4	2152	+10.5	2153	+10.6	2154	+10.7	2155	+10.8	2156	+10.9	2157	+11.0	2158
+11.1	2159	+11.2	2160	+11.3	2161	+11.4	2162	+11.5	2163	+11.6	2164	+11.7	2165	+11.8	2166	+11.9	2167	+12.0	2168

Table 3. Post-mixer Trim and Mix-point Gain

[X61] – Mix-point gain (③), and Post-mixer trim (④) level values, (continued)

[X61] dB value															
+12.1	2169	+12.2	2170	+12.3	2171	+12.4	2172	+12.5	2173	+12.6	2174	+12.7	2175	+12.8	2176
+13.1	2179	+13.2	2180	+13.3	2181	+13.4	2182	+13.5	2183	+13.6	2184	+13.7	2185	+13.8	2186
+14.1	2189	+14.2	2190	+14.3	2191	+14.4	2192	+14.5	2193	+14.6	2194	+14.7	2195	+14.8	2196
+15.1	2199	+15.2	2200	+15.3	2201	+15.4	2202	+15.5	2203	+15.6	2204	+15.7	2205	+15.8	2206
+16.1	2209	+16.2	2210	+16.3	2211	+16.4	2212	+16.5	2213	+16.6	2214	+16.7	2215	+16.8	2216
+17.1	2219	+17.2	2220	+17.3	2221	+17.4	2222	+17.5	2223	+17.6	2224	+17.7	2225	+17.8	2226
+18.1	2229	+18.2	2230	+18.3	2231	+18.4	2232	+18.5	2233	+18.6	2234	+18.7	2235	+18.8	2236
+19.1	2239	+19.2	2240	+19.3	2241	+19.4	2242	+19.5	2243	+19.6	2244	+19.7	2245	+19.8	2246
+20.1	2249	+20.2	2250	+20.3	2251	+20.4	2252	+20.5	2253	+20.6	2254	+20.7	2255	+20.8	2256
+21.1	2259	+21.2	2260	+21.3	2261	+21.4	2262	+21.5	2263	+21.6	2264	+21.7	2265	+21.8	2266
+22.1	2269	+22.2	2270	+22.3	2271	+22.4	2272	+22.5	2273	+22.6	2274	+22.7	2275	+22.8	2276
+23.1	2279	+23.2	2280	+23.3	2281	+23.4	2282	+23.5	2283	+23.6	2284	+23.7	2285	+23.8	2286
+24.1	2289	+24.2	2290	+24.3	2291	+24.4	2292	+24.5	2293	+24.6	2294	+24.7	2295	+24.8	2296

Table 4. Mix-point Gain Only

Mic/line gain (①)

$\boxed{X62}$	$\boxed{X62}$ Value	\boxed{dB} Value	$\boxed{X62}$ Value													
-17.9	1869	-17.8	1870	-17.7	1871	-17.6	1872	-17.5	1873	-17.4	1874	-17.3	1875	-17.2	1876	-17.1
-16.9	1879	-16.8	1880	-16.7	1881	-16.6	1882	-16.5	1883	-16.4	1884	-16.3	1885	-16.2	1886	-16.1
-15.9	1889	-15.8	1890	-15.7	1891	-15.6	1892	-15.5	1893	-15.4	1894	-15.3	1895	-15.2	1896	-15.1
-14.9	1899	-14.8	1900	-14.7	1901	-14.6	1902	-14.5	1903	-14.4	1904	-14.3	1905	-14.2	1906	-14.1
-13.9	1909	-13.8	1910	-13.7	1911	-13.6	1912	-13.5	1913	-13.4	1914	-13.3	1915	-13.2	1916	-13.1
-12.9	1919	-12.8	1920	-12.7	1921	-12.6	1922	-12.5	1923	-12.4	1924	-12.3	1925	-12.2	1926	-12.1
-11.9	1929	-11.8	1930	-11.7	1931	-11.6	1932	-11.5	1933	-11.4	1934	-11.3	1935	-11.2	1936	-11.1
-10.9	1939	-10.8	1940	-10.7	1941	-10.6	1942	-10.5	1943	-10.4	1944	-10.3	1945	-10.2	1946	-10.1
-9.9	1949	-9.8	1950	-9.7	1951	-9.6	1952	-9.5	1953	-9.4	1954	-9.3	1955	-9.2	1956	-9.1
-8.9	1959	-8.8	1960	-8.7	1961	-8.6	1962	-8.5	1963	-8.4	1964	-8.3	1965	-8.2	1966	-8.1
-7.9	1969	-7.8	1970	-7.7	1971	-7.6	1972	-7.5	1973	-7.4	1974	-7.3	1975	-7.2	1976	-7.1
-6.9	1979	-6.8	1980	-6.7	1981	-6.6	1982	-6.5	1983	-6.4	1984	-6.3	1985	-6.2	1986	-6.1
-5.9	1989	-5.8	1990	-5.7	1991	-5.6	1992	-5.5	1993	-5.4	1994	-5.3	1995	-5.2	1996	-5.1
-4.9	1999	-4.8	2000	-4.7	2001	-4.6	2002	-4.5	2003	-4.4	2004	-4.3	2005	-4.2	2006	-4.1
-3.9	2009	-3.8	2010	-3.7	2011	-3.6	2012	-3.5	2013	-3.4	2014	-3.3	2015	-3.2	2016	-3.1
-2.9	2019	-2.8	2020	-2.7	2021	-2.6	2022	-2.5	2023	-2.4	2024	-2.3	2025	-2.2	2026	-2.1
-1.9	2029	-1.8	2030	-1.7	2031	-1.6	2032	-1.5	2033	-1.4	2034	-1.3	2035	-1.2	2036	-1.1
-0.9	2039	-0.8	2040	-0.7	2041	-0.6	2042	-0.5	2043	-0.4	2044	-0.3	2045	-0.2	2046	-0.1
+0.1	2049	+0.2	2050	+0.3	2051	+0.4	2052	+0.5	2053	+0.6	2054	+0.7	2055	+0.8	2056	+0.9
+1.1	2059	+1.2	2060	+1.3	2061	+1.4	2062	+1.5	2063	+1.6	2064	+1.7	2065	+1.8	2066	+1.9
+2.1	2069	+2.2	2070	+2.3	2071	+2.4	2072	+2.5	2073	+2.6	2074	+2.7	2075	+2.8	2076	+2.9
+3.1	2079	+3.2	2080	+3.3	2081	+3.4	2082	+3.5	2083	+3.6	2084	+3.7	2085	+3.8	2086	+3.9
+4.1	2089	+4.2	2090	+4.3	2091	+4.4	2092	+4.5	2093	+4.6	2094	+4.7	2095	+4.8	2096	+4.9
+5.1	2099	+5.2	2100	+5.3	2101	+5.4	2102	+5.5	2103	+5.6	2104	+5.7	2105	+5.8	2106	+5.9
+6.1	2109	+6.2	2110	+6.3	2111	+6.4	2112	+6.5	2113	+6.6	2114	+6.7	2115	+6.8	2116	+6.9
+7.1	2119	+7.2	2120	+7.3	2121	+7.4	2122	+7.5	2123	+7.6	2124	+7.7	2125	+7.8	2126	+7.9
+8.1	2129	+8.2	2130	+8.3	2131	+8.4	2132	+8.5	2133	+8.6	2134	+8.7	2135	+8.8	2136	+8.9
+9.1	2139	+9.2	2140	+9.3	2141	+9.4	2142	+9.5	2143	+9.6	2144	+9.7	2145	+9.8	2146	+9.9
+10.1	2149	+10.2	2150	+10.3	2151	+10.4	2152	+10.5	2153	+10.6	2154	+10.7	2155	+10.8	2156	+10.9
+11.1	2159	+11.2	2160	+11.3	2161	+11.4	2162	+11.5	2163	+11.6	2164	+11.7	2165	+11.8	2166	+11.9

Table 5. $\boxed{X62}$ – Level Control and Mix-point Selection

X62 — Mic/line gain (①), (continued)

dB Value	X62														
+12.1	2169	+12.2	2170	+12.3	2171	+12.4	2172	+12.5	2173	+12.6	2174	+12.7	2175	+12.8	2176
+13.1	2179	+13.2	2180	+13.3	2181	+13.4	2182	+13.5	2183	+13.6	2184	+13.7	2185	+13.8	2186
+14.1	2189	+14.2	2190	+14.3	2191	+14.4	2192	+14.5	2193	+14.6	2194	+14.7	2195	+14.8	2196
+15.1	2199	+15.2	2200	+15.3	2201	+15.4	2202	+15.5	2203	+15.6	2204	+15.7	2205	+15.8	2206
+16.1	2209	+16.2	2210	+16.3	2211	+16.4	2212	+16.5	2213	+16.6	2214	+16.7	2215	+16.8	2216
+17.1	2219	+17.2	2220	+17.3	2221	+17.4	2222	+17.5	2223	+17.6	2224	+17.7	2225	+17.8	2226
+18.1	2229	+18.2	2230	+18.3	2231	+18.4	2232	+18.5	2233	+18.6	2234	+18.7	2235	+18.8	2236
+19.1	2239	+19.2	2240	+19.3	2241	+19.4	2242	+19.5	2243	+19.6	2244	+19.7	2245	+19.8	2246
+20.1	2249	+20.2	2250	+20.3	2251	+20.4	2252	+20.5	2253	+20.6	2254	+20.7	2255	+20.8	2256
+21.1	2259	+21.2	2260	+21.3	2261	+21.4	2262	+21.5	2263	+21.6	2264	+21.7	2265	+21.8	2266
+22.1	2269	+22.2	2270	+22.3	2271	+22.4	2272	+22.5	2273	+22.6	2274	+22.7	2275	+22.8	2276
+23.1	2279	+23.2	2280	+23.3	2281	+23.4	2282	+23.5	2283	+23.6	2284	+23.7	2285	+23.8	2286
+24.1	2289	+24.2	2290	+24.3	2291	+24.4	2292	+24.5	2293	+24.6	2294	+24.7	2295	+24.8	2296
+25.1	2299	+25.2	2300	+25.3	2301	+25.4	2302	+25.5	2303	+25.6	2304	+25.7	2305	+25.8	2306
+26.1	2309	+26.2	2310	+26.3	2311	+26.4	2312	+26.5	2313	+26.6	2314	+26.7	2315	+26.8	2316
+27.1	2319	+27.2	2320	+27.3	2321	+27.4	2322	+27.5	2323	+27.6	2324	+27.7	2325	+27.8	2326
+28.1	2329	+28.2	2330	+28.3	2331	+28.4	2332	+28.5	2333	+28.6	2334	+28.7	2335	+28.8	2336
+29.1	2339	+29.2	2340	+29.3	2341	+29.4	2342	+29.5	2343	+29.6	2344	+29.7	2345	+29.8	2346
+30.1	2349	+30.2	2350	+30.3	2351	+30.4	2352	+30.5	2353	+30.6	2354	+30.7	2355	+30.8	2356
+31.1	2359	+31.2	2360	+31.3	2361	+31.4	2362	+31.5	2363	+31.6	2364	+31.7	2365	+31.8	2366
+32.1	2369	+32.2	2370	+32.3	2371	+32.4	2372	+32.5	2373	+32.6	2374	+32.7	2375	+32.8	2376
+33.1	2379	+33.2	2380	+33.3	2381	+33.4	2382	+33.5	2383	+33.6	2384	+33.7	2385	+33.8	2386
+34.1	2389	+34.2	2390	+34.3	2391	+34.4	2392	+34.5	2393	+34.6	2394	+34.7	2395	+34.8	2396
+35.1	2399	+35.2	2400	+35.3	2401	+35.4	2402	+35.5	2403	+35.6	2404	+35.7	2405	+35.8	2406
+36.1	2409	+36.2	2410	+36.3	2411	+36.4	2412	+36.5	2413	+36.6	2414	+36.7	2415	+36.8	2416
+37.1	2419	+37.2	2420	+37.3	2421	+37.4	2422	+37.5	2423	+37.6	2424	+37.7	2425	+37.8	2426
+38.1	2429	+38.2	2430	+38.3	2431	+38.4	2432	+38.5	2433	+38.6	2434	+38.7	2435	+38.8	2436
+39.1	2439	+39.2	2440	+39.3	2441	+39.4	2442	+39.5	2443	+39.6	2444	+39.7	2445	+39.8	2446
+40.1	2449	+40.2	2450	+40.3	2451	+40.4	2452	+40.5	2453	+40.6	2454	+40.7	2455	+40.8	2456
+41.1	2459	+41.2	2460	+41.3	2461	+41.4	2462	+41.5	2463	+41.6	2464	+41.7	2465	+41.8	2466
+42.1	2469	+42.2	2470	+42.3	2471	+42.4	2472	+42.5	2473	+42.6	2474	+42.7	2475	+42.8	2476

