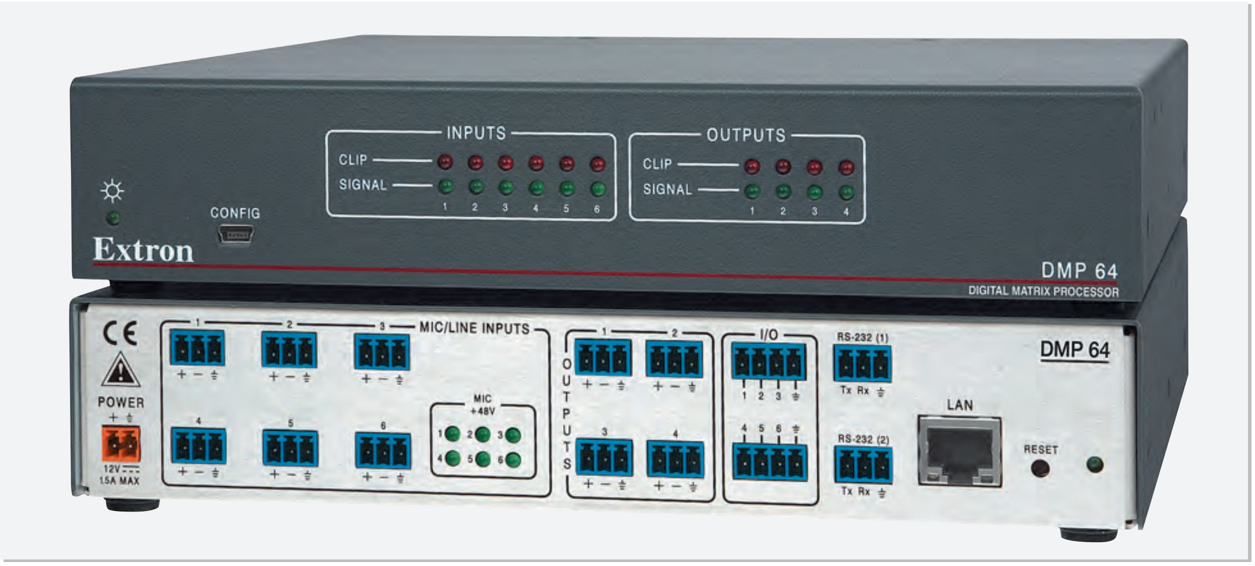
User Manual

Audio Products
Mixers and Processors

DMP 64

Digital Matrix Processor



Extron® Electronics
INTERFACING, SWITCHING AND CONTROL

68-1790-01 Rev. B
04 10

Safety Instructions • English

This symbol is intended to alert the user of important operating and maintenance (servicing) instructions in the literature provided with the equipment.

This symbol 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.

Caution

Read Instructions • Read and understand all safety and operating instructions before using the equipment.

Retain Instructions • The safety instructions should be kept for future reference.

Follow Warnings • Follow all warnings and instructions marked on the equipment or in the user information.

Avoid Attachments • Do not use tools or attachments that are not recommended by the equipment manufacturer because they may be hazardous.

Warning

Power sources • This equipment should be operated only from the power source indicated on the product. This equipment is intended to be used with a main power system with a grounded (neutral) conductor. The third (grounding) pin is a safety feature, do not attempt to bypass or disable it.

Power disconnection • To remove power from the equipment safely, remove all power cords from the rear of the equipment, or the desktop power module (if detachable), or from the power source receptacle (wall plug).

Power cord protection • Power cords should be routed so that they are not likely to be stepped on or pinched by items placed upon or against them.

Servicing • Refer all servicing to qualified service personnel. There are no user-serviceable parts inside. To prevent the risk of shock, do not attempt to service this equipment yourself because opening or removing covers may expose you to dangerous voltage or other hazards.

Slots and openings • If the equipment has slots or holes in the enclosure, these are provided to prevent overheating of sensitive components inside. These openings must never be blocked by other objects.

Lithium battery • There is a danger of explosion if battery is incorrectly replaced. Replace it only with the same or equivalent type recommended by the manufacturer. Dispose of used batteries according to the manufacturer's instructions.

Consignes de Sécurité • Français

Ce symbole sert à avertir l'utilisateur que la documentation fournie avec le matériel contient des instructions importantes concernant l'exploitation et la maintenance (réparation).

Ce symbole sert à avertir l'utilisateur de la présence dans le boîtier de l'appareil de tensions dangereuses non isolées posant des risques d'électrocution.

Attention

Lire les instructions• Prendre connaissance de toutes les consignes de sécurité et d'exploitation avant d'utiliser le matériel.

Conserver les instructions• Ranger les consignes de sécurité afin de pouvoir les consulter à l'avenir.

Respecter les avertissements • Observer tous les avertissements et consignes marqués sur le matériel ou présentés dans la documentation utilisateur.

Eviter les pièces de fixation • Ne pas utiliser de pièces de fixation ni d'outils non recommandés par le fabricant du matériel car cela risquerait de poser certains dangers.

Avertissement

Alimentations • Ne faire fonctionner ce matériel qu'avec la source d'alimentation indiquée sur l'appareil. Ce matériel doit être utilisé avec une alimentation principale comportant un fil de terre (neutre). Le troisième contact (de mise à la terre) constitue un dispositif de sécurité : n'essayez pas de la contourner ni de la désactiver.

Déconnexion de l'alimentation• Pour mettre le matériel hors tension sans danger, déconnectez tous les cordons d'alimentation de l'arrière de l'appareil ou du module d'alimentation de bureau (s'il est amovible) ou encore de la prise secteur.

Protection du cordon d'alimentation • Acheminer les cordons d'alimentation de manière à ce que personne ne risque de marcher dessus et à ce qu'ils ne soient pas écrasés ou pincés par des objets.

Réparation-maintenance • Faire exécuter toutes les interventions de réparation-maintenance par un technicien qualifié. Aucun des éléments internes ne peut être réparé par l'utilisateur. Afin d'éviter tout danger d'électrocution, l'utilisateur ne doit pas essayer de procéder lui-même à ces opérations car l'ouverture ou le retrait des couvercles risquent de l'exposer à de hautes tensions et autres dangers.

Fentes et orifices • Si le boîtier de l'appareil comporte des fentes ou des orifices, ceux-ci servent à empêcher les composants internes sensibles de surchauffer. Ces ouvertures ne doivent jamais être bloquées par des objets.

Lithium Batterie • Il a danger d'explosion s'll y a remplacment incorrect de la batterie. Remplacer uniquement avec une batterie du meme type ou d'un ype equivalent recommande par le constructeur. Mettre au reut les batteries usagees conformement aux instructions du fabricant.

Sicherheitsanleitungen • Deutsch

Dieses Symbol soll dem Benutzer in der im Lieferumfang enthaltenen Dokumentation besonders wichtige Hinweise zur Bedienung und Wartung (Instandhaltung) geben.

Dieses Symbol soll den Benutzer darauf aufmerksam machen, daß im Inneren des Gehäuses dieses Produktes gefährliche Spannungen, die nicht isoliert sind und die einen elektrischen Schock verursachen können, herrschen.

Achtung

Lesen der Anleitungen • Bevor Sie das Gerät zum ersten Mal verwenden, sollten Sie alle Sicherheits-und Bedienungsanleitungen genau durchlesen und verstehen.

Aufbewahren der Anleitungen • Die Hinweise zur elektrischen Sicherheit des Produktes sollten Sie aufbewahren, damit Sie im Bedarfsfall darauf zurückgreifen können.

Befolgen der Warnhinweise • Befolgen Sie alle Warnhinweise und Anleitungen auf dem Gerät oder in der Benutzerdokumentation.

Keine Zusatzgeräte • Verwenden Sie keine Werkzeuge oder Zusatzgeräte, die nicht ausdrücklich vom Hersteller empfohlen wurden, da diese eine Gefahrenquelle darstellen können.

Vorsicht

Stromquellen • Dieses Gerät sollte nur über die auf dem Produkt angegebene Stromquelle betrieben werden. Dieses Gerät wurde für eine Verwendung mit einer Hauptstromleitung mit einem geerdeten (neutralen) Leiter konzipiert. Der dritte Kontakt ist für einen Erdanschluß, und stellt eine Sicherheitsfunktion dar. Diese sollte nicht umgangen oder außer Betrieb gesetzt werden.

Stromunterbrechung • Um das Gerät auf sichere Weise vom Netz zu trennen, sollten Sie alle Netzkabel aus der Rückseite des Gerätes, aus der externen Stomversorgung (falls dies möglich ist) oder aus der Wandsteckdose ziehen.

Schutz des Netzkabels • Netzkabel sollten stets so verlegt werden, daß sie nicht im Weg liegen und niemand darauf treten kann oder Objekte darauf- oder unmittelbar dagegengestellt werden können.

Wartung • Alle Wartungsmaßnahmen sollten nur von qualifiziertem Servicepersonal durchgeführt werden. Die internen Komponenten des Gerätes sind wartungsfrei. Zur Vermeidung eines elektrischen Schocks versuchen Sie in keinem Fall, dieses Gerät selbst öffnen, da beim Entfernen der Abdeckungen die Gefahr eines elektrischen Schlags und/oder andere Gefahren bestehen.

Schlitze und Öffnungen • Wenn das Gerät Schlitze oder Löcher im Gehäuse aufweist, dienen diese zur Vermeidung einer Überhitzung der empfindlichen Teile im Inneren. Diese Öffnungen dürfen niemals von anderen Objekten blockiert werden.

Litium-Batterie • Explosionsgefahr, falls die Batterie nicht richtig ersetzt wird. Ersetzen Sie verbrauchte Batterien nur durch den gleichen oder einen vergleichbaren Batterietyp, der auch vom Hersteller empfohlen wird. Entsorgen Sie verbrauchte Batterien bitte gemäß den Herstelleranweisungen.

Instrucciones de seguridad • Español

Este símbolo se utiliza para advertir al usuario sobre instrucciones importantes de operación y mantenimiento (o cambio de partes) que se desean destacar en el contenido de la documentación suministrada con los equipos.

Este símbolo se utiliza para advertir al usuario sobre la presencia de elementos con voltaje peligroso sin protección aislante, que puedan encontrarse dentro de la caja o alojamiento del producto, y que puedan representar riesgo de electrocución.

Precaucion

Leer las instrucciones • Leer y analizar todas las instrucciones de operación y seguridad, antes de usar el equipo.

Conservar las instrucciones • Conservar las instrucciones de seguridad para futura consulta.

Obedecer las advertencias • Todas las advertencias e instrucciones marcadas en el equipo o en la documentación del usuario, deben ser obedecidas.

Evitar el uso de accesorios • No usar herramientas o accesorios que no sean especificamente recomendados por el fabricante, ya que podrian implicar riesgos.

Advertencia

Alimentación eléctrica • Este equipo debe conectarse únicamente a la fuente/tipo de alimentación eléctrica indicada en el mismo. La alimentación eléctrica de este equipo debe provenir de un sistema de distribución general con conductor neutro a tierra. La tercera pata (puesta a tierra) es una medida de seguridad, no puentearia ni eliminaria.

Desconexión de alimentación eléctrica • Para desconectar con seguridad la acometida de alimentación eléctrica al equipo, desenchufar todos los cables de alimentación en el panel trasero del equipo, o desenchufar el módulo de alimentación (si fuera independiente), o desenchufar el cable del receptáculo de la pared.

Protección del cables de alimentación • Los cables de alimentación eléctrica se deben instalar en lugares donde no sean pisados ni apretados por objetos que se puedan apoyar sobre ellos.

Reparaciones/mantenimiento • Solicitar siempre los servicios técnicos de personal calificado. En el interior no hay partes a las que el usuario deba acceder. Para evitar riesgo de electrocución, no intentar personalmente la reparación/mantenimiento de este equipo, ya que al abrir o extraer las tapas puede quedar expuesto a voltajes peligrosos u otros riesgos.