X62 — Mic/line gain (①), (continued)

dB Value	X62	dB Value	X62	cB Value	X62	dB Value	X62								
+43.1	2479	+43.2	2480	+43.3	2481	+43.4	2482	+43.5	2483	+43.6	2484	+43.7	2485	+43.8	2486
+44.1	2489	+44.2	2490	+44.3	2491	+44.4	2492	+44.5	2493	+44.6	2494	+44.7	2495	+44.8	2496
+45.1	2499	+45.2	2500	+45.3	2501	+45.4	2502	+45.5	2503	+45.6	2504	+45.7	2505	+45.8	2506
+46.1	2509	+46.2	2510	+46.3	2511	+46.4	2512	+46.5	2513	+46.6	2514	+46.7	2515	+46.8	2516
+47.1	2519	+47.2	2520	+47.3	2521	+47.4	2522	+47.5	2523	+47.6	2524	+47.7	2525	+47.8	2526
+48.1	2529	+48.2	2530	+48.3	2531	+48.4	2532	+48.5	2533	+48.6	2534	+48.7	2535	+48.8	2536
+49.1	2539	+49.2	2540	+49.3	2541	+49.4	2542	+49.5	2543	+49.6	2544	+49.7	2545	+49.8	2546
+50.1	2549	+50.2	2550	+50.3	2551	+50.4	2552	+50.5	2553	+50.6	2554	+50.7	2555	+50.8	2556
+51.1	2559	+51.2	2560	+51.3	2561	+51.4	2562	+51.5	2563	+51.6	2564	+51.7	2565	+51.8	2566
+52.1	2569	+52.2	2570	+52.3	2571	+52.4	2572	+52.5	2573	+52.6	2574	+52.7	2575	+52.8	2576
+53.1	2579	+53.2	2580	+53.3	2581	+53.4	2582	+53.5	2583	+53.6	2584	+53.7	2585	+53.8	2586
+54.1	2589	+54.2	2590	+54.3	2591	+54.4	2592	+54.5	2593	+54.6	2594	+54.7	2595	+54.8	2596
+55.1	2599	+55.2	2600	+55.3	2601	+55.4	2602	+55.5	2603	+55.6	2604	+55.7	2605	+55.8	2606
+56.1	2609	+56.2	2610	+56.3	2611	+56.4	2612	+56.5	2613	+56.6	2614	+56.7	2615	+56.8	2616
+57.1	2619	+57.2	2620	+57.3	2621	+57.4	2622	+57.5	2623	+57.6	2624	+57.7	2625	+57.8	2626
+58.1	2629	+58.2	2630	+58.3	2631	+58.4	2632	+58.5	2633	+58.6	2634	+58.7	2635	+58.8	2636
+59.1	2639	+59.2	2640	+59.3	2641	+59.4	2642	+59.5	2643	+59.6	2644	+59.7	2645	+59.8	2646
+60.1	2649	+60.2	2650	+60.3	2651	+60.4	2652	+60.5	2653	+60.6	2654	+60.7	2655	+60.8	2656
+61.1	2659	+61.2	2660	+61.3	2661	+61.4	2662	+61.5	2663	+61.6	2664	+61.7	2665	+61.8	2666
+62.1	2669	+62.2	2670	+62.3	2671	+62.4	2672	+62.5	2673	+62.6	2674	+62.7	2675	+62.8	2676
+63.1	2679	+63.2	2680	+63.3	2681	+63.4	2682	+63.5	2683	+63.6	2684	+63.7	2685	+63.8	2686
+64.1	2689	+64.2	2690	+64.3	2691	+64.4	2692	+64.5	2693	+64.6	2694	+64.7	2695	+64.8	2696
+65.1	2699	+65.2	2700	+65.3	2701	+65.4	2702	+65.5	2703	+65.6	2704	+65.7	2705	+65.8	2706
+66.1	2709	+66.2	2710	+66.3	2711	+66.4	2712	+66.5	2713	+66.6	2714	+66.7	2715	+66.8	2716
+67.1	2719	+67.2	2720	+67.3	2721	+67.4	2722	+67.5	2723	+67.6	2724	+67.7	2725	+67.8	2726
+68.1	2729	+68.2	2730	+68.3	2731	+68.4	2732	+68.5	2733	+68.6	2734	+68.7	2735	+68.8	2736
+69.1	2739	+69.2	2740	+69.3	2741	+69.4	2742	+69.5	2743	+69.6	2744	+69.7	2745	+69.8	2746
+70.1	2749	+70.2	2750	+70.3	2751	+70.4	2752	+70.5	2753	+70.6	2754	+70.7	2755	+70.8	2756
+71.1	2759	+71.2	2760	+71.3	2761	+71.4	2762	+71.5	2763	+71.6	2764	+71.7	2765	+71.8	2766
+72.1	2769	+72.2	2770	+72.3	2771	+72.4	2772	+72.5	2773	+72.6	2774	+72.7	2775	+72.8	2776

[X62] – Mic/line gain (◎), (continued)

dB Value	[X62]												
+73.1	2779	+73.2	2780	+73.3	2781	+73.4	2782	+73.5	2783	+73.6	2784	+73.7	2785
+74.1	2789	+74.2	2790	+74.3	2791	+74.4	2792	+74.5	2793	+74.6	2794	+74.7	2795
+75.1	2799	+75.2	2800	+75.3	2801	+75.4	2802	+75.5	2803	+75.6	2804	+75.7	2805
+76.1	2809	+76.2	2810	+76.3	2811	+76.4	2812	+76.5	2813	+76.6	2814	+76.7	2815
+77.1	2819	+77.2	2820	+77.3	2821	+77.4	2822	+77.5	2823	+77.6	2824	+77.7	2825
+78.1	2829	+78.2	2830	+78.3	2831	+78.4	2832	+78.5	2833	+78.6	2834	+78.7	2835
+79.1	2839	+79.2	2840	+79.3	2841	+79.4	2842	+79.5	2843	+79.6	2844	+79.7	2845

X63 – Pre-mixer gain (②), Virtual return gain (⑥), and output volume (⑤)

NOTE: Pre-mixer gain (②) and virtual return gain (⑥) range: -100.0 dB to +12.0 dB.
Output volume (⑤) range: -100.0 dB to 0.0 dB.

dB Value	X63														
-99.9	1049	-99.8	1050	-99.7	1051	-99.6	1052	-99.5	1053	-99.4	1054	-99.3	1055	-99.2	1056
-98.9	1059	-98.8	1060	-98.7	1061	-98.6	1062	-98.5	1063	-98.4	1064	-98.3	1065	-98.2	1066
-97.9	1069	-97.8	1070	-97.7	1071	-97.6	1072	-97.5	1073	-97.4	1074	-97.3	1075	-97.2	1076
-96.9	1079	-96.8	1080	-96.7	1081	-96.6	1082	-96.5	1083	-96.4	1084	-96.3	1085	-96.2	1086
-95.9	1089	-95.8	1090	-95.7	1091	-95.6	1092	-95.5	1093	-95.4	1094	-95.3	1095	-95.2	1096
-94.9	1099	-94.8	1100	-94.7	1101	-94.6	1102	-94.5	1103	-94.4	1104	-94.3	1105	-94.2	1106
-93.9	1109	-93.8	1110	-93.7	1111	-93.6	1112	-93.5	1113	-93.4	1114	-93.3	1115	-93.2	1116
-92.9	1119	-92.8	1120	-92.7	1121	-92.6	1122	-92.5	1123	-92.4	1124	-92.3	1125	-92.2	1126
-91.9	1129	-91.8	1130	-91.7	1131	-91.6	1132	-91.5	1133	-91.4	1134	-91.3	1135	-91.2	1136
-90.9	1139	-90.8	1140	-90.7	1141	-90.6	1142	-90.5	1143	-90.4	1144	-90.3	1145	-90.2	1146
-89.9	1149	-89.8	1150	-89.7	1151	-89.6	1152	-89.5	1153	-89.4	1154	-89.3	1155	-89.2	1156
-88.9	1159	-88.8	1160	-88.7	1161	-88.6	1162	-88.5	1163	-88.4	1164	-88.3	1165	-88.2	1166
-87.9	1169	-87.8	1170	-87.7	1171	-87.6	1172	-87.5	1173	-87.4	1174	-87.3	1175	-87.2	1176
-86.9	1179	-86.8	1180	-86.7	1181	-86.6	1182	-86.5	1183	-86.4	1184	-86.3	1185	-86.2	1186
-85.9	1189	-85.8	1190	-85.7	1191	-85.6	1192	-85.5	1193	-85.4	1194	-85.3	1195	-85.2	1196
-84.9	1199	-84.8	1200	-84.7	1201	-84.6	1202	-84.5	1203	-84.4	1204	-84.3	1205	-84.2	1206
-83.9	1209	-83.8	1210	-83.7	1211	-83.6	1212	-83.5	1213	-83.4	1214	-83.3	1215	-83.2	1216
-82.9	1219	-82.8	1220	-82.7	1221	-82.6	1222	-82.5	1223	-82.4	1224	-82.3	1225	-82.2	1226
-81.9	1229	-81.8	1230	-81.7	1231	-81.6	1232	-81.5	1233	-81.4	1234	-81.3	1235	-81.2	1236
-80.9	1239	-80.8	1240	-80.7	1241	-80.6	1242	-80.5	1243	-80.4	1244	-80.3	1245	-80.2	1246
-79.9	1249	-79.8	1250	-79.7	1251	-79.6	1252	-79.5	1253	-79.4	1254	-79.3	1255	-79.2	1256
-78.9	1259	-78.8	1260	-78.7	1261	-78.6	1262	-78.5	1263	-78.4	1264	-78.3	1265	-78.2	1266
-77.9	1269	-77.8	1270	-77.7	1271	-77.6	1272	-77.5	1273	-77.4	1274	-77.3	1275	-77.2	1276
-76.9	1279	-76.8	1280	-76.7	1281	-76.6	1282	-76.5	1283	-76.4	1284	-76.3	1285	-76.2	1286
-75.9	1289	-75.8	1290	-75.7	1291	-75.6	1292	-75.5	1293	-75.4	1294	-75.3	1295	-75.2	1296
-74.9	1299	-74.8	1300	-74.7	1301	-74.6	1302	-74.5	1303	-74.4	1304	-74.3	1305	-74.2	1306
-73.9	1309	-73.8	1310	-73.7	1311	-73.6	1312	-73.5	1313	-73.4	1314	-73.3	1315	-73.2	1316
-72.9	1319	-72.8	1320	-72.7	1321	-72.6	1322	-72.5	1323	-72.4	1324	-72.3	1325	-72.2	1326
-71.9	1329	-71.8	1330	-71.7	1331	-71.6	1332	-71.5	1333	-71.4	1334	-71.3	1335	-71.2	1336

Table 6. X63 – Pre-mixer, Virtual Return, and Output Level Control

[x63] — Pre-mixer gain (②), Virtual return gain (⑥), and output volume (⑤), (continued)

dB Value	[x63]														
-70.9	1339	-70.8	1340	-70.7	1341	-70.6	1342	-70.5	1343	-70.4	1344	-70.3	1345	-70.2	1346
-69.9	1349	-69.8	1350	-69.7	1351	-69.6	1352	-69.5	1353	-69.4	1354	-69.3	1355	-69.2	1356
-68.9	1359	-68.8	1360	-68.7	1361	-68.6	1362	-68.5	1363	-68.4	1364	-68.3	1365	-68.2	1366
-67.9	1369	-67.8	1370	-67.7	1371	-67.6	1372	-67.5	1373	-67.4	1374	-67.3	1375	-67.2	1376
-66.9	1379	-66.8	1380	-66.7	1381	-66.6	1382	-66.5	1383	-66.4	1384	-66.3	1385	-66.2	1386
-65.9	1389	-65.8	1390	-65.7	1391	-65.6	1392	-65.5	1393	-65.4	1394	-65.3	1395	-65.2	1396
-64.9	1399	-64.8	1400	-64.7	1401	-64.6	1402	-64.5	1403	-64.4	1404	-64.3	1405	-64.2	1406
-63.9	1409	-63.8	1410	-63.7	1411	-63.6	1412	-63.5	1413	-63.4	1414	-63.3	1415	-63.2	1416
-62.9	1419	-62.8	1420	-62.7	1421	-62.6	1422	-62.5	1423	-62.4	1424	-62.3	1425	-62.2	1426
-61.9	1429	-61.8	1430	-61.7	1431	-61.6	1432	-61.5	1433	-61.4	1434	-61.3	1435	-61.2	1436
-60.9	1439	-60.8	1440	-60.7	1441	-60.6	1442	-60.5	1443	-60.4	1444	-60.3	1445	-60.2	1446
-59.9	1449	-59.8	1450	-59.7	1451	-59.6	1452	-59.5	1453	-59.4	1454	-59.3	1455	-59.2	1456
-58.9	1459	-58.8	1460	-58.7	1461	-58.6	1462	-58.5	1463	-58.4	1464	-58.3	1465	-58.2	1466
-57.9	1469	-57.8	1470	-57.7	1471	-57.6	1472	-57.5	1473	-57.4	1474	-57.3	1475	-57.2	1476
-56.9	1479	-56.8	1480	-56.7	1481	-56.6	1482	-56.5	1483	-56.4	1484	-56.3	1485	-56.2	1486
-55.9	1489	-55.8	1490	-55.7	1491	-55.6	1492	-55.5	1493	-55.4	1494	-55.3	1495	-55.2	1496
-54.9	1499	-54.8	1500	-54.7	1501	-54.6	1502	-54.5	1503	-54.4	1504	-54.3	1505	-54.2	1506
-53.9	1509	-53.8	1510	-53.7	1511	-53.6	1512	-53.5	1513	-53.4	1514	-53.3	1515	-53.2	1516
-52.9	1519	-52.8	1520	-52.7	1521	-52.6	1522	-52.5	1523	-52.4	1524	-52.3	1525	-52.2	1526
-51.9	1529	-51.8	1530	-51.7	1531	-51.6	1532	-51.5	1533	-51.4	1534	-51.3	1535	-51.2	1536
-50.9	1539	-50.8	1540	-50.7	1541	-50.6	1542	-50.5	1543	-50.4	1544	-50.3	1545	-50.2	1546
-49.9	1549	-49.8	1550	-49.7	1551	-49.6	1552	-49.5	1553	-49.4	1554	-49.3	1555	-49.2	1556
-48.9	1559	-48.8	1560	-48.7	1561	-48.6	1562	-48.5	1563	-48.4	1564	-48.3	1565	-48.2	1566
-47.9	1569	-47.8	1570	-47.7	1571	-47.6	1572	-47.5	1573	-47.4	1574	-47.3	1575	-47.2	1576
-46.9	1579	-46.8	1580	-46.7	1581	-46.6	1582	-46.5	1583	-46.4	1584	-46.3	1585	-46.2	1586
-45.9	1589	-45.8	1590	-45.7	1591	-45.6	1592	-45.5	1593	-45.4	1594	-45.3	1595	-45.2	1596
-44.9	1599	-44.8	1600	-44.7	1601	-44.6	1602	-44.5	1603	-44.4	1604	-44.3	1605	-44.2	1606
-43.9	1609	-43.8	1610	-43.7	1611	-43.6	1612	-43.5	1613	-43.4	1614	-43.3	1615	-43.2	1616
-42.9	1619	-42.8	1620	-42.7	1621	-42.6	1622	-42.5	1623	-42.4	1624	-42.3	1625	-42.2	1626
-41.9	1629	-41.8	1630	-41.7	1631	-41.6	1632	-41.5	1633	-41.4	1634	-41.3	1635	-41.2	1636

[63] — Pre-mixer gain (②), Virtual return gain (⑥), and output volume (⑤), (continued)

dB Value	[X63]														
-40.9	1639	-40.8	1640	-40.7	1641	-40.6	1642	-40.5	1643	-40.4	1644	-40.3	1645	-40.2	1646
-39.9	1649	-39.8	1650	-39.7	1651	-39.6	1652	-39.5	1653	-39.4	1654	-39.3	1655	-39.2	1656
-38.9	1659	-38.8	1660	-38.7	1661	-38.6	1662	-38.5	1663	-38.4	1664	-38.3	1665	-38.2	1666
-37.9	1669	-37.8	1670	-37.7	1671	-37.6	1672	-37.5	1673	-37.4	1674	-37.3	1675	-37.2	1676
-36.9	1679	-36.8	1680	-36.7	1681	-36.6	1682	-36.5	1683	-36.4	1684	-36.3	1685	-36.2	1686
-35.9	1689	-35.8	1690	-35.7	1691	-35.6	1692	-35.5	1693	-35.4	1694	-35.3	1695	-35.2	1696
-34.9	1699	-34.8	1700	-34.7	1701	-34.6	1702	-34.5	1703	-34.4	1704	-34.3	1705	-34.2	1706
-33.9	1709	-33.8	1710	-33.7	1711	-33.6	1712	-33.5	1713	-33.4	1714	-33.3	1715	-33.2	1716
-32.9	1719	-32.8	1720	-32.7	1721	-32.6	1722	-32.5	1723	-32.4	1724	-32.3	1725	-32.2	1726
-31.9	1729	-31.8	1730	-31.7	1731	-31.6	1732	-31.5	1733	-31.4	1734	-31.3	1735	-31.2	1736
-30.9	1739	-30.8	1740	-30.7	1741	-30.6	1742	-30.5	1743	-30.4	1744	-30.3	1745	-30.2	1746
-29.9	1749	-29.8	1750	-29.7	1751	-29.6	1752	-29.5	1753	-29.4	1754	-29.3	1755	-29.2	1756
-28.9	1759	-28.8	1760	-28.7	1761	-28.6	1762	-28.5	1763	-28.4	1764	-28.3	1765	-28.2	1766
-27.9	1769	-27.8	1770	-27.7	1771	-27.6	1772	-27.5	1773	-27.4	1774	-27.3	1775	-27.2	1776
-26.9	1779	-26.8	1780	-26.7	1781	-26.6	1782	-26.5	1783	-26.4	1784	-26.3	1785	-26.2	1786
-25.9	1789	-25.8	1790	-25.7	1791	-25.6	1792	-25.5	1793	-25.4	1794	-25.3	1795	-25.2	1796
-24.9	1799	-24.8	1800	-24.7	1801	-24.6	1802	-24.5	1803	-24.4	1804	-24.3	1805	-24.2	1806
-23.9	1809	-23.8	1810	-23.7	1811	-23.6	1812	-23.5	1813	-23.4	1814	-23.3	1815	-23.2	1816
-22.9	1819	-22.8	1820	-22.7	1821	-22.6	1822	-22.5	1823	-22.4	1824	-22.3	1825	-22.2	1826
-21.9	1829	-21.8	1830	-21.7	1831	-21.6	1832	-21.5	1833	-21.4	1834	-21.3	1835	-21.2	1836
-20.9	1839	-20.8	1840	-20.7	1841	-20.6	1842	-20.5	1843	-20.4	1844	-20.3	1845	-20.2	1846
-19.9	1849	-19.8	1850	-19.7	1851	-19.6	1852	-19.5	1853	-19.4	1854	-19.3	1855	-19.2	1856
-18.9	1859	-18.8	1860	-18.7	1861	-18.6	1862	-18.5	1863	-18.4	1864	-18.3	1865	-18.2	1866
-17.9	1869	-17.8	1870	-17.7	1871	-17.6	1872	-17.5	1873	-17.4	1874	-17.3	1875	-17.2	1876
-16.9	1879	-16.8	1880	-16.7	1881	-16.6	1882	-16.5	1883	-16.4	1884	-16.3	1885	-16.2	1886
-15.9	1889	-15.8	1890	-15.7	1891	-15.6	1892	-15.5	1893	-15.4	1894	-15.3	1895	-15.2	1896
-14.9	1899	-14.8	1900	-14.7	1901	-14.6	1902	-14.5	1903	-14.4	1904	-14.3	1905	-14.2	1906
-13.9	1909	-13.8	1910	-13.7	1911	-13.6	1912	-13.5	1913	-13.4	1914	-13.3	1915	-13.2	1916
-12.9	1919	-12.8	1920	-12.7	1921	-12.6	1922	-12.5	1923	-12.4	1924	-12.3	1925	-12.2	1926
-11.9	1929	-11.8	1930	-11.7	1931	-11.6	1932	-11.5	1933	-11.4	1934	-11.3	1935	-11.2	1936