Ranuras y aberturas • Si el equipo posee ranuras o orificios en su caja/alojamiento, es para evitar el sobrecalientamiento de componentes internos sensibles. Estas aberturas nunca se deben obstruir con otros objetos.

Batería de litio • Existe riesgo de explosión si esta batería se coloca en la posición incorrecta. Cambiar esta batería únicamente con el mismo tipo (o su equivalente) recomendado por el fabricante. Desachar las baterías usadas siguiendo las instrucciones del fabricante.

安全须知 • 中文

这个符号提示用户该设备用户手册中有重要的操作和维护说明。

这个符号警告用户该设备机壳内有暴露的危险电压，有触电危险。

注意

阅读说明书 • 用户使用该设备前必须阅读并理解所有安全和使用说明。

保存说明书 • 用户应保存安全说明书以备将来使用。

遵守警告 • 用户应遵守产品和用户指南上的所有安全和操作说明。

避免追加 • 不要使用该产品厂商没有推荐的工具或追加设备，以避免危险。

警告

电源 • 该设备只能使用产品上标明的电源。 设备必须使用有地线的供电系统供电。 第三条线（地线）是安全设施，不能不用或跳过 。

拔掉电源 • 为安全地从设备拔掉电源，请拔掉所有设备后或桌面电源的电源线，或任何接到市电系统的电源线。

电源线保护 • 妥善布线， 避免被踩踏，或重物挤压。

维护 • 所有维修必须由认证的维修人员进行。 设备内部没有用户可以更换的零件。为避免出现触电危险不要自己试图打开设备盖子维修该设备。

通风孔 • 有些设备机壳上有通风槽或孔，它们是用来防止机内敏感元件过热。 不要用任何东西挡住通风孔。

锂电池 • 不正确的更换电池会有爆炸的危险。必须使用与厂家推荐的相同或相近型号的电池。按照生产厂的建议处理废弃电池。

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. Operation is subject to the following two conditions:

1. This device may not cause harmful interference.
2. This device must accept any interference received, including interference that may cause undesired operation.

The Class A limits are designed to 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 harmful interference, in which case the user will be required to correct the interference at his own expense.

NOTE: This unit was tested with shielded cables on the peripheral devices. Shielded cables must be used with the unit to ensure compliance with FCC emissions limits.

Contents

Introduction

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

- **About This Manual**
- **About the DMP 64 Digital Matrix Processor**
- **Features**

About This Manual

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

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

Notes, Cautions, Tips, and Warnings

NOTE: Notes call attention to information that may be of special importance.

CAUTION: Indicates a potential hazard to equipment or data exists and is included to help prevent damage.

TIP: Tech Tips provide technical information that may be helpful during installation, in performing a procedure or adjustment, or operation of the DMP 64.

WARNING: Indicates a potential hazard to personal safety exists.

About the DMP 64 Digital Matrix Processor

The DMP 64 is a standalone audio matrix processor with 6 microphone/line inputs and 4 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 of DSP processors and mixing matrixes is performed using the Extron DSP Configurator program from a host computer via any of the communication ports, RS-232, USB or Ethernet (high-speed ports recommended). 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, and then recalled through the DSP Configurator or a control system using 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.

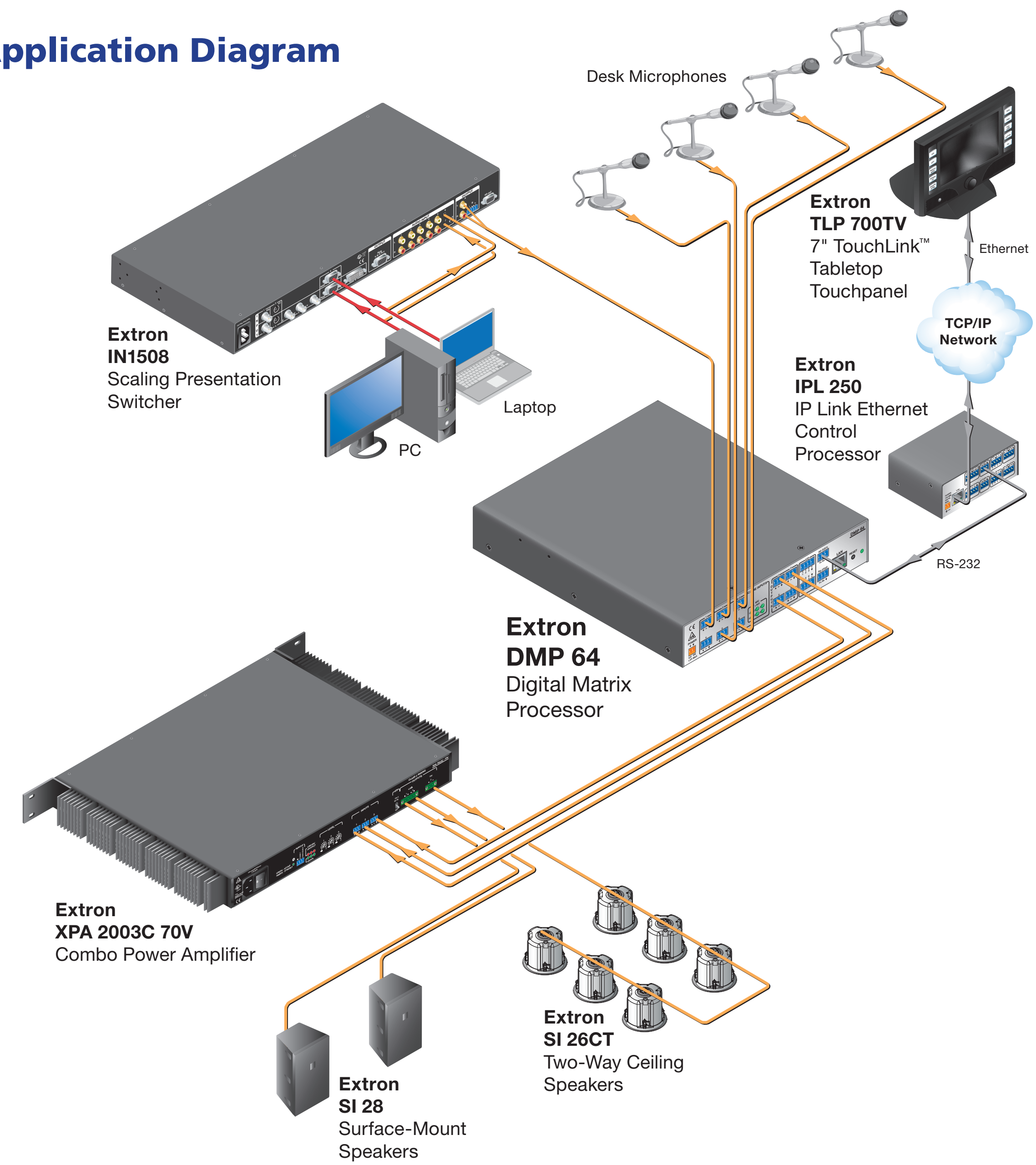
Features

- **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 32/64-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.
- **Two RS-232 ports —** The DMP 64 is equipped with both primary and secondary RS-232 serial ports for divided room applications.
- **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 SIS™ - Simple Instruction Set 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.

- **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.
- **Internal 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.

Tabletop Use

Each DMP 64 comes with 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 may be greater than room ambient temperature. Therefore, install the unit in an environment compatible with the maximum ambient temperature (Tma = +122 °F, +50 °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" Universal 1U or Basic rack shelf (Extron RSU 129, part #**60-190-01**; or Extron RSB 129, part #**60-604-02**).

To rack mount the DMP 64:

1. If rubber feet were previously installed on the bottom of the DMP 64, remove them.
2. Mount the DMP 64 on the rack shelf, using two 4-40 x 3/16 inch screws in opposite (diagonal) corners to secure the unit to the shelf.

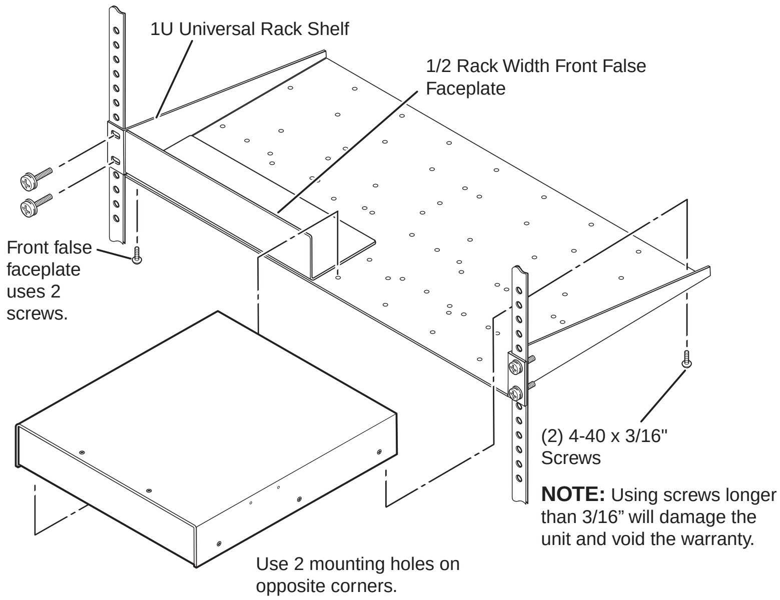


Figure 1. **Mounting the DMP 64 on a Universal Rack Shelf**

3. Install blank panel(s) or other unit(s) on the rack shelf.

Furniture Mounting

Furniture mount the DMP 64 using the optional mounting kit (Extron MBU 125, part #**70-077-01**, under-desk; or Extron MBD 129, part #**70-077-02**, through-desk) as follows:

1. Attach the selected mounting brackets with the machine screws provided.
2. If feet were previously installed on the bottom of the cabinet, remove them.

Table or Wall Mounting

The 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.

Table or wall mount the DMP 64 as follows:

1. Hold the unit with the attached brackets against the underside of the table or other furniture, or against the wall. Mark the location of the screw holes of the bracket on the mounting surface.
2. Drill 3/32 inch (2 mm) diameter pilot holes, 1/4 inch (6.4 mm) deep in the mounting surface at the marked screw locations.
3. Insert #8 wood screws into the four pilot holes. Tighten each screw into the mounting surface until just less than 1/4 inch of the screw's head protrudes.
4. Align the mounting screws with the slots in the brackets and place the unit against the surface, with the screws through the bracket slots.
5. Slide the unit slightly forward or back, then tighten all four screws to secure it in place.

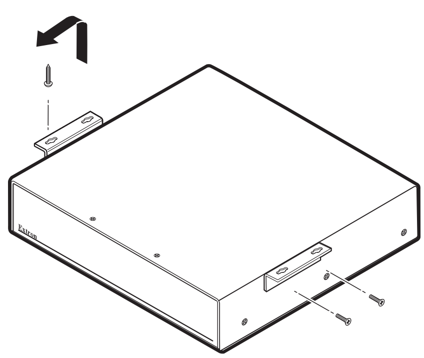


Figure 2. **MBU 125, Under Desk Mounting**

Through-desk Mounting

Mount the DMP 64 through a desk or podium as follows:

1. Cut the proper sized hole in the mounting surface.
2. Hold the DMP 64 with the attached brackets against the underside of the table or other furniture. Mark the location of the screw holes of the bracket on the mounting surface.
3. Drill 3/32 inch (2 mm) diameter pilot holes, 1/4 inch (6.3 mm) deep in the mounting surface at the marked screw locations.
4. Insert four #8 wood screws through the bracket and into the four pilot holes. Tighten all four screws to secure the unit in place.