[x63] — Pre-mixer gain (②), Virtual return gain (⑥), and output volume (⑤), (continued)

dB Value	[x63]														
-10.9	1939	-10.8	1940	-10.7	1941	-10.6	1942	-10.5	1943	-10.4	1944	-10.3	1945	-10.2	1946
-9.9	1949	-9.8	1950	-9.7	1951	-9.6	1952	-9.5	1953	-9.4	1954	-9.3	1955	-9.2	1956
-8.9	1959	-8.8	1960	-8.7	1961	-8.6	1962	-8.5	1963	-8.4	1964	-8.3	1965	-8.2	1966
-7.9	1969	-7.8	1970	-7.7	1971	-7.6	1972	-7.5	1973	-7.4	1974	-7.3	1975	-7.2	1976
-6.9	1979	-6.8	1980	-6.7	1981	-6.6	1982	-6.5	1983	-6.4	1984	-6.3	1985	-6.2	1986
-5.9	1989	-5.8	1990	-5.7	1991	-5.6	1992	-5.5	1993	-5.4	1994	-5.3	1995	-5.2	1996
-4.9	1999	-4.8	2000	-4.7	2001	-4.6	2002	-4.5	2003	-4.4	2004	-4.3	2005	-4.2	2006
-3.9	2009	-3.8	2010	-3.7	2011	-3.6	2012	-3.5	2013	-3.4	2014	-3.3	2015	-3.2	2016
-2.9	2019	-2.8	2020	-2.7	2021	-2.6	2022	-2.5	2023	-2.4	2024	-2.3	2025	-2.2	2026
-1.9	2029	-1.8	2030	-1.7	2031	-1.6	2032	-1.5	2033	-1.4	2034	-1.3	2035	-1.2	2036
-0.9	2039	-0.8	2040	-0.7	2041	-0.6	2042	-0.5	2043	-0.4	2044	-0.3	2045	-0.2	2046
														-0.1	2047
														0.0	2048

Pre-mixer gain (②), Virtual return gain (⑥) only

	+0.1	2049	+0.2	2050	+0.3	2051	+0.4	2052	+0.5	2053	+0.6	2054	+0.7	2055	+0.8	2056	+0.9	2057	+1.0	2058
+1.1	2059	+1.2	2060	+1.3	2061	+1.4	2062	+1.5	2063	+1.6	2064	+1.7	2065	+1.8	2066	+1.9	2067	+2.0	2068	
+2.1	2069	+2.2	2070	+2.3	2071	+2.4	2072	+2.5	2073	+2.6	2074	+2.7	2075	+2.8	2076	+2.9	2077	+3.0	2078	
+3.1	2079	+3.2	2080	+3.3	2081	+3.4	2082	+3.5	2083	+3.6	2084	+3.7	2085	+3.8	2086	+3.9	2087	+4.0	2088	
+4.1	2089	+4.2	2090	+4.3	2091	+4.4	2092	+4.5	2093	+4.6	2094	+4.7	2095	+4.8	2096	+4.9	2097	+5.0	2098	
+5.1	2099	+5.2	2100	+5.3	2101	+5.4	2102	+5.5	2103	+5.6	2104	+5.7	2105	+5.8	2106	+5.9	2107	+6.0	2108	
+6.1	2109	+6.2	2110	+6.3	2111	+6.4	2112	+6.5	2113	+6.6	2114	+6.7	2115	+6.8	2116	+6.9	2117	+7.0	2118	
+7.1	2119	+7.2	2120	+7.3	2121	+7.4	2122	+7.5	2123	+7.6	2124	+7.7	2125	+7.8	2126	+7.9	2127	+8.0	2128	
+8.1	2129	+8.2	2130	+8.3	2131	+8.4	2132	+8.5	2133	+8.6	2134	+8.7	2135	+8.8	2136	+8.9	2137	+9.0	2138	
+9.1	2139	+9.2	2140	+9.3	2141	+9.4	2142	+9.5	2143	+9.6	2144	+9.7	2145	+9.8	2146	+9.9	2147	+10.0	2148	
+10.1	2149	+10.2	2150	+10.3	2151	+10.4	2152	+10.5	2153	+10.6	2154	+10.7	2155	+10.8	2156	+10.9	2157	+11.0	2158	
+11.1	2159	+11.2	2160	+11.3	2161	+11.4	2162	+11.5	2163	+11.6	2164	+11.7	2165	+11.8	2166	+11.9	2167	+12.0	2168	

HTML Operation

This section describes HTML operation and control of the DMP 64, including:

- [Download the Startup Page](#)
- [Status Tab](#)
- [Configuration Tab](#)
- [File Management Tab](#)
- [Control Tab](#)
- [Special Characters](#)

The DMP 64 can be controlled and operated through its Ethernet port, connected by LAN or WAN, using a web browser such as the Microsoft® Internet Explorer. The browser display of device status or operation has the appearance of web pages. This chapter describes the factory-installed HTML pages, which are always available and cannot be erased or overwritten.

NOTE: If the Ethernet connection to the device is unstable, try turning off the proxy server in the Web browser. In Microsoft Internet Explorer, click **Tools > Internet Options > Connections > LAN Settings**, uncheck the "**Use a proxy server...**" box, and then click **OK**.

Download the Startup Page

Access the device using HTML pages as follows:

1. Start the Web browser program.
2. Click in the browser **Address** field.
3. Enter the device IP address directly into the **Address** field.

NOTE: If the local system administrators have not changed the value, the factory-specified default IP address is 192.168.254.254.

4. If a custom display page is available, enter a slash (/) and the file name to open.

NOTE: The browser **Address** field should display the address in the following format: xxx.xxx.xxx.xxx/{*optional_file_name*.HTML}. The following characters are invalid in file names:
{space} + ~ , @ = ' [] { } < > ' " ; : > \ ?

5. Press the keyboard <Enter> key. The device checks to see if it is password-protected.
 - a. If the device is not password-protected, it checks and downloads the HTML pages (proceed to step 7).
 - b. If the device is password-protected, the device downloads the Connect to page (see figure 62).



Figure 62. Connect To Page

6. Click in the Password field and type in the appropriate administrator or user password. Click the **OK** button.

NOTE: A User Name entry is not required.

7. The device checks several possibilities, in the following order, and then responds accordingly:
 - a. Does the address include a specific file name, such as **10.13.156.10/file_name.HTML**? **If true**, the device downloads that HTML page.
 - b. Is there a file in the device memory named "**index.HTML**"? **If true**, the device downloads "index.HTML" as the default startup page.
 - c. **If neither of the conditions are true**, the device downloads the factory-installed default startup page, "**hortxe_index.HTML**" (see figure 63 on the next page), also known as the System Status page.

Status Tab

System Status Page

The System Status page (see figure 63) provides an overall view of the status of the device, including system information, power supply status, and serial port settings. The System Status page is the default page when establishing a connection to the device. Access the System Status page from other pages by clicking the **Status** tab.

The screenshot shows a Windows Internet Explorer window displaying the 'Extron Electronics' website. The main menu at the top includes 'Status', 'Configuration', 'File Management', and 'Control'. The 'Status' tab is selected, leading to the 'System Status' page. The page is divided into several sections: 'System Information' (Unit Name: DMP-64-04-EC-7F, Model: DMP 64, Part Number: 60-1054-01, Date: 8/12/2009, Time: 4:33 PM), 'Power Status' (listing voltages for +3.3V, +5V, -5V, +15V, and -15V), and 'Serial Port Settings' (two tables showing port configurations for RS-232 and RS-422/485). The bottom of the page shows the URL 'http://10.13.193.109/nortve_status.html' and a status bar indicating 'Local intranet' and '100%'. The status bar also shows 'Logged on: Admin' and 'Log Off'.

Figure 63. System Status Page

Configuration Tab

System Settings Page

Click the **Configuration** tab to download the System Settings page (see figure 64). The screen consists of fields to view and edit IP administration and system settings. Passwords and Firmware Upgrade pages are accessed by clicking the appropriate link on the left. For basic information about IP addresses and subnetting, see **Ethernet (LAN) Port** on page 93.