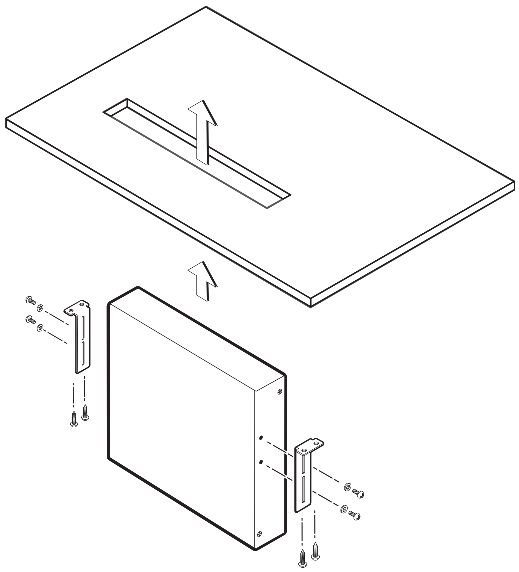


Figure 3. **MBD 129, Through-desk Mounting**

Rear Panel Features and Cabling

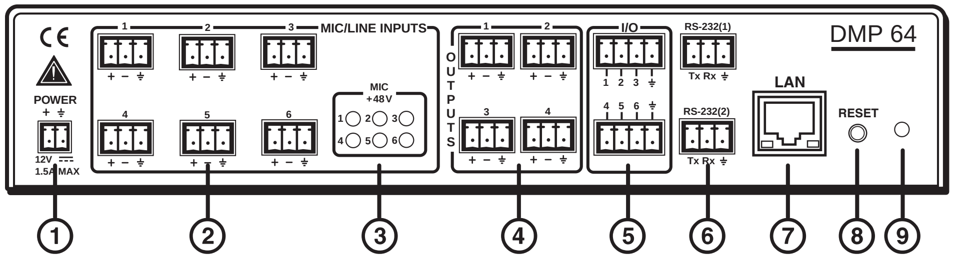


Figure 4. DMP 64 Rear Panel

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.

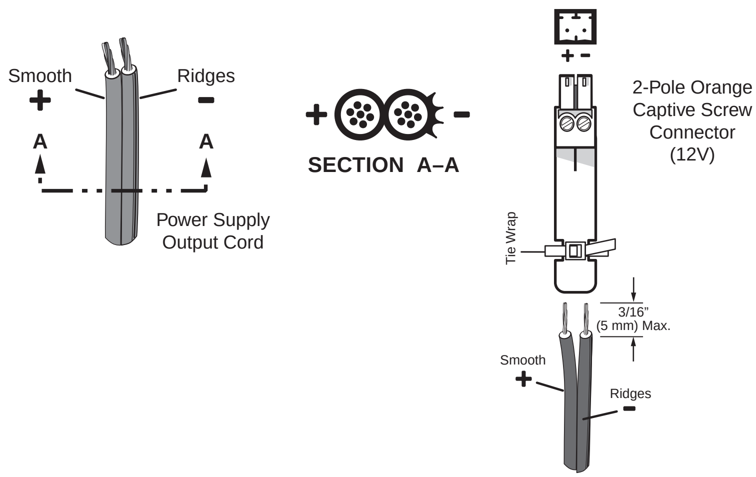


Figure 5. Power Supply Wiring

CAUTION: 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.

CAUTION: 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.

WARNING: The two power cord wires must be kept separate while the power supply is plugged in. Remove power before wiring.

NOTE: To verify the polarity before connection, check the no load power supply output with a voltmeter.

CAUTION: 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.

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

> **Note:** 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 may be connected to these inputs. See the following diagram for wiring instructions.

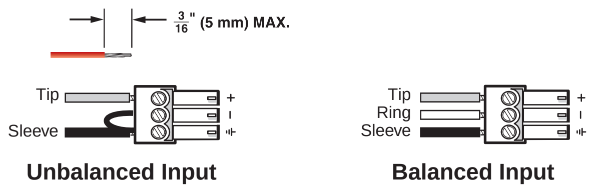


Figure 6. Wiring Balanced or Unbalanced Mic and Line Inputs

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.

> **Note:** Condenser microphones require phantom power. Dynamic microphones **do not** require power.

> **Caution:** Never set an unbalanced dynamic microphone to **48 V**. Doing so may 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.

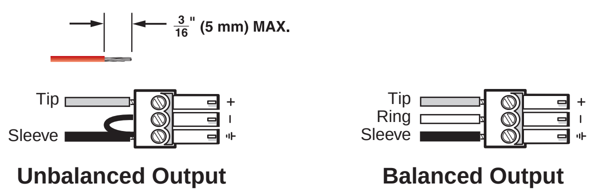


Figure 7. Output Connector Wiring

> **Caution:** Connect the sleeve to ground (Gnd). Connecting the sleeve only to a negative (-) terminal will damage the audio output circuits.

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, temperature sensors, etc.

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** for additional information.

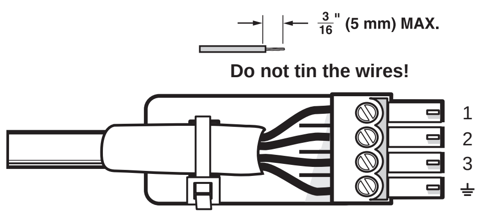


Figure 8. **Digital I/O Wiring**

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.

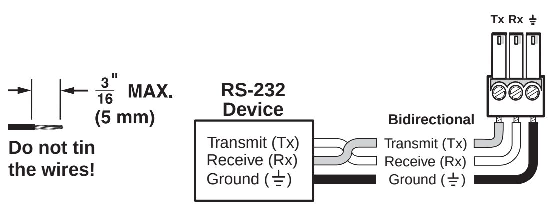


Figure 9. **RS-232 Wiring**

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 additional information on Ethernet cabling in chapter 4, "SIS Programming and Control".

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**, for additional details.

USB Configuration port (front panel)

A front panel configuration port uses an Extron USB A Male to USB Mini B Male Configuration Cable, **26-654-06** for connection to a PC computer via the USB port. See **Install the USB Driver** for USB driver installation details.

Operation

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

- **DMP 64 Operation**
- **Front Panel Operation**
- **Rear Panel Operation**
- **DSP Processing/Signal Flow**
- **Processor Blocks**
- **Mic/Line Input Channel**
- **Virtual Bus Returns**
- **Primary Mix Matrix**
- **Secondary Mix Matrix**
- **Line Output Channels**

DMP 64 Operation

The DMP 64 does not have physical controls. Configuration and operation are accomplished using a PC running Windows XP or better 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 SIS™ Simple Instruction Set using hyper-terminal or DataViewer.

There are several front and rear panel operational indicators outlined in the following pages.

Front Panel Operation

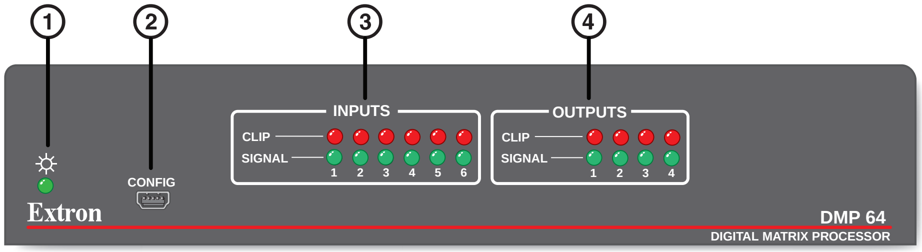


Figure 10. DMP 64 Front Panel

① **Power LED** — 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** for details.

The DMP 64 appears as a USB peripheral with bi-directional communication. The USB connection can be used for software operation (**Windows-based Program Control**), and SIS control, (**Software Control**).

③ **Input Indicators** — Stacked red (signal clipping) and green (signal present) LEDs for inputs 1 – 6 . Each stack 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 – 4 . Each LED stack 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.

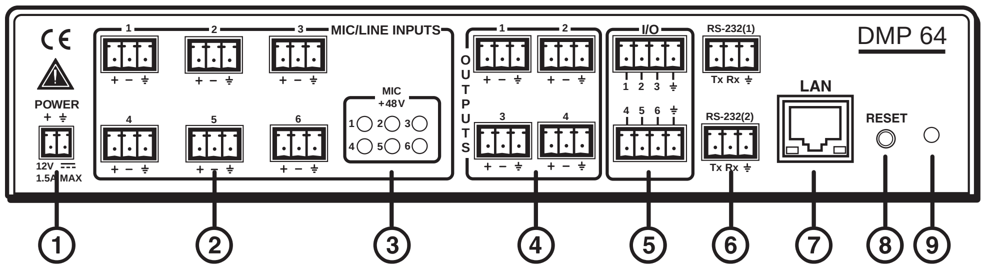


Figure 11. **DMP 64 Rear Panel**

Rear Panel Operation

① ② ④ ⑤ ⑥ See **Rear Panel Features and Cabling** for details.

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

> **NOTE:** Condenser microphones require phantom power. Dynamic microphones **do not** require power.
> When a line level source is connected, be certain the +48V phantom power is off (unchecked).

> **CAUTION:** Never set an unbalanced dynamic microphone to **+48V**. Doing so may damage the microphone. For condenser mics, verify the mic will safely operate at +48 VDC.

⑦ **LAN** — The LAN connector has a green LED 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** later in this chapter for additional information.

⑨ **Power/Reset LED** — The green LED indicator adjacent to the reset button duplicates the front panel LED operation. See **Reset Actuator and LED** later in this chapter for additional information.

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 power indicator LED flashes 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**), using the **Firmware Loader** in the DSP Configurator program (see **DMP Software**), or using the Extron standalone Firmware Loader software application available on the included disc or at www.extron.com.

Reset Actuator and LED

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

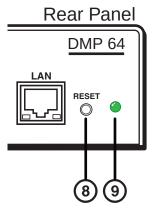


Figure 12. **Reset button and LED**

Hardware Reset Modes:

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

With power on, when the reset button is held down, every three seconds the rear panel LED will pulse (blink). At the first blink Mode 3 is available, at the second blink Mode 4 is available and the third blink indicates Mode 5 is available. The reset modes have separate and distinct functions outlined below. Additional information is available in **DMP 64 Hardware Reset Modes**.

MODE 1 — Firmware reset: Disconnect power to the DMP 64. Press and hold the reset button while applying power to return the firmware to the version shipped with the unit from the factory. Event scripting will not start when powered on in this mode. This allows recovering a unit with incorrect or corrupt firmware.

All user files and settings are maintained. Some user web pages may not work correctly if returning the unit to an earlier firmware release.

MODE 3 — Events reset: With power on, press and hold the reset button until the reset LED blinks once (~3 seconds). Release the reset button, then within one (1) second press it again to toggle events On or Off, depending on the current state.
If the event logging is currently stopped, following the momentary (<1 sec.) press, the reset LED will flash twice indicating events logging has begun.

If the events are currently running, i.e., digital I/O scripts, following the momentary (<1 sec.) press, the reset LED will flash three times indicating the events logging has stopped.

Each flash will last for 0.25 seconds. If the second momentary press does not occur within 1 second Mode 3 is exited.

MODE 4 — IP Address reset: With power on, press and hold the reset button about 6 seconds until the reset LED blinks twice. Release the reset button, then within 1 second, press it again to reset the IP settings.

Mode 4 will:

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

If a second momentary press does not occur within 1 second, the reset will be ignored.

MODE 5 — Factory default reset: With power on, press and hold the reset button until the reset LED blinks 3 times (~9 seconds). Release then momentarily (<1 second) press the reset button to return the DMP 64 to factory default conditions. If the second momentary press does not occur within 1 second, the reset is exited.

The default (reset) state of the device is:

- All mix-points are 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 four-pin Digital I/O ports are used to monitor or drive TTL level digital signals. The ports consist of two banks of three I/Os with the 4th pin used as a ground providing six ports total. The DSP Configurator software provides selection of a script from a list, to 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** for additional information.

DSP Processing/Signal Flow

The diagram below shows the signal flow and DSP processing per signal chain. Signal chains and matrixes are described in the following sections.

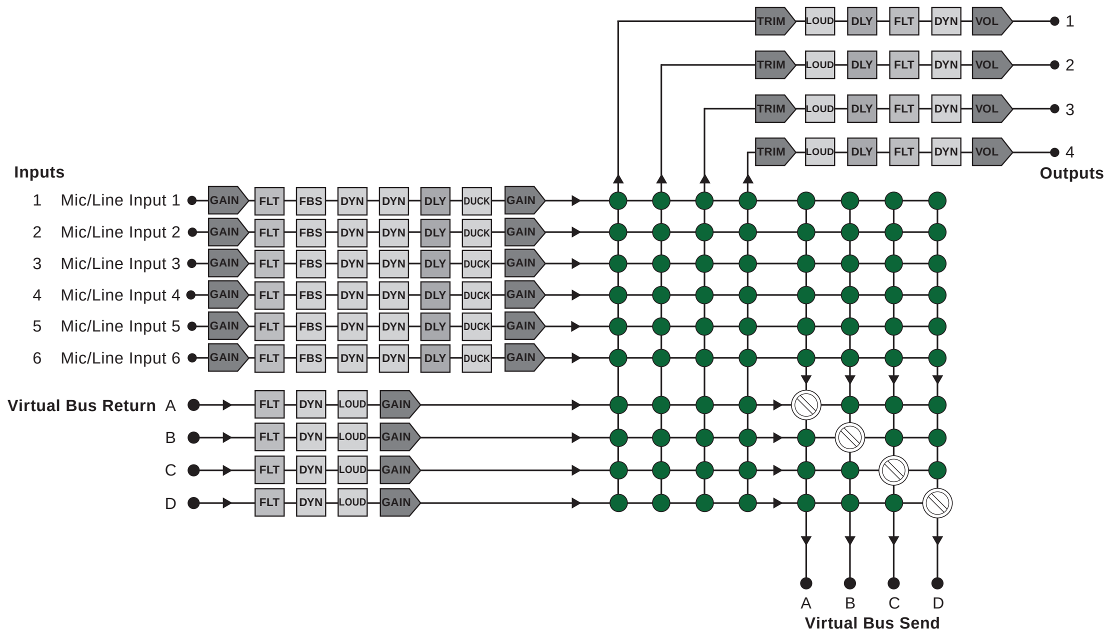


Figure 13. **DMP 64 DSP Signal Flow Block Diagram**

All signal routing, processing, and level control (gain/trim/volume), are accomplished using software control from a PC connected to the DMP 64 via the USB configuration port, the LAN connection (recommended) or one of the RS-232 ports. The DSP Configurator program provides complete control while the internal web pages and SIS commands provide more limited control.

This chapter describes the signal processing including parameter ranges, and how to mix inputs and outputs using the DSP Configurator control program. Detailed information for configuring the processor blocks, mix-points and level controls are found in **DMP Software**.

Mic/Line Input Signal Chain

Input signal chain GUI elements from left to right are as follows:

- **Gain (GAIN) –** mono gain control with a range from -18 to 80 dB includes mute button. A polarity switch (+/-) and phantom power on/off is provided, and a meter displays post-fader audio level in dBFS.
- **Filter (FILT) –** up to five frequency filters can be inserted in any combination of High Pass, Low Pass, Bass & Treble shelving, or Parametric Equalizer.
- **Feedback Suppressor (FBS) –** Provides 15 dynamic filters and 5 fixed filters for effective feedback control (primarily of microphone inputs).
- **Dynamics 1 (DYN) –** Dynamics processors vary the dynamic level, (the range of loudest to softest signals). Choices include AGC, compressor, limiter, or noise gate.
- **Dynamics 2 (DYN) –** a second dynamics processor can be inserted from a choice of AGC, compressor, limiter, or noise gate.

- **Delay (DLY) –** channel delay can be set by feet or meters modified by an air temperature parameter, or by direct time insertion from 0 - 200 ms.
- **Ducking (DUCK) –** mic ducking can be set up using the ducking dialog box.
- **Gain (GAIN) –** Pre-mixer matrix gain control with a range of -100 to +12 dB.

Virtual Bus Returns

Virtual return elements from left to right are as follows:

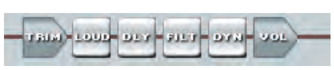
- **Filter (FILT) –** up to three frequency filters can be inserted in any combination of High Pass, Low Pass, Bass & Treble shelving, or Parametric EQ.
- **Dynamics (DYN) –** a single dynamics processor can be inserted from a choice of AGC, compressor, limiter, or noise gate.
- **Loudness (LOUD) –** adjusts the signal to compensate for changes in human perception at the high and low end of the hearing curve using a predetermined filter curve in response to the sum of the virtual return gain and the Calibration Adjustment slider value.
- **Gain (GAIN) –** mono gain control with a range from -100 to +12 dB includes mute button.

Mix Matrixes

The dual matrix contains 76 mix-points, each containing a single fader with a range of -35 to +25 dB, plus a mute control. The 76 mix-points are grouped into two matrixes. Dual matrix features are as follows:

- **Primary Matrix –** a mic/line input section and a virtual bus return section feed the primary matrix, which routes incoming signals to the line outputs.
- **Secondary Matrix –** a mic/line input section and a virtual bus return section feed the secondary matrix, which routes to the virtual bus sends.

Line Outputs



Line Output elements from left to right are as follows:

- **Trim (TRIM) –** Post-mixer mono trim control with a range of -12 to +12 dB
- **Loudness (LOUD) –** adjusts the signal to a predetermined filter curve in response to the sum of the line output Volume value and the Calibration Adjustment slider value to compensate for changes in human perception at the high and low end of the hearing curve.
- **Delay (DLY) –** channel delay can be set by feet or meters modified by an air temperature parameter, or by direct time insertion from 0 - 200 ms.
- **Filter (FILT) –** up to nine frequency filters can be inserted in any combination of High Pass, Low Pass, tone (Bass & Treble shelving), or parametric.
- **Dynamics (DYN) –** a single dynamics processor can be inserted from a choice of AGC, compressor, limiter, or noise gate.
- **Volume (VOL) –** a mono fader adjusts output volume from -100 to 0 dB and includes a mute control. A polarity switch (+/-) is provided and a meter displaying output audio level in dBFS.

Processor Blocks

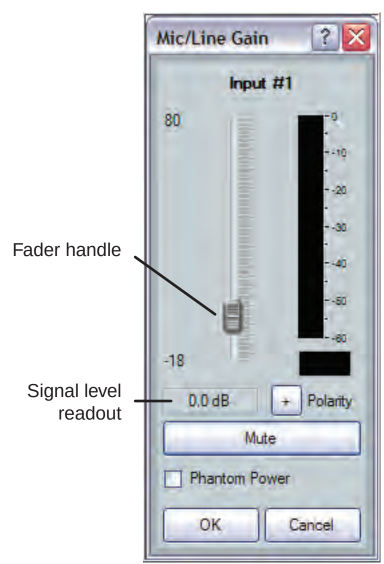
Processor blocks are placed in the signal chain to perform specific tasks. There are level control blocks, signal processor blocks and mix-point matrix blocks (with level control). Level control processors do not have to be inserted, they are always active. The following sections provide details of navigation, menus, and other interface operation. The processor blocks, while performing different functions, have several common elements.

- **Insert —** All blocks (except level controls) may be inserted by right-clicking on the desired box and selecting from the context menu or by double-clicking and making a selection.
- **Remove a process —** Active processors can be removed by right-clicking on the box and selecting **Delete** or by selecting the block and pressing delete on the keyboard. This sets the parameters back to default and bypasses the block. An active processor may be replaced by right-clicking and inserting a new processor. A warning will appear to indicate the previous processor is about to be replaced.
- **Mute —** When a level block (gain, trim or volume) is muted, all signal flow is blocked. When mute is active a red mark appears in the lower left of the block. Mix-point mute is indicated by shadowing the mix-point.
- **Bypass —** When bypass is active signal flow passes through the block without processing, regardless of the settings. When bypass is removed, the signal will be processed according to the parameter settings. A red mark appears in the lower left of the block (shown below) to indicate it has been inserted, but is currently bypassed.

Mic/Line Input Channel

There are six (6) mono mic/line input channels. Channel controls and processing blocks described in the following sections are identical for each of the six inputs.

Mic/Line (Input) Gain

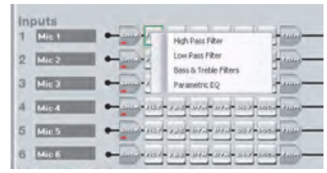


Mic/line inputs provide gain settings to accommodate consumer and pro line level sources, plus microphone level sources. Each input channel gain block provides a mono long-throw fader for gain and attenuation. Range for the control is –18 to +80 dB. Adjustments are made using the slider or by entering the desired dB level directly into the indicator box.

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:

- Direct adjustment. Click and hold the fader handle, then drag it to desired level in 0.1 dB steps.
- Click or tab to fader handle, then up/down arrow to desired level in 1 dB steps. Page Up/Page Down increases/decreases level in 10 dB steps.
- Click in or tab to the level readout field. Type a new value, then press <Enter> or <Tab> to another area.

Filter



Each mic/line input channel 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 via a context window/processor list when the block is right-clicked. After the processor is inserted, a double-click opens the setup dialog box.

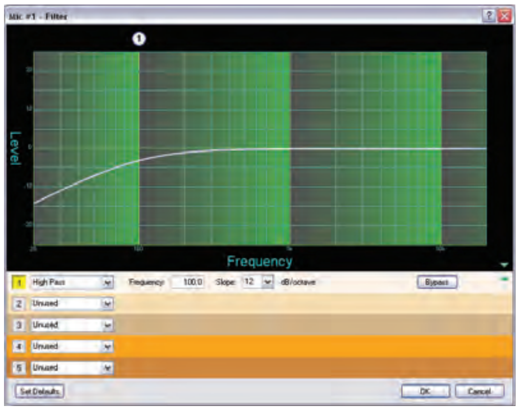


Figure 14. **Filter Block Dialog Box**

Additional filters are inserted by opening the filter block dialog box, then selecting a filter type from the drop-down filter selection list. All filter parameters are modified via the Filter block dialog box. Each filter loads with all applicable default parameters displayed to the right of each drop-down filter selection list.

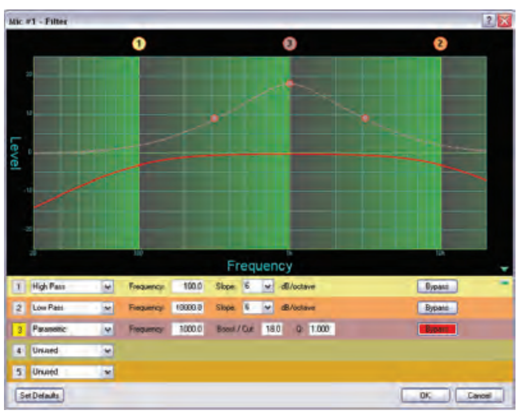


Figure 15. **Filter Dialog Box, Filters Added**

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 in figure 15 is highlighted in yellow, indicating it is the filter in focus. 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 bypassed. When active (not bypassed), the line is solid.