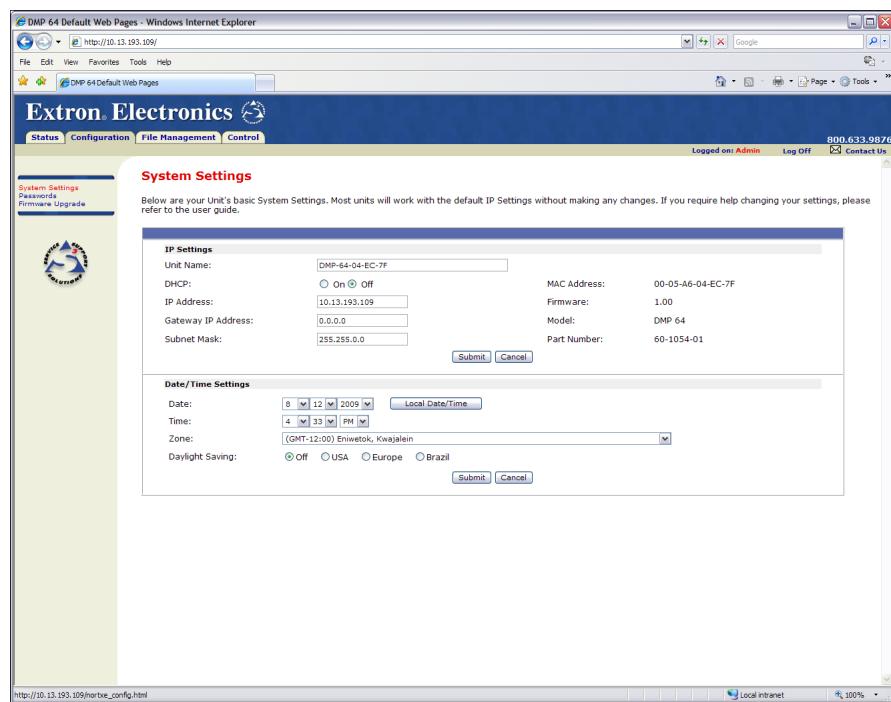


Figure 64. System Settings Page

On password-protected connections, there are two levels of protection: administrator and user. Administrators have full access to the Passwords and Firmware Upgrade pages. Users have view only access.

- Ethernet connection to the device, either entering SIS commands (see **SIS Programming and Control** on page 92), or using the Extron DSP Configurator Program (see **DMP Software** on page 14), is password-protected.
- Connection via any RS-232 port **is not** password-protected.

IP Settings Fields

The IP settings fields provide a location for viewing and editing settings unique to the Ethernet interface. After editing any of the settings on this page, click the **Submit** button at the bottom of the section to make the changes.

Unit Name Field

The **Unit Name** field contains the name of the device. This name field can be changed to any valid name, up to 24 alphanumeric characters.

NOTE: The following characters are invalid in the matrix name:
+ ~ , @ = ' [] { } < > ' " ; : > \ and ?.

DHCP Selection

The **DHCP On** selection directs the device to ignore any entered IP addresses and obtain its IP address from a Dynamic Host Configuration Protocol (DHCP) server (if the network is DHCP capable). The **DHCP Off** selection turns DHCP off. Contact the local system administrator for additional information on your network.

IP Address Field

The **IP Address** field contains the IP address encoded in the flash memory of the connected device.

Valid IP addresses consist of four 1-, 2-, or 3-digit numeric subfields separated by dots (periods). Each field can be numbered from **000** through **255**. Leading zeroes, up to three digits total per field, are optional. Values of **256** and above are invalid.

The factory-installed default address is **192.168.254.254**, but if this conflicts with other equipment at the installation site, change the IP address to any valid value.

NOTE: IP address changes can cause conflicts with other equipment. Only local system administrators should change IP addresses.

Gateway IP Address Field

The **Gateway IP Address** field identifies the address of the gateway to the mail server to be used if the device and the mail server are not on the same subnet.

The gateway IP address has the same validity rules as the system IP address.

Subnet Mask Field

The **Subnet Mask** field is used to determine whether the device is on the same subnet as the mail server when you are subnetting.

MAC Address Field

The Media Access Control (MAC) address is hardcoded in the device and cannot be changed.

Firmware Field

The Firmware field displays the current firmware version being used by the device.

Model Field

The Model field displays the Extron model number of the device.

Part Number Field

The Part Number field displays the Extron Electronics part number of the device.

Date/Time Settings Fields

The **Date/Time settings** fields (see figure 65) provide a location for viewing and setting the time functions.

The screenshot shows a 'Date/Time Settings' form. It includes fields for Date (10/16), Time (11:06), Zone (GMT-08:00), and Daylight Saving (Off). A dropdown menu for the year is open, showing options from 2000 to 2010, with '2008' selected. Buttons for 'Submit' and 'Cancel' are at the bottom right.

Figure 65. Date/Time Settings Fields

Change the date and time settings as follows:

1. Click the desired variable box. Adjustable settings include month, day, year, hours, minutes, AM/PM, and (time) zone. A drop-down box appears (the year drop-down box is selected in figure 65).
2. If all variable selections are not visible, click and drag the slider or click the up button or down button until the desired variable is visible.
3. Click the desired variable.

NOTE: If setting the time, set the local time. The **Zone** variable allows you to then select the offset from Greenwich Mean Time (GMT).

The Zone field identifies the standard time zone selected and displays the amount of time, in hours and minutes, the local time varies from GMT international time reference.

4. Repeat steps 1 through 3 for other variables that need to be changed.
5. If appropriate, click the **Daylight Savings** selection to turn on the daylight savings time feature.

NOTE: When Daylight Saving Time is on, the device automatically updates its internal clock between Standard Time and Daylight Saving Time in the spring and fall on the date the time change occurs in the country or region selected. When Daylight Saving Time is turned off, the device does not adjust its time reference.

6. Click **Submit** to enter the changes.

Passwords Page

Access the **Passwords** page (see figure 66), by clicking the **Passwords** link on the system settings page.

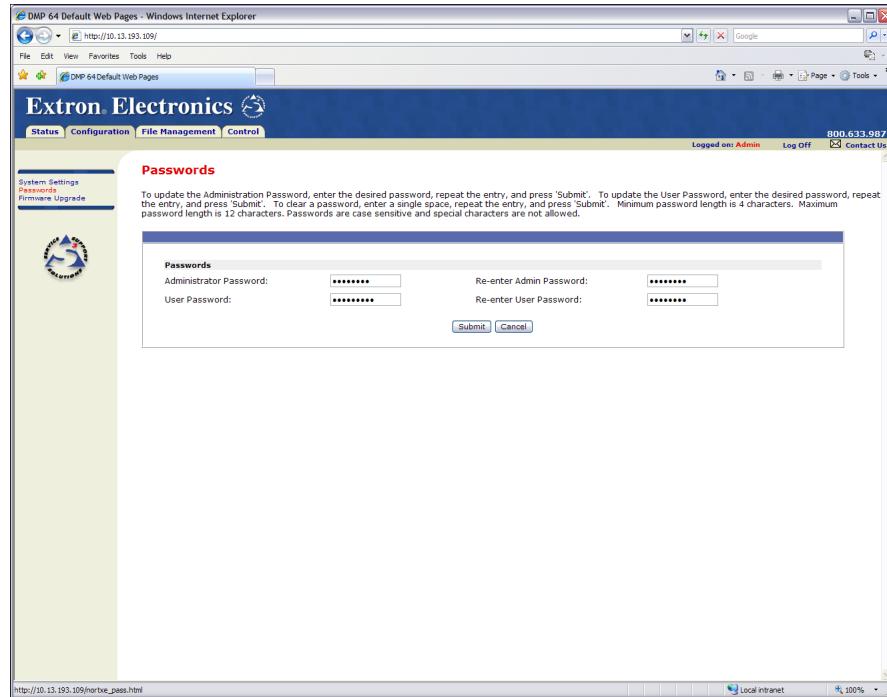


Figure 66. Passwords Page

The fields on the passwords page are for entering and verifying administrator and user passwords. Passwords are case sensitive and limited to 12 uppercase and lowercase alphanumeric characters. Each password must be entered twice: once in the User Password field and then again in the Re-enter User Password field. Characters in these fields are masked by asterisks (*****). If password protection is not desired, leave the User Password field and the Re-enter User Password field blank. After entering the desired password in both fields, click **Submit**.

NOTE: An administrator password must be created before a user password can be created.

Change a Password

To change a password, type the new password in the User Password and Re-enter User Password fields and click **Submit**.

Clear a Password

To clear an existing password so that no password is required, enter a single space in the User Password and Re-enter User Password fields and click **Submit**.

Firmware Upgrade Page

The Firmware Upgrade page provides a way to verify the current firmware version and to replace the firmware without taking the device out of service. Click the **Firmware Upgrade** link on the System Configuration page to access the Firmware Upgrade page (see figure 67).

The current firmware version is displayed above the upload box for reference.

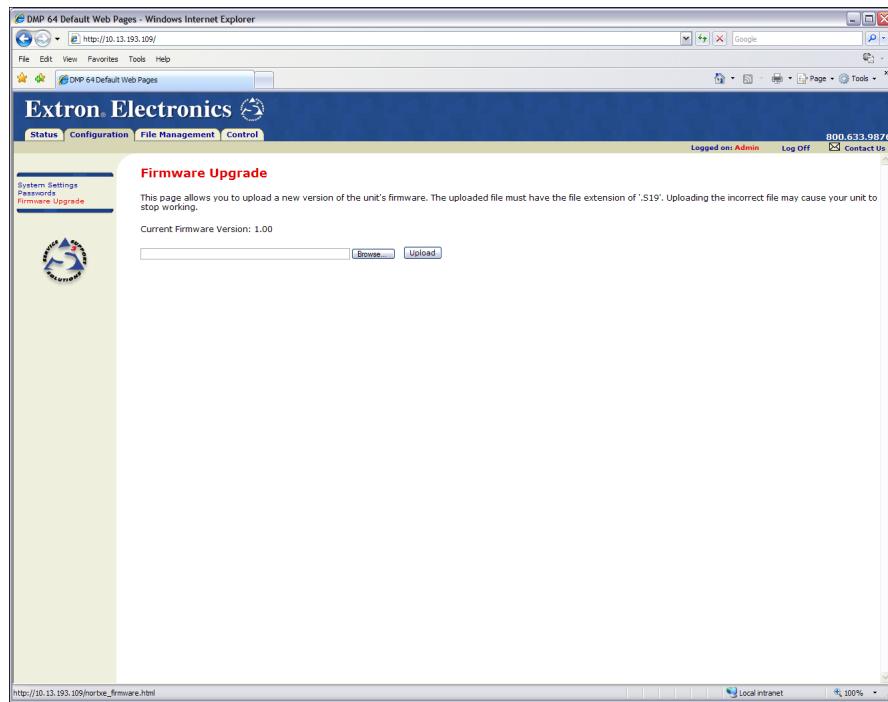


Figure 67. Firmware Upgrade Page

To update the device firmware:

NOTE: The Firmware Upgrade page is **only** for replacing the firmware that controls device operation.