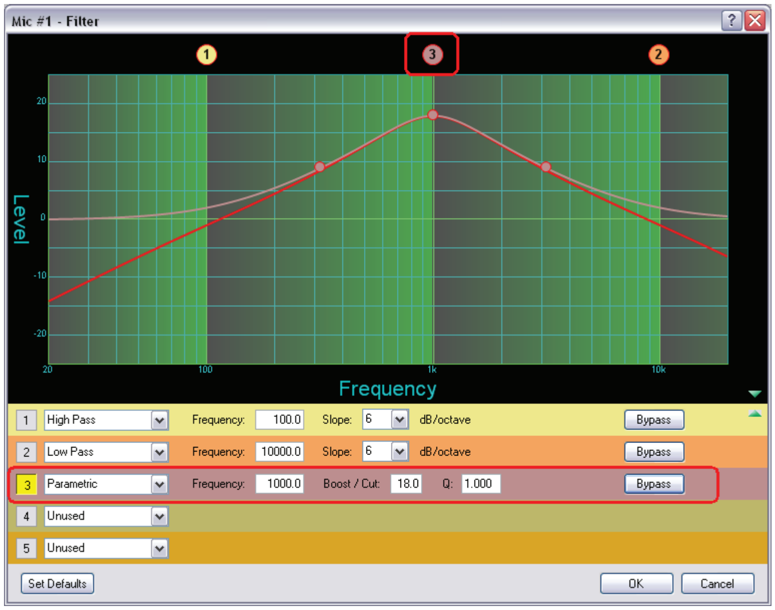


Figure 16. **Filter Dialog Box, Filter Not Bypassed**

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. The composite response of all filters is displayed in red.

Above the graph, each filter has a "handle" (circled in red above) placed directly above the cutoff or center frequency whose number corresponds to the filter number (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 possible 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	100.0 Hz	Boost/Cut: 0.0 dB	Slope: 6 dB
Treble	8000.0 Hz	Boost/Cut: 0.0 dB	Slope: 6 dB

Filter Parameter	Settings Range
Frequency	20 Hz to 20 kHz
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)

Parametric (Equalizer)

Up to five parametric filters can be placed in the filter box at one time. Each may be set to a different frequency creating a 5 band parametric equalizer. The control will boost or cut the center frequency, and by changing the Q value, the range of affected frequencies can be widened or narrowed around the center frequency. In general, the higher the Q, the narrower the affected bandwidth.

To demonstrate how Q affects the filter, see the following filter block (figure 18) 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 200 Hz on the left and 5000 Hz on the right, a bandwidth of about 4800 Hz.

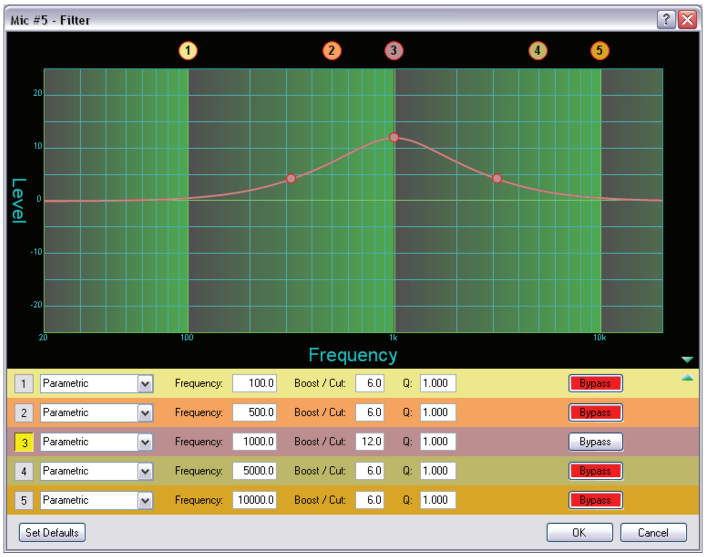


Figure 17. **Parametric Filter Dialog Box, 1000 Hz**

The above dialog box 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.

Figure 18. **All Parametric Filters Active**

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

The parametric filter allows frequency selection accurate to 0.1 Hz and either 6 or 12 dB of slope. Notice 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 will change.

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.

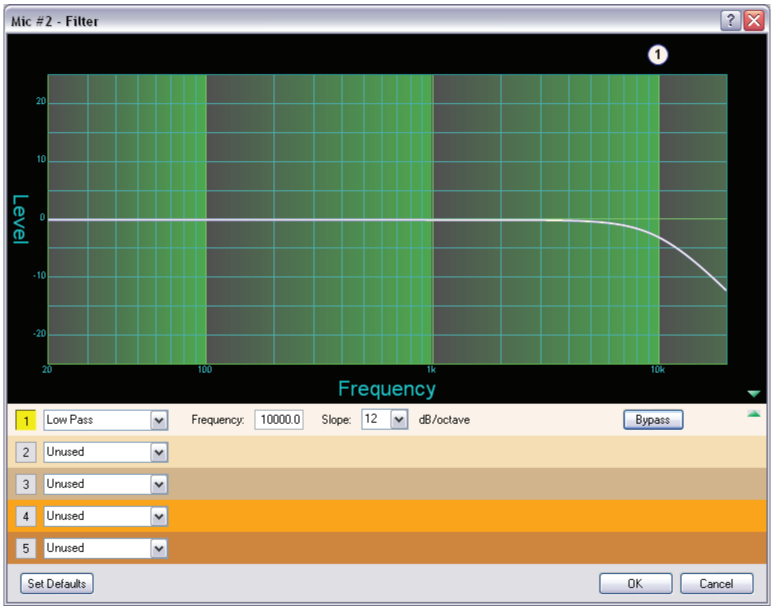


Figure 19. Low Pass Filter Response Curve

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 may be added to the filter. Adding this filter automatically inserts both a bass and treble control row in the dialog box. 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.

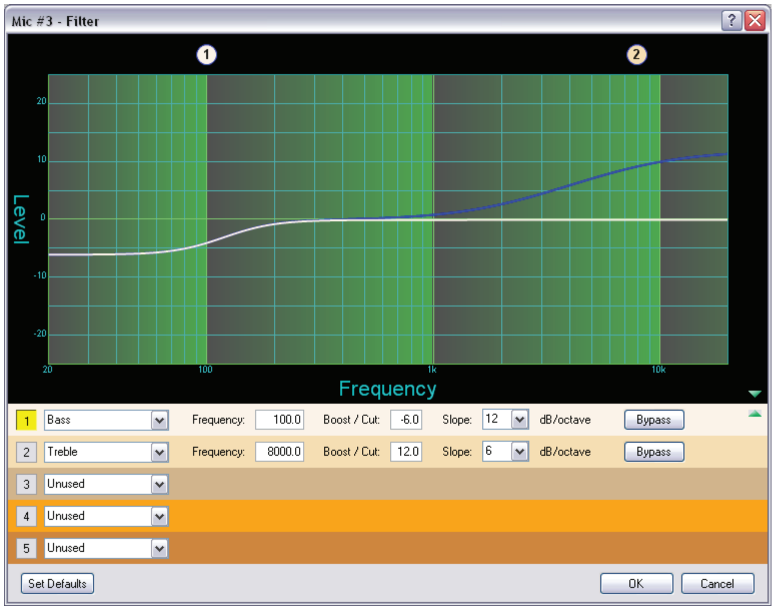


Figure 20. **Bass and Treble Shelving**

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

Feedback Suppressor

The Feedback Suppressor (FBS) is used when there is indication of feedback in live situations. Dynamic filters automatically detect feedback on a live microphone channel, and engage a set of up to five fixed and 15 dynamic filters to counteract the frequency peaks at the detected feedback frequency. Up to 15 separate filters may be employed at any time. The 15 filters act in a FIFO, or first in, first out rotation. If all 15 filters are being employed, when an additional feedback frequency is detected it will overwrite 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 3 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 for dynamic filters:

- **Clear All —** clears all dynamic filter settings.
- **Lock —** locks the dynamic filters to 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.

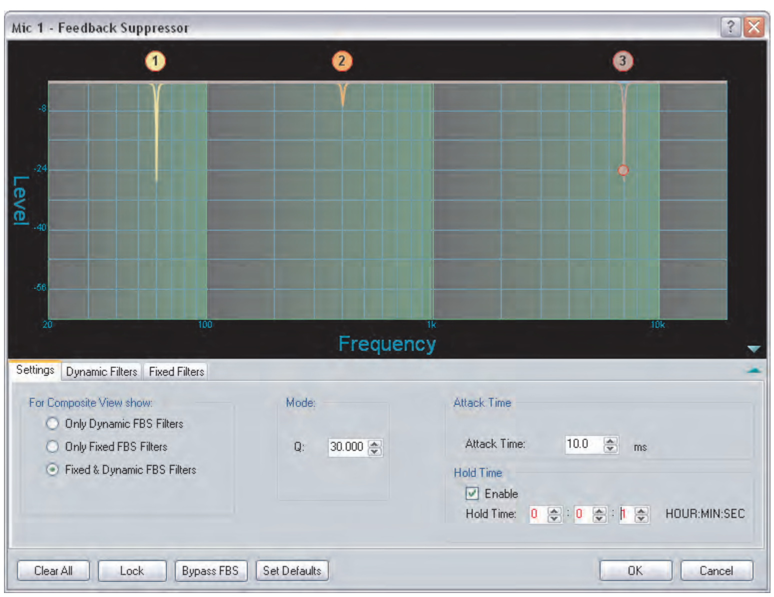


Figure 21. Feedback Suppressor

FBS Settings Tab

The settings tab enables selection of the feedback suppressor parameters.

- **For Composite View show —** content of the graph view is set by one of three radio 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 Q on the parametric equalizers, Q 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 may provide greater feedback suppression but with more possible impact to the tonal response of the source audio.
  Suggested values for specific applications are:
    - Q=7 (Voice with considerable feedback potential)
    - Q=30 (Voice with less feedback potential)
    - Q=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 occurs and when it 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).

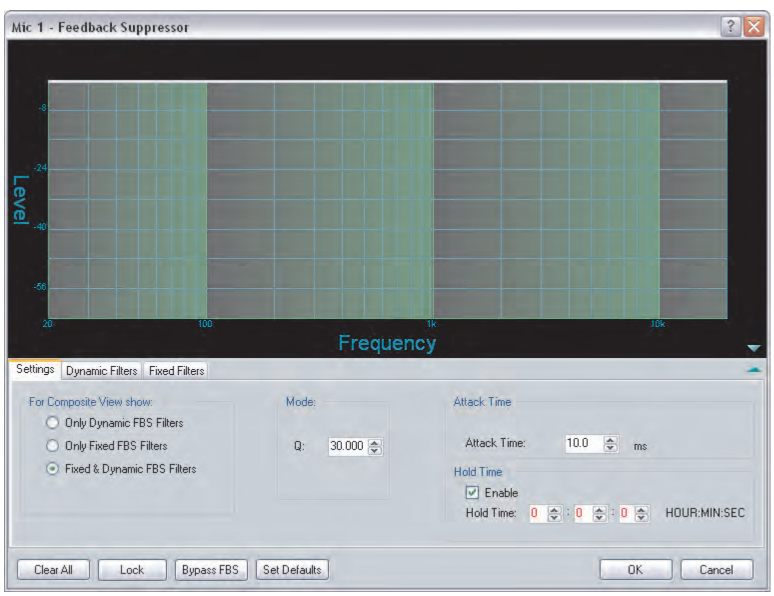


Figure 22. FBS Settings Tab

FBS Dynamic Filters Tab

This tab contains the fifteen dynamic filters, with a scroll bar to display filters hidden due to window 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 will clear all dynamic filters.

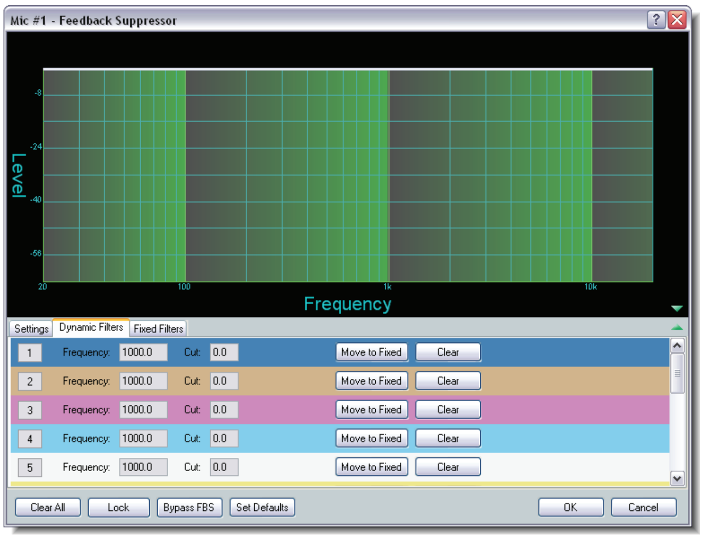


Figure 23. **FBS Dynamic Filters Tab**

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 may be engaged. The lock mode of operation is temporary, and is intended to be used during setup of the FBS. When the FBS dialog window is closed, lock mode is automatically disengaged.

If there are specific dynamic filters the operator 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 will automatically clear that frequency from the dynamic filter.

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

FBS Fixed Filters Tab

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 manually made from the Fixed Filters tab.

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

Figure 24. 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 will display **Notch** as the filter type and display the parameters copied from the dynamic filter. Once a filter is active as a fixed filter, settings can be modified or adjusted if needed. Fixed filters can also be individually bypassed using the **Bypass** button.

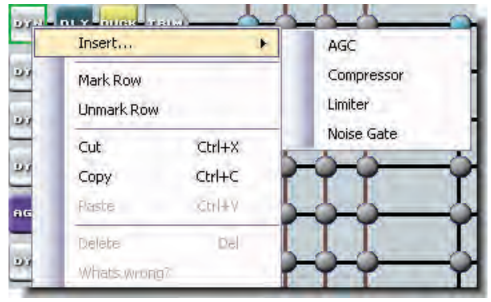
FBS Settings Ranges and Fixed Filter Defaults

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 sec. 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

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 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.

All parameters are displayed in a text box and have a resolution to 0.1 increments. Parameters can be set by direct entry in the text box to replace existing text, then pressing Enter or tabbing/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/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/right arrow keys (Page Up/Down keys adjust in increments of 10 ms).

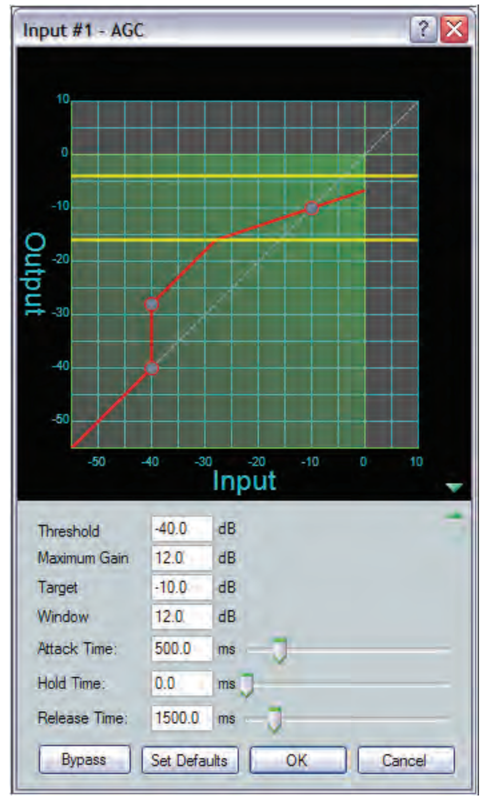
The table below lists each dynamics processor type, parameters, and factory default settings.

Parameter	AGC	Compressor	Limiter	Gate
Threshold	-40.0 dB	-30.0 dB	-10.0 dB	-65.0 dB
Max Gain	12.0 dB	x	x	x
Target	-10.0 dB	x	x	x
Window	12.0 dB	x	x	x
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	x	2.0 :1	x	20.0 :1
Hold Time	0.0 ms	100.0 ms	50.0 ms	300.0 ms
Max. Attenuation	x	x	x	25.0 dB
Soft Knee	x	Off	Off	x

Details of the individual dynamics blocks follow.

AGC (Automatic gain control)

AGC adjusts the gain level of a signal based upon the input strength to achieve a more consistent volume. Above a set threshold, weaker signals receive more gain to reach a user-defined target level. Signals stronger than the target receive gain reduction to reduce the signal towards the target.



Threshold — is the input level where maximum gain will be applied (after the attack time is exceeded). From the graph follow the input level (on the X-axis) at -40 dB up to where the 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 (Maximum Gain).
Threshold level can be adjusted from -80.0 to 0.0 dB in 0.1 dB increments.
Default is -40.0 dB.

Maximum Gain — is the highest amplification applied to a signal exceeding the threshold and up to the lower limit of the Window (see below).
Maximum Gain can be set from 0.0 dB to +60 dB in 0.1 dB increments.
Default is 12.0 dB.

Target — is 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.
Target level can be adjusted from -40 dB to 0.0 dB in 0.1 dB increments.
Default is -10.0 dB.

Window — is a specified range above and below the target level. When the signal reaches the lower limit of the window, gain control begins scaling in a linear fashion toward the target level to achieve smoother results. The window range can be set in 0.1 dB increments from 0.0 dB to 20.0 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.

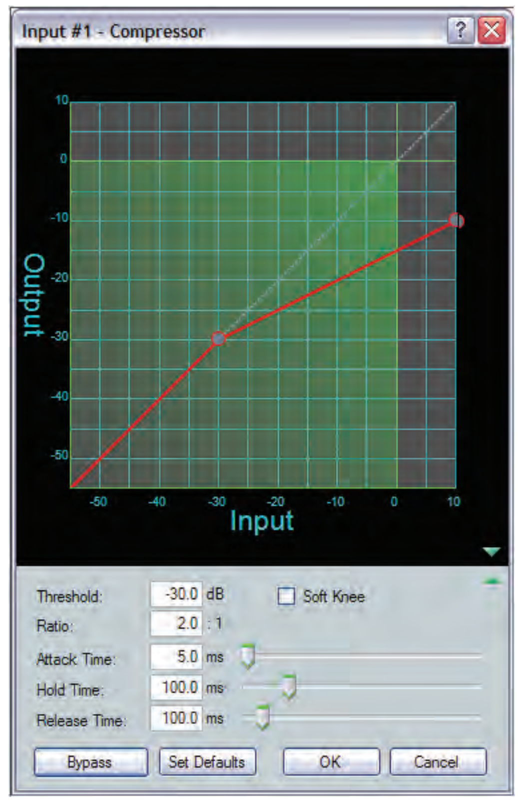
Release Time — adjusts the time it takes to return the signal to normal (unprocessed) levels after the signal no longer exceeds the threshold level setting. Release time begins only after hold time is reached.
Release time can be adjusted from 10.0 to 10000.0 ms in 0.1 ms increments.
Default is 1500.0 ms.

Compressor

The compressor regulates signal level by reducing 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.



Threshold — is the input signal level above which compression 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 -30.0 dB.

Ratio — is 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.

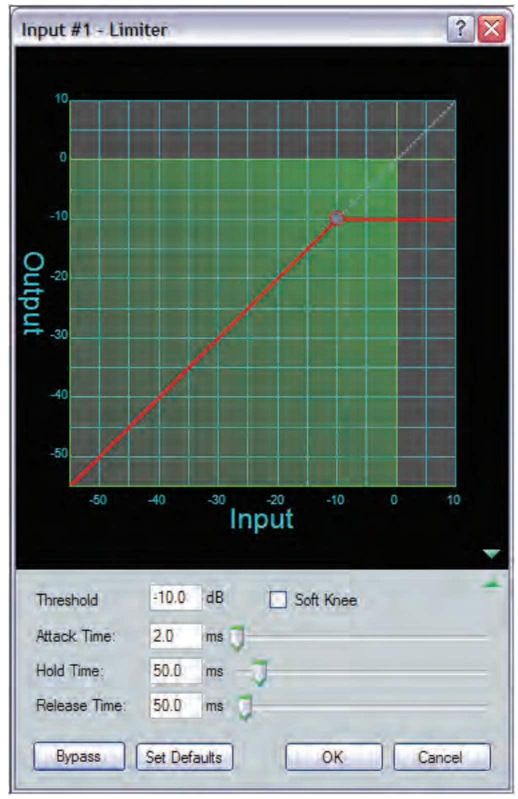
Hold Time — adjusts how long compression continues after the input signal drops below the threshold and before release time begins.
Hold time can be adjusted from 0.0 to 500.0 ms in 0.1 ms increments.
Default is 100.0 ms.

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 — Click 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 ∞:1. The ratio is fixed and cannot be changed.



Threshold — is 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.

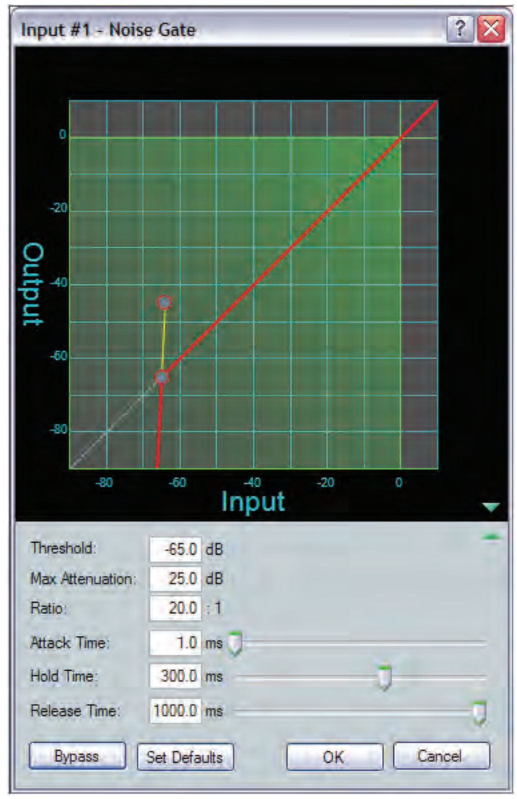
Hold Time — adjusts how long limiting continues after the input signal drops below the threshold and before release time begins.
Hold time can be adjusted from 0.0 to 500.0 ms in 0.1 ms increments.
Default is 50.0 ms.

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 — Click 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. There is no processing of the signal above the threshold. Above the threshold level, the signal passes, 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.



Threshold — is the input signal level below which gating begins (subject to attack time) and above which gating 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 -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 — is the input signal level reduction when gating is engaged.
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 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 300.0 ms.

Release Time — adjusts the time it takes to return the signal to normal (unprocessed) levels after the signal is no longer below 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 1000.0 ms.

Delay

Audio Delay is used to sync audio to video or to time-align speakers that are placed at different distances from the listener. The DMP 64 can set delay by one of three criteria: time, feet, or meters. The default unit 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** or tabbing/clicking away from the field.

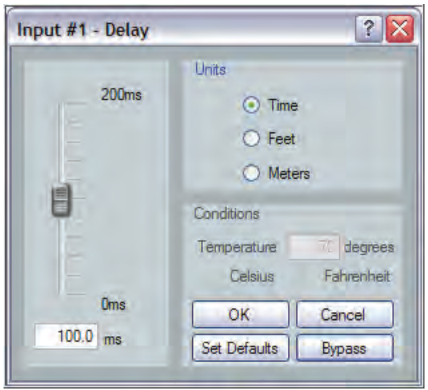


Figure 25. Delay Dialog

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 the time in 0.1 ms increments 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° Fahrenheit.

NOTE: Set a temperature value first, then set the distance.