1. Visit the Extron Web site, www.extron.com, and click the **Download Center** tab.
2. Click the **Firmware** link (see figure 68 on the next page).
3. Select the appropriate firmware file and click **Download**.
4. Enter the requested information.

5. Click **Download** to copy the firmware to your computer.

The screenshot shows a website navigation bar with links for Products, Applications, Technologies, Company, and Download. A callout box labeled 'NOTE:' states: 'The version, release date, and size shown are example values only.' Below the navigation is a 'Download Center' section titled 'Firmware (28 files)'. It lists a file: 'DMP 64 Digital Matrix Processor Firmware for DMP 64' (version V1.01, released January 17, 2011, 2.2 MB). A red circle labeled '1' highlights the 'Download' link in the navigation bar. A red circle labeled '2' highlights the 'Firmware' link in the dropdown menu. A red circle labeled '3' highlights the 'Download' link next to the file listing. Below this, a larger 'Download Center' form is shown with fields for Name, Company, Title, and E-mail, all highlighted with a red circle labeled '4'. A red circle labeled '5' highlights the 'Download DMP64_FW1x01.exe' button.

NOTE: The version, release date, and size shown are example values only.

Download Center

Firmware (28 files)

- DMP 64 Digital Matrix Processor Firmware for DMP 64

19-2247-50 V1.01 January 17, 2011 2.2 MB

Release Notes

Download

Download Center

Download DMP 64 FW1x01.exe

Please provide the following information.

4

* Name: John Smith

* Company: Virginia Colony

Title: Planter

* E-mail: Jsmith@folklore.net

5

Download DMP64_FW1x01.exe

Remember Me (Cookies must be enabled)

Note: By downloading this software you agree to our [terms and conditions](#).

Figure 68. Location of Firmware Upgrade Files on the Website

6. Click **Run** twice (see figure 69, ⑥). The PC downloads the firmware update from the Extron Web site and starts the installation program to extract the firmware file.

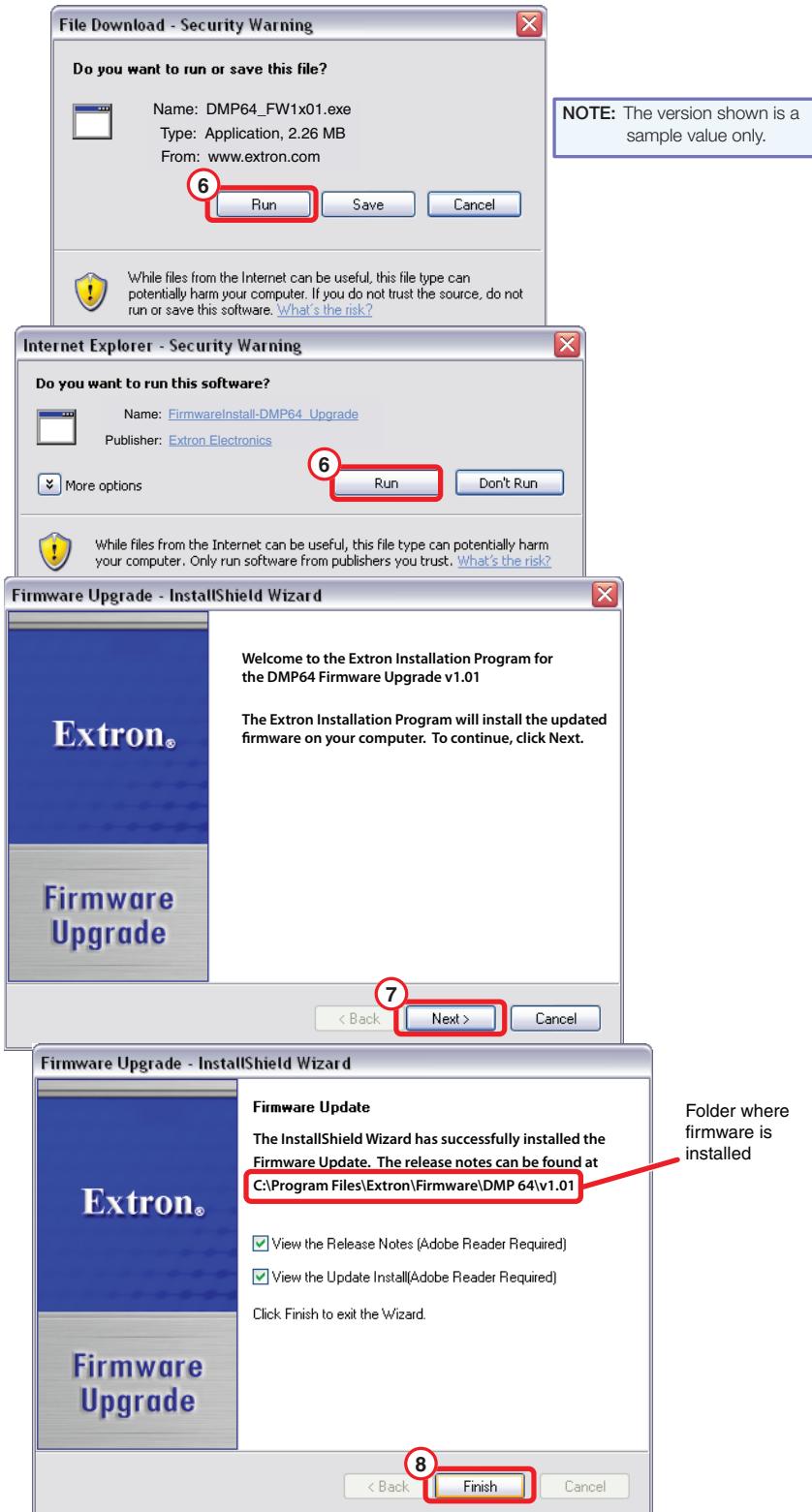


Figure 69. Downloading Firmware Upgrade Files

7. Click **Next** (see figure 69, ⑦ on previous page). The program extracts and places the firmware files in a folder identified in the InstallShield Wizard window.

NOTE: Write down the folder where the firmware file is saved.

8. Click **Finish** (see figure 69, ⑧) to exit the program.
9. Connect the PC to the device via the Ethernet port.
10. Access the device using the HTML pages (see [Download the Startup Page](#) on page 120 .)
11. Click the **Configuration** tab.
12. Click the **Firmware Upgrade** link.
13. Click the **Browse** button. An open file window appears.
14. Navigate to the folder where the firmware upgrade file was saved. Select the file.

NOTE: Valid firmware files must have the file extension ‘.S19’. Any other file extension is **not** a firmware upgrade. The original factory-installed firmware is permanently available on the device. If the attempted firmware upload fails for any reason, the device automatically reverts to the factory-installed firmware.

15. Click **Open**.
16. Click **Upload**. The firmware upload to the device can take a few minutes.

File Management Tab

File Management Page

To delete files such as HTML pages from the connected device or to upload custom files to the device, click the **File Management** tab. The device downloads the file management HTML page (see figure 70).

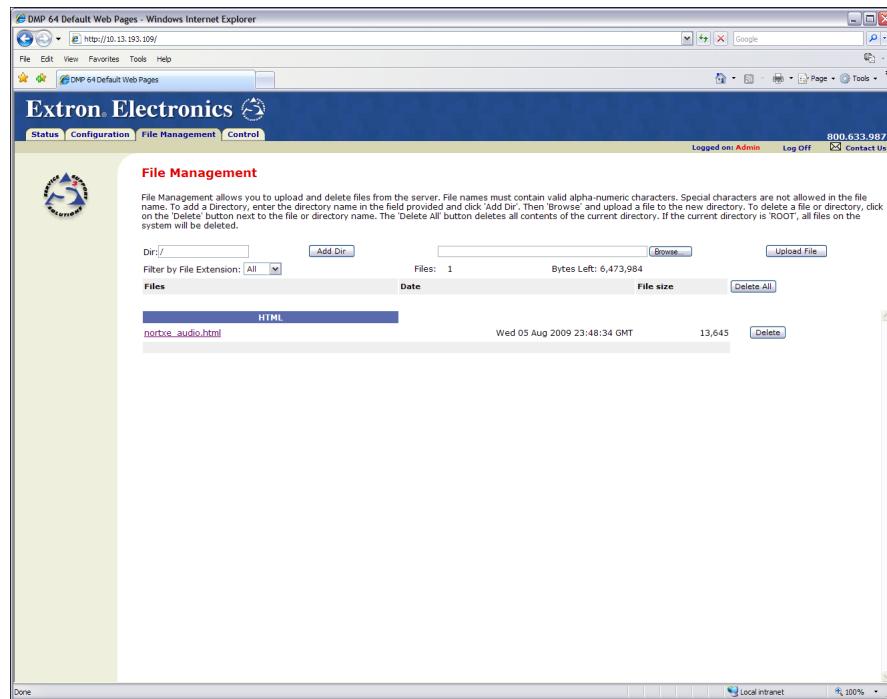


Figure 70. File Management Page

NOTE: The files listed in figure 70 are shown for example only.

To delete a file, click the **Delete** button at the right of that file.

Upload your own file as follows:

NOTE: The following characters are invalid in file names:

{space} + ~ , @ = ' [] { } < > ' " ; : > \ and ?.

1. Click the **Browse** button.
2. Browse through the system and select the desired files.

NOTE: If you want one of the pages that you create and upload to be the default startup page, name that file "index.HTML."

3. Click the **Upload File** button. The selected files appear in the list.

Control Tab

Audio Settings Page

The Audio Settings page provides a way to set the input audio gain and attenuation, output volume, and mix-point adjustments including level control, and mute or unmute. Access the Audio Settings page by clicking the **Audio Settings** link on the control page (see figure 71).

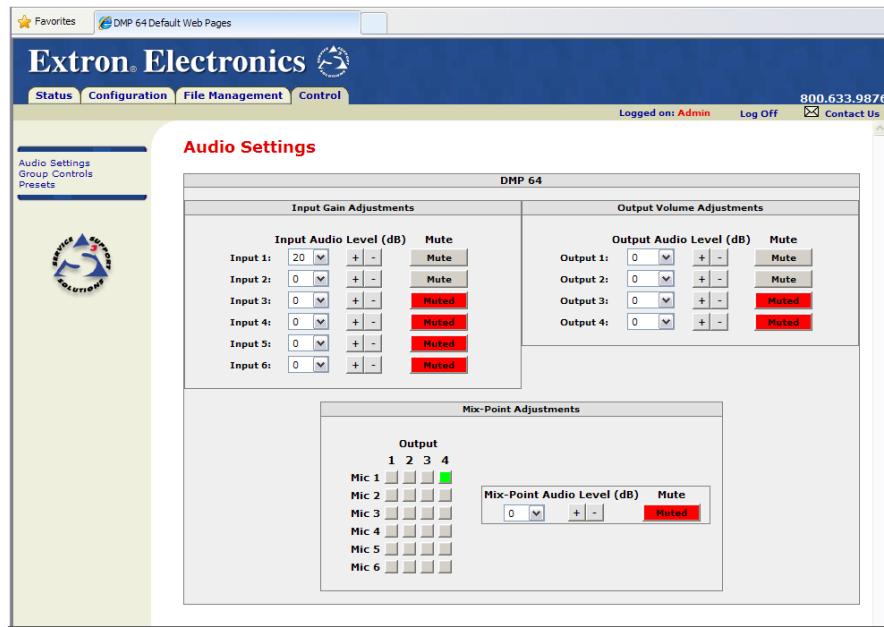


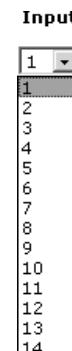
Figure 71. Video and Audio Settings Page

Change the Input Gain and Attenuation

Users can set the level of audio gain or attenuation (-18 dB to +80 dB) for each input from the audio settings page. Audio levels can be adjusted so there are no noticeable volume differences between sources.

Change an input audio level setting as follows:

1. Next to the desired input number, click the input level drop-down box. A drop-down scroll box appears (right).
2. Click and drag the slider or click the scroll up button or scroll down button until the desired audio level is visible.
Alternately the + and - boxes increment the level up or down one step.
3. Click the desired gain or attenuation value. The range is -18 to +80 dBu.
4. Repeat steps 1 through 3 for each input.



Mute and Unmute Inputs and Outputs

Pressing the **Mute** button toggles mute on or off. When muted, the **Mute** button is red and displays Muted. When unmuted it returns to gray and displays Mute.

Mute and unmute as follows:

1. Next to the desired input or output press the **Mute** button to toggle mute on (button turns red) or off (button is gray).
2. Repeat for each input and output.

Mute and Unmute the Mixer-points

The HTML mixer does not include control for the virtual bus send and receive mixer. When a mixer-point is muted (default) no signal passes. When unmuted, the signal passes from selected input to the selected output.

Mute and unmute the mixer-points as follows:

1. Click the desired input and output combination. The selected mix-point turns green. Mic 1 to Output 4 is shown selected (see figure 72).

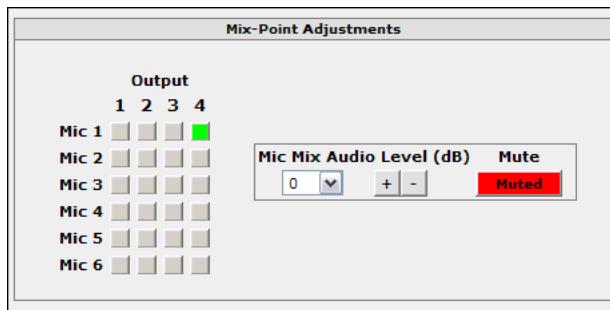


Figure 72. Output Selection Drop Box

2. Click the **Mute** button to either toggle mute on or off.
3. Repeat steps 1 and 2 for each I/O combination.

Change the Mic Mix Audio Level

To change the mix-point audio input level:

1. Click the desired input and output combination. The selected mix-point turns green. Mic 1 to Output 4 is shown selected (see figure 72).
2. Click the **Mic Mix Audio Level** drop-down box. A list box appears. Alternately the + and - boxes increment the level up or down one step.
3. Click and drag the slider or click the scroll up button or scroll down button until the desired audio level is visible.
4. Click the desired gain or attenuation value. The range is -35 to +25 dB.
5. Repeat steps 1 through 3 for each mix-point.

Change the Output Volume Level

The output volume attenuates the signal from full volume down to 0 volume in 100 steps. Users can set individual output volume levels from a range of –100 dB (full attenuation, minimum volume) to 0 dB (no attenuation, full volume).

Change an output audio level setting as follows:

1. Next to the desired output number, click the **output level** drop-down box. A drop-down list box appears.
2. Click and drag the slider or click the scroll up button or scroll down button until the desired audio level is visible.
Alternately the + and - boxes increment the level up or down one step.
3. Click the desired attenuation value. The range is –100 to 0 dB.
4. Repeat steps 1 through 3 for each output.

Group Controls Page

If group controls have been set using SIS commands or DSP Configurator, the group controls page provides access to those controls (see figure 73). Group controls cannot be set using the HTML pages. Access the group controls page by clicking the **Group Controls** link on the left of the control page.

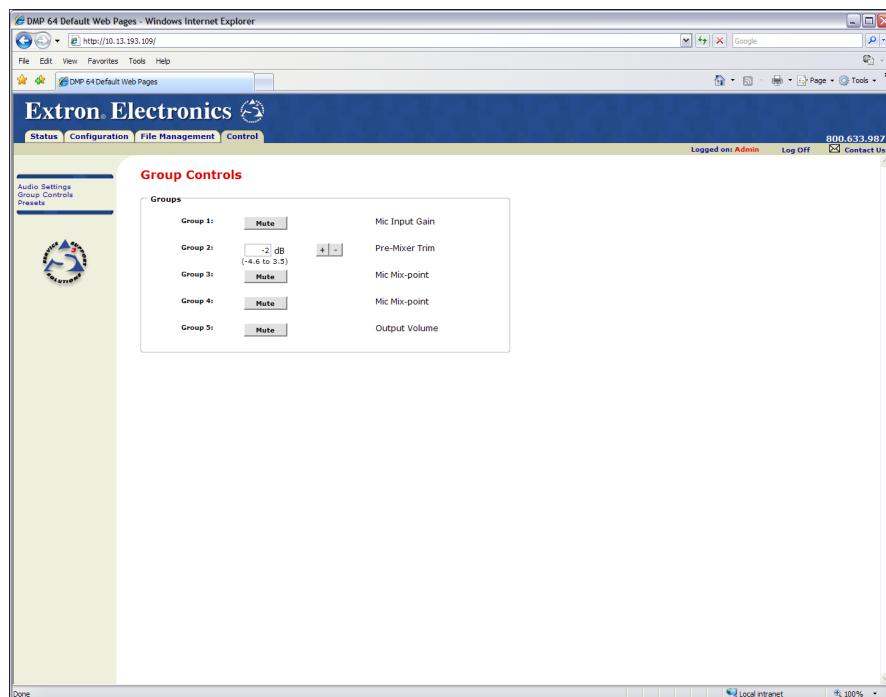


Figure 73. Group Controls Page

To adjust a group control:

- Next to the desired group control, directly input a value by clicking in the box and typing a value, or use the + and – box to increment or decrement the value by one.
- Repeat for each group.

NOTE: The range of each group control is displayed under the value box.

Presets Page

The Presets page is used to save new presets created on the Audio Settings page (see figure 74). Presets saved using this HTML page only include the gain controls on that page. However, presets created using DSP Configurator include all signal processing blocks. When saving a preset from the HTML page, be certain not to overwrite those presets created by DSP Configurator.

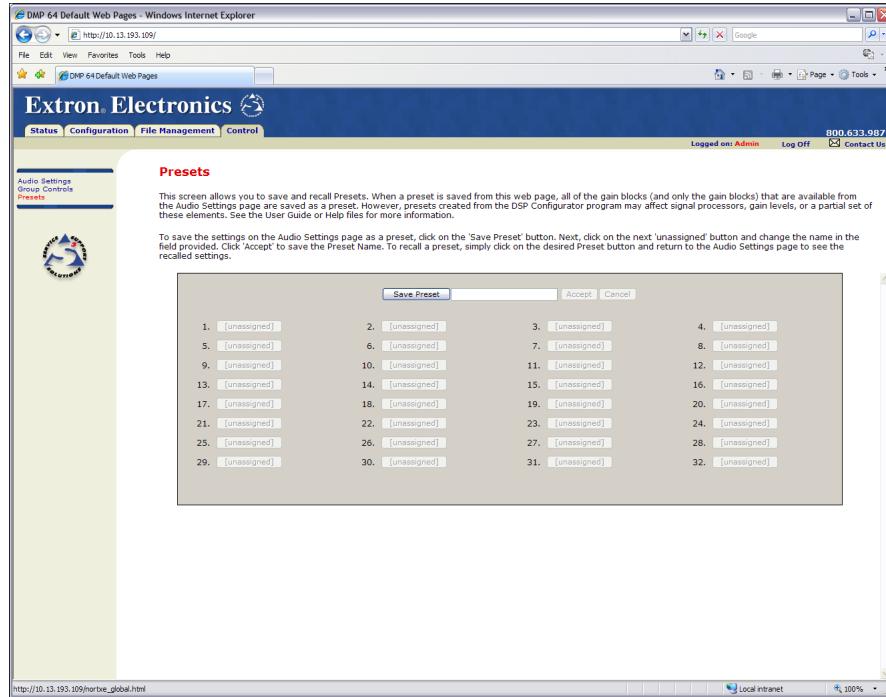


Figure 74. Presets

To recall a preset to be the current configuration, click the button associated with the desired preset.

NOTE: When presets are **recalled** using the HTML page, signal processing adjustments saved as part of the preset (under DSP Configurator control only) are recalled and overwrite the current audio settings.

When a partial preset is recalled, it overwrites that portion of the current configuration addressed by the preset, leaving the remainder unchanged.

Special Characters

The HTML language reserves certain characters for specific functions. The device will not accept these characters as part of preset names, the device name, passwords, or locally created file names.

The device rejects the following characters:

{space} + ~ , @ = ' [] { } < > ' " ; : > \ ?