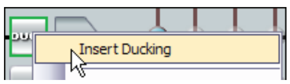
Ducking

Ducking provides a means to **duck**, or lower the level of one or more input signal targets when a specified source must take precedence. Ducking lasts for the duration of the ducking source signal (plus hold and release time) and restores the duck target(s) original level once the ducking source signal has ceased.

Ducking may be useful when:

- Program material needs to attenuate in order to accentuate the voice of a narrator,
- One microphone is used by a chairman or master of ceremonies, and needs to have priority over other mics and/or program material, or
- A paging mic must attenuate all other signals.

Ducking processor blocks are individually inserted from a context menu as shown below. Only a ducking source needs to be inserted. Ducking targets are enabled from the Ducker Configuration dialog.



Ducking is configured in a window which opens when an active ducking processor block is double-clicked (see figure 26 below). Ducking can be globally set up from a single configuration window, which opens when any of the active ducking processor blocks are double-clicked. When a ducking processor block is inserted, it is automatically set to **Enable Mic/Line Source**. All inactive ducking processor blocks have **Enable Mic/Line Source** unchecked by default.

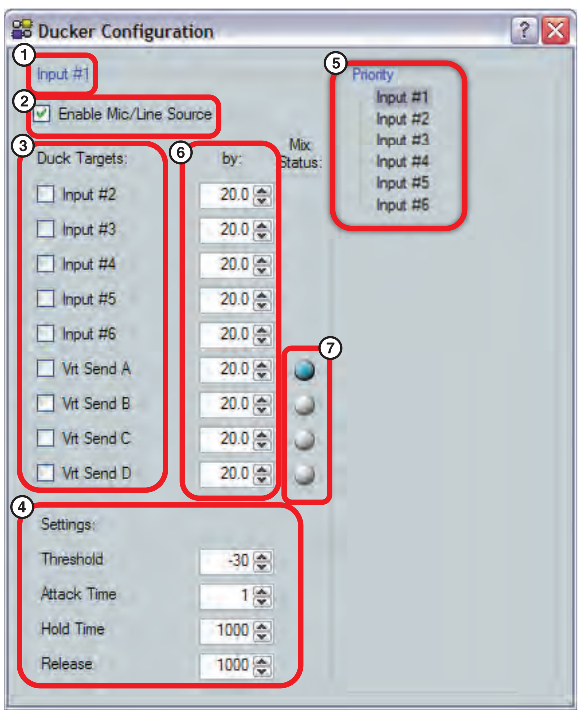


Figure 26. Ducker Configuration Dialog

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

Ducking Configuration Dialog

① *Current source indicator*

Shows the selected input. 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 via 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 and the ducker processor block is unlit.

③ *Duck Targets:*

Shows all potential input targets. Only inputs that are checked will be ducked. The current source is not available as a target (a source cannot duck itself). If the current source has been 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 soon 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. Priority levels are displayed in tree fashion. Input channels that are targets being ducked by a source are shown as indented below the source. Any input channel displayed in the tree is an active link. Click any input channel to select that channel as the current source. The current source indicator (①) reflects the selected input channel.

⑥ *By: (Target gain reduction amount)*

Individual attenuation settings for each duck target in dB. The default is 20.0 dB. If additional attenuation of the target(s) is required, increase this value.
The attenuation range is 80.0 to 0.0 dB in 0.1 dB increments.

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.

Priority

In some cases, multiple levels of ducking may be required enabling an input source to take precedence over all but one other input.

In this example inputs 2-6 are set to duck when Input #1 has a signal above the ducking threshold. Input #2 is set to duck inputs 5-6. Since Input #1 has previously been set to duck Input #2, Input #1 is disabled (grayed out) to prevent contradictory priorities.

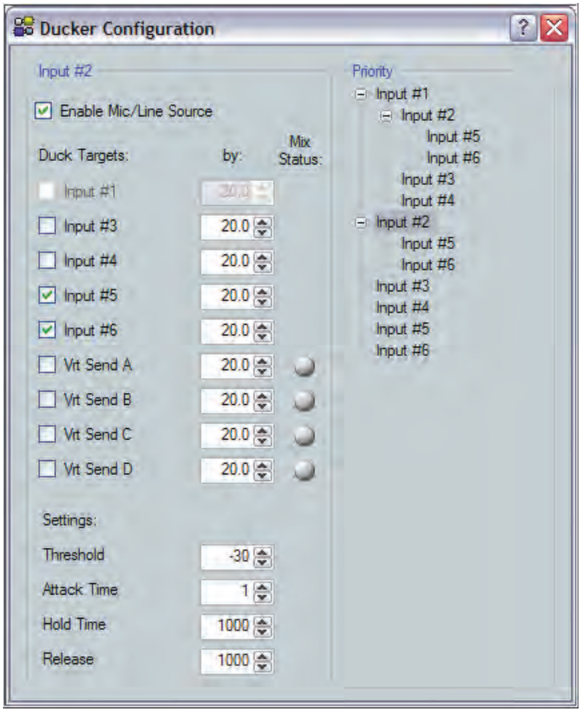


Figure 27. **Ducker Configuration, Input Priority**

Notice the priority tree on the right. The inputs are arranged by their priority status. Input #1 has all other ducked inputs under it, therefore if a signal is detected, it will trigger Inputs 2-6 to duck. If Input #2 detects a signal and there is no signal on Input #1, Input #2 will trigger inputs 5-6 to duck. However if the Input #1 signal exceeds the threshold, it will then duck all inputs including Input #2. Ducking attenuation is not additive. When an input target is ducked, regardless of how far down the priority line it is, the maximum attenuation is what is set in the **"by:"** column near the center of the dialog box.

See **Ducker Tutorials** for additional information.

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, described in mic/line input section Filter, with the exception that only three filters are allowed. See **Filter**, 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, described in the previous Dynamics section. See **Dynamics** for additional information.

Loudness (LOUD)

There is one loudness processor available on each virtual path. Loudness function and interface is identical to the Output channel Loudness block, described at **Loudness (LOUD)**.

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, **Adjusting Pre-mixer Gain**. Fader adjustment is in 1 dB increments, while adjustments can be entered manually to 0.1 dB resolution. Default is unmuted at unity (0.0 dB) gain.

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 GUI 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 of the six mic/line inputs and four virtual bus returns 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 may occur in the line output signal chain).

Shown below is a drawing 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-D, are shown as end points in this block diagram, they are connected A-A, B-B, C-C and D-D.

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

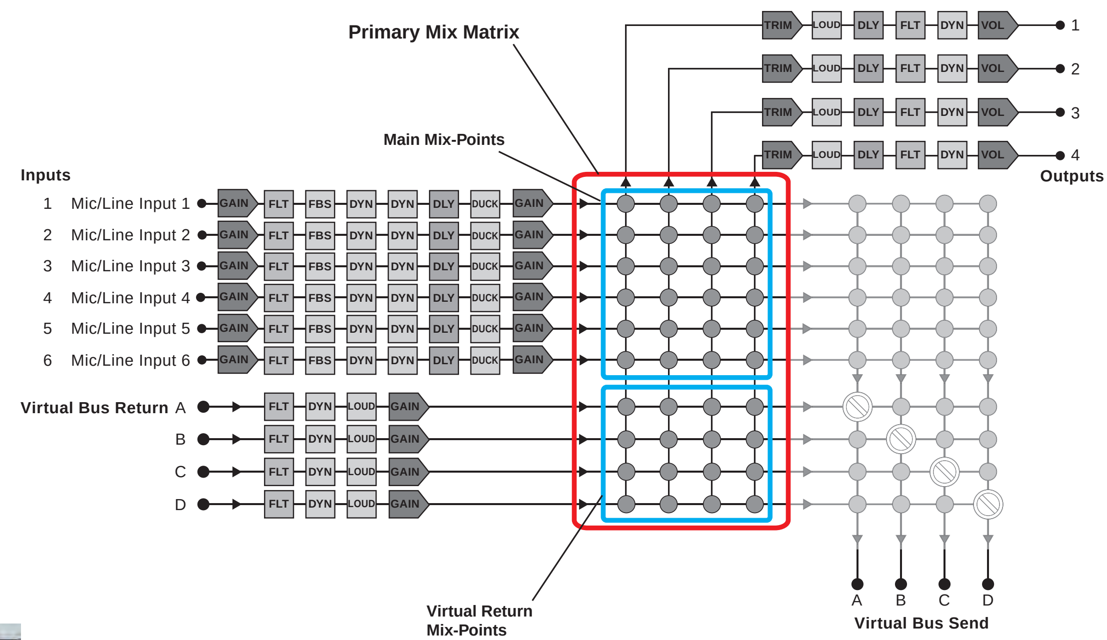
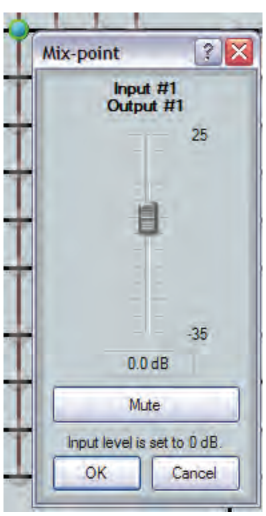


Figure 28. **Primary Mix Matrix (outlined in red)**



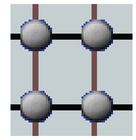
Clicking a mix-point brings focus to that mix-point. Double-clicking a mix-point opens a configuration dialog window with the following components:

- **Mono Fader —** sets mix level 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 course adjustment (Page Up/Down) increases/ decreases in 5 dB increments.
- **Mute —** Mutes and unmutes the signal to the output bus.
- **OK/Cancel —** click **OK** to accept changes and close the window. **Cancel** ignores changes and closes the window.

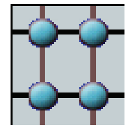
The title above the fader reflects the input and output channel names for the mix-point. The example on the left is the Input #1 to Output #1 mix-point.

Mix-point GUI behavior:

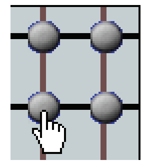
No mix information — a faint gray ball behind the mix-point indicates it is muted (contains no mix information).



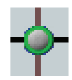
Mix information — a solid teal-colored "bubble" indicates the mix-point is unmuted.



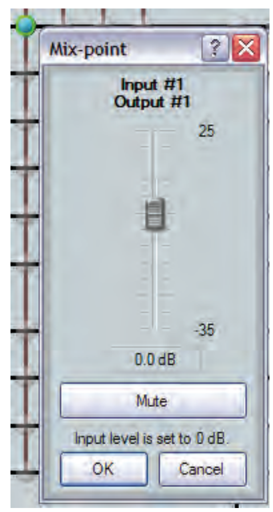
Mouse-over — the cursor changes to a hand when a mouse-over occurs at a mix-point whether the mix-point contains mix information or not.



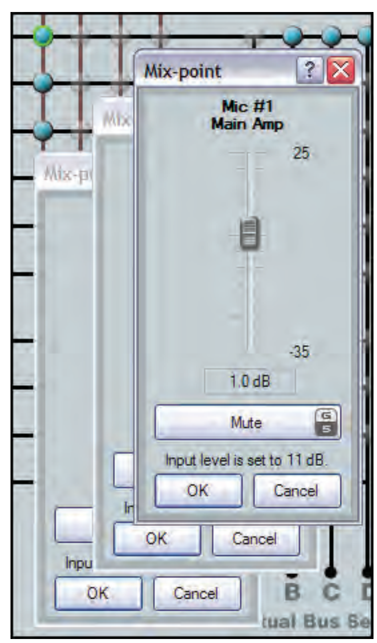
Single-click — a single click brings focus, indicated by a dark green circle around either the ball or bubble, depending on mix status.



Double-click — double-click to open 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 mix-point bubble will be gray. If unmuted, the bubble will be 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.