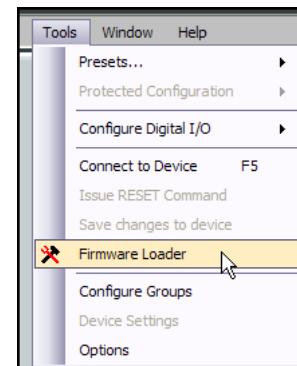
Reference Information

This section contains reference information for the DMP 64, including:

- [Firmware Loader](#)
- [DMP 64 Hardware Reset Modes](#)
- [Mounting the DMP 64](#)

Firmware Loader

The DSP Configurator program includes a firmware loader program which allows replacing the firmware without taking the DMP 64 out of service. Download the desired firmware file from the Extron website, (see [Firmware Upgrade Page](#) on page 127 for instructions.)



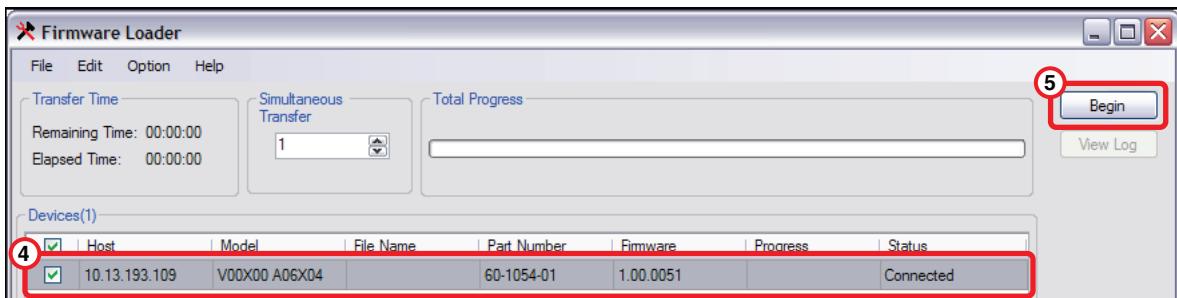
To access the firmware uploader:

1. Select **Tools>Firmware Loader**.
2. The **Add Device** dialog box appears. Type the IP address of the DMP 64, then press **OK**.

NOTE: If the IP has not been changed, the default IP address is: 192.168.255.255

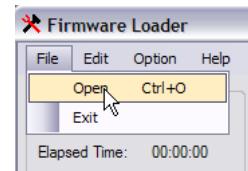


The Firmware Loader screen appears.



3. From the toolbar, select **File>Open**.
4. Locate the downloaded firmware file and click on it.
5. Click **Begin** on the main screen. The total progress bar tracks the loading progress.
6. When the upload is finished, select **File>Exit** to exit the program.

The firmware upload is complete.



DMP 64 Hardware Reset Modes

DMP 64 Reset Mode Summary			
Mode	Mode Activation	Result	Purpose/Notes
Use Factory Firmware	1 Press and hold the reset button, then apply power. NOTE: After a mode 1 reset, update the DMP 64 firmware to the latest version. DO NOT operate the firmware version that results from this mode reset.	The DMP 64 reverts to the factory default firmware. Event scripting does not start if the DMP 64 is powered on in this mode. All user files and settings (drivers, adjustments, IP settings) are maintained. NOTE: If you do not want to update the firmware, or perform a mode 1 reset by mistake, cycle power to the DMP 64 to return to the firmware version running prior to the reset.	This mode reverts to the factory default firmware version if incompatibility issues arise with user-loaded firmware. NOTE: User-defined web pages may not work correctly if using an earlier firmware version.
Run/Stop Events	3 With the power on, press and hold the reset button until the Power LED blinks once (3 sec.), then release and within 1 second press reset momentarily (<1 sec.). NOTE: Nothing happens if the momentary press does not occur within 1 second.	Mode 3 toggles events on or off. Front panel level indicators blink twice to indicate events has toggled on, or three times to indicate event logging has toggled off.	Useful for troubleshooting
Reset all IP Settings	4 With power on, press and hold the reset button for about 6 sec. until the Power LED blinks twice (once at 3 sec., again at 6 sec.), then release and within 1 second press reset momentarily (<1 sec.). NOTE: Nothing happens if the momentary press does not occur within 1 second.	Mode 4: <ul style="list-style-type: none"> Enables ARP capability. Set the IP address to default. Sets the subnet to default. Sets the gateway address to default. Sets port mapping back to default. Turns DHCP off. Turns events off. 	Enables resetting IP address information using ARP and MAC address.
Reset to Factory Defaults	5 With power on, press and hold the reset button for about 9 sec. until the Power LED blinks three times (once at 3 sec., again at 6 sec., again at 9 sec.), then release and within 1 second press Reset momentarily (<1 sec.). NOTE: Nothing happens if the momentary press does not occur within 1 second.	Mode 5 performs a complete reset to factory defaults, except for firmware: <ul style="list-style-type: none"> Does everything mode 4 reset does. All mix-points muted and set to 0 dB. All outputs unmuted and set to 0 dB. DSP Processing returned to defaults and bypassed. All inputs muted and set to 0 dB. All presets and group master memory cleared. 	Useful to start over with configuration or uploading, or to restart events.

Mounting the DMP 64

The 1U high, half rack width, 9.5 inch deep DMP 64 Digital Matrix Processor can be:

- Set on a table,
- Mounted on a rack shelf,
- Mounted under a desk or tabletop, or
- Mounted on a projector bracket.

Tabletop Use

The DMP 64 includes rubber feet (not installed). For tabletop use, attach a self-adhesive rubber foot to each corner of the bottom of the unit.

UL Rack Mounting Guidelines

The following Underwriters Laboratories (UL) guidelines pertain to the safe installation of the DMP 64 in a rack.

1. **Elevated operating ambient temperature** — If installed in a closed or multi-unit rack assembly, the operating ambient temperature of the rack environment can be greater than room ambient temperature. Therefore, install the unit in an environment compatible with the maximum ambient temperature ($T_{ma} = +122^{\circ}\text{F}, +50^{\circ}\text{C}$) specified by Extron.
2. **Reduced air flow** — Install the equipment in a rack so that the amount of air flow required for safe operation of the equipment is not compromised.
3. **Mechanical loading** — Mount the equipment in the rack so that a hazardous condition is not achieved due to uneven mechanical loading.
4. **Circuit overloading** — Connect the equipment to the supply circuit and consider the effect that circuit overloading might have on overcurrent protection and supply wiring. Appropriate consideration of equipment nameplate ratings should be used when addressing this concern.
5. **Reliable earthing (grounding)** — Maintain reliable grounding of rack-mounted equipment. Pay particular attention to supply connections other than direct connections to the branch circuit (e.g. use of power strips).

Rack Mounting

For optional rack mounting, do not install the rubber feet. Mount the DMP 64 on a 19 inch Universal 1U or Basic rack shelf. Follow the instructions included with the rack mount kit to install the DMP 64.

Furniture Mounting

Furniture mount the DMP 64 using the optional mounting kit (Extron MBU 125, part number **70-077-01**, under-desk, or Extron MBD 129, part number **70-077-02**, through-desk). Follow the instructions included with the mounting kit to install the DMP 64.

Table or Wall Mounting

The MBU 125 table or wall mounting brackets extend approximately 1/4 inch (6.4 mm) above the top surface of the enclosure. This design allows for an air space between the enclosure and the surface to which it is mounted. Follow the instructions included with the mounting kit to install the DMP 64.

Extron Warranty

Extron Electronics warrants this product against defects in materials and workmanship for a period of three years from the date of purchase. In the event of malfunction during the warranty period attributable directly to faulty workmanship and/or materials, Extron Electronics will, at its option, repair or replace said products or components, to whatever extent it shall deem necessary to restore said product to proper operating condition, provided that it is returned within the warranty period, with proof of purchase and description of malfunction to:

USA, Canada, South America, and Central America:

Extron Electronics
1230 South Lewis Street
Anaheim, CA 92805
U.S.A.

Europe and Africa:

Extron Europe
Hanzeboulevard 10
3825 PH Amersfoort
The Netherlands

Asia:

Extron Asia Pte Ltd
135 Joo Seng Road, #04-01
PM Industrial Bldg.
Singapore 368363
Singapore

Japan:

Extron Electronics, Japan
Kyodo Building, 16 Ichibancho
Chiyoda-ku, Tokyo 102-0082
Japan

China:

Extron China
686 Ronghua Road
Songjiang District
Shanghai 201611
China

Middle East:

Extron Middle East
Dubai Airport Free Zone
F12, PO Box 293666
United Arab Emirates, Dubai

This Limited Warranty does not apply if the fault has been caused by misuse, improper handling care, electrical or mechanical abuse, abnormal operating conditions, or if modifications were made to the product that were not authorized by Extron.

NOTE: If a product is defective, please call Extron and ask for an Application Engineer to receive an RA (Return Authorization) number. This will begin the repair process.

USA: 714.491.1500 or 800.633.9876
Asia: 65.6383.4400

Europe: 31.33.453.4040
Japan: 81.3.3511.7655

Units must be returned insured, with shipping charges prepaid. If not insured, you assume the risk of loss or damage during shipment. Returned units must include the serial number and a description of the problem, as well as the name of the person to contact in case there are any questions.

Extron Electronics makes no further warranties either expressed or implied with respect to the product and its quality, performance, merchantability, or fitness for any particular use. In no event will Extron Electronics be liable for direct, indirect, or consequential damages resulting from any defect in this product even if Extron Electronics has been advised of such damage.

Please note that laws vary from state to state and country to country, and that some provisions of this warranty may not apply to you.

Extron Headquarters +1.800.633.9876 (Inside USA/Canada Only) Extron USA - West +1.714.491.1500 +1.714.491.1517 FAX	Extron Europe +800.3987.6673 (Inside Europe Only) Extron USA - East +1.919.850.1000 +1.919.850.1001 FAX	Extron Asia 800.3987.6673 (Inside Asia Only) +81.3.3511.7655 +81.3.3511.7656 FAX +31.33.453.4040 +31.33.453.4050 FAX	Extron Japan +81.3.3511.7655 +81.3.3511.7656 FAX +65.6383.4400 +65.6383.4664 FAX	Extron China +4000.398766 (Inside China Only) +86.21.3760.1568 +86.21.3760.1566 FAX	Extron Middle East +971.4.299.1800 +971.4.299.1880 FAX	Extron Korea +82.2.3444.1571 +82.2.3444.1575 FAX	Extron India 1800.3070.3777 (Inside India Only) +91.80.3055.3777 +91.80.3055.3737 FAX
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