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

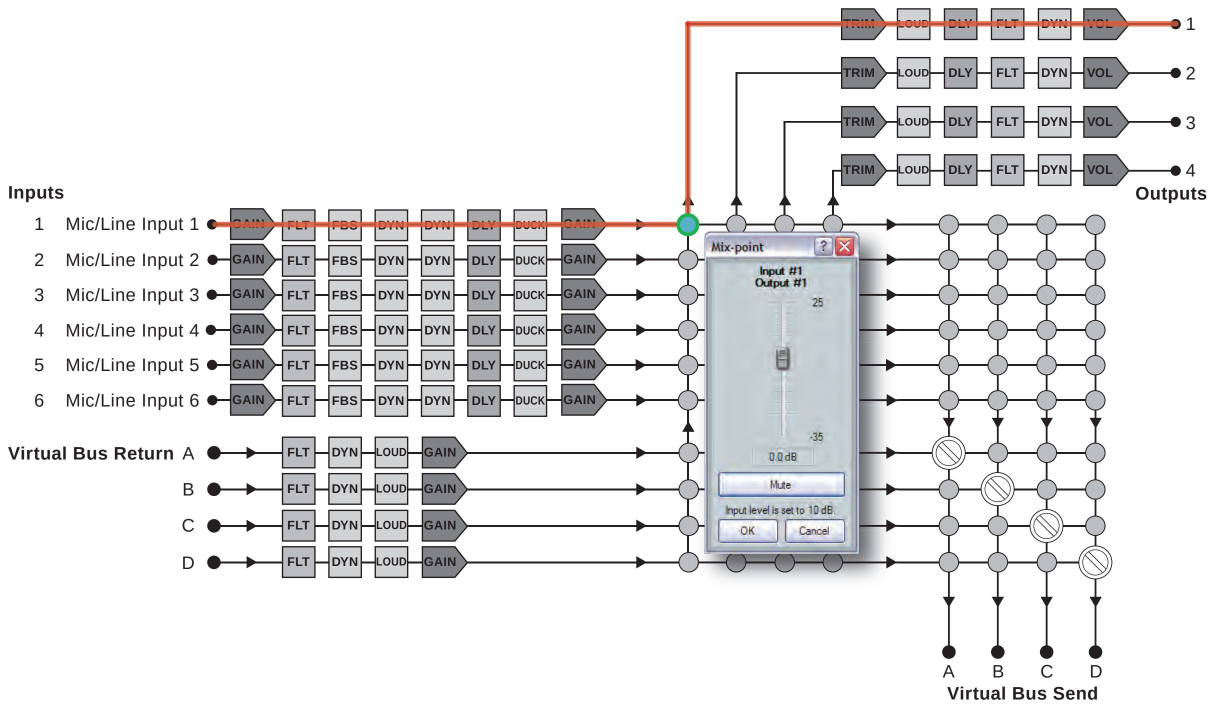


Figure 29. **Input 1 to Output 1**

In the first example, figure 29, input audio from Mic/Line Input #1 is processed and arrives at the primary mix-point. A double-click on the mix-point opens the dialog box. When the mute button is released on input 1 of the primary mix-point (shown above), the mix junction 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.

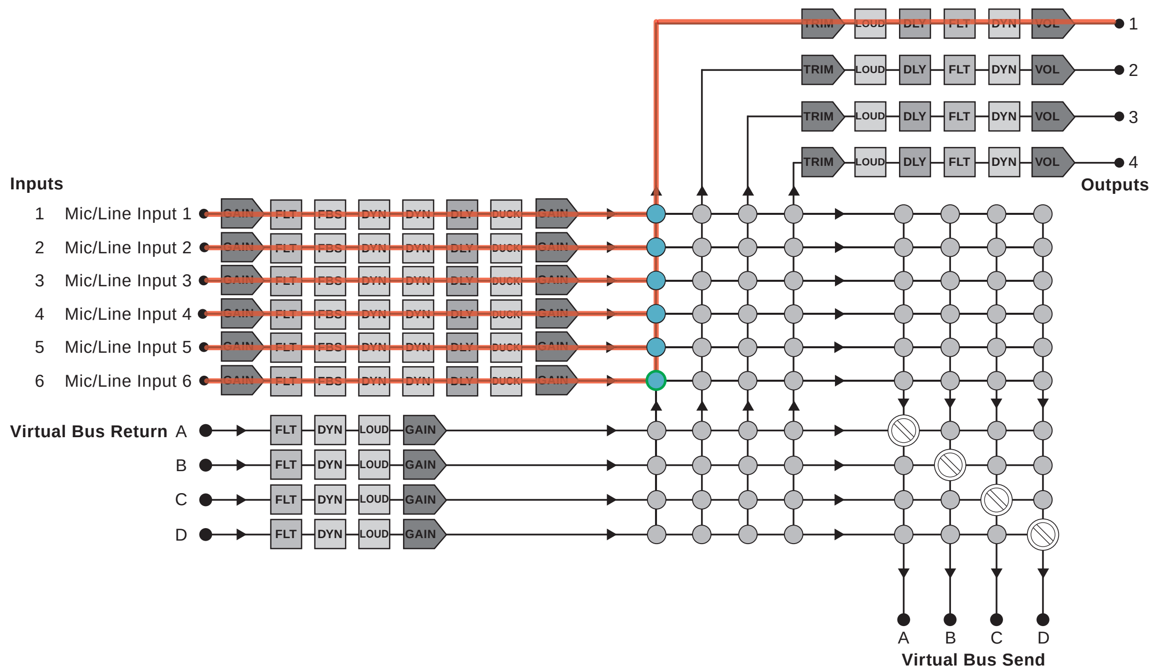


Figure 30. All Inputs to Output 1

In the next example, figure 30 above, input audio from all six mic/line inputs are processed individually and arrive at the primary mix-point. When the individual mix-point mute buttons are released, the primary mix-point junctions turn teal to indicate the routing, and all six signals are routed to output 1. Open the individual mix-point dialog boxes to adjust signal levels to the output.

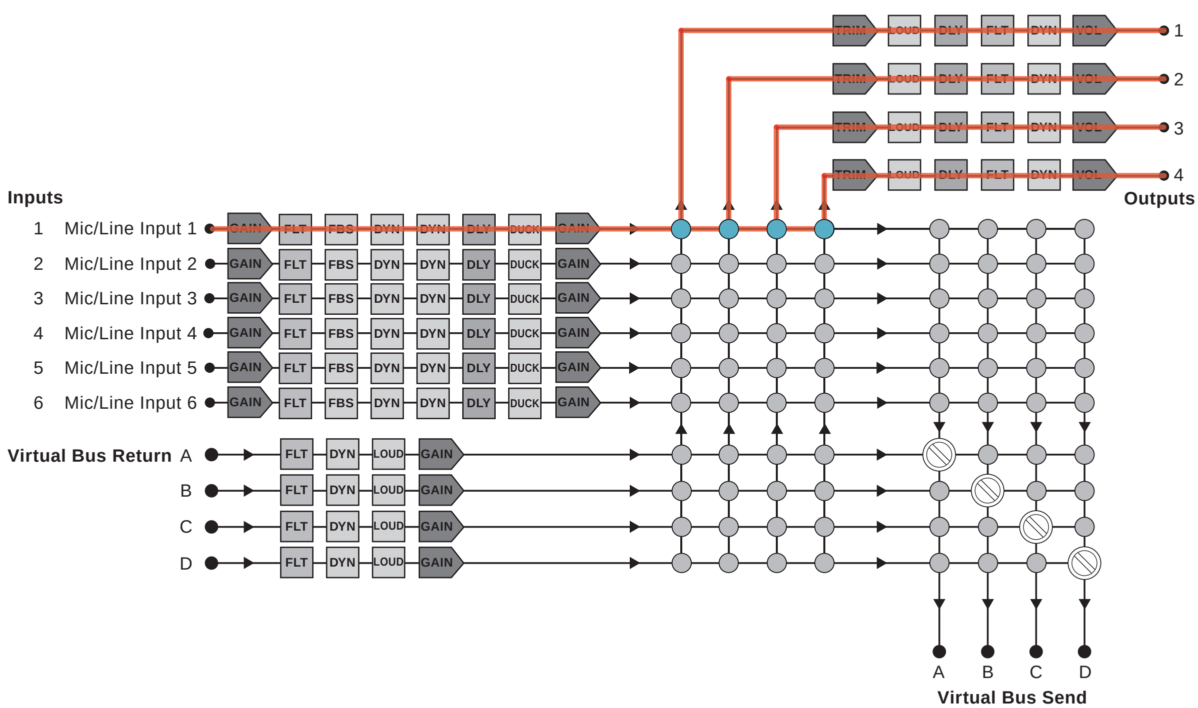


Figure 31. Input 1 to All Outputs

In the example in figure 31 above, input 1 has been routed to all four outputs by unmuting the mix-point for mic/line input 1 on each output (1-4) bus. Again, the primary mix-points are teal to indicate the 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 GUI 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 send busses. Each of the six mic/line and four virtual return inputs is connected to a mix-point for virtual bus A-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 32 below) that allows 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.

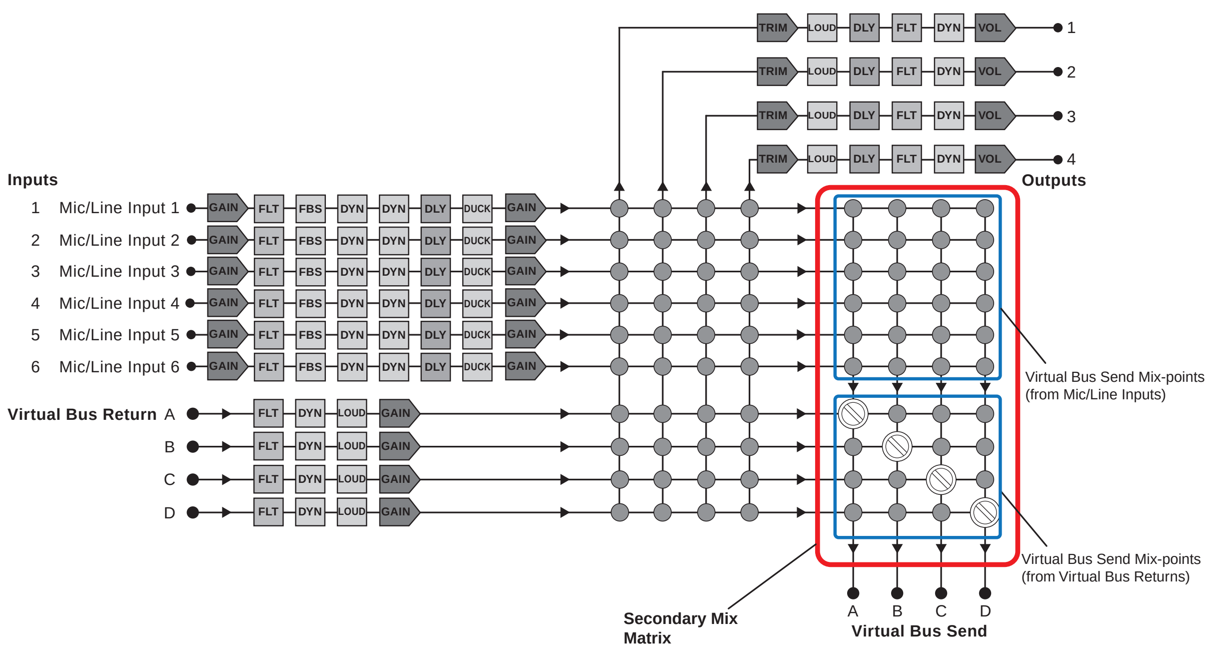


Figure 32. Secondary Mix Matrix

In the example in figure 33, input 1 is sent to the virtual bus send by muting all 4 outputs 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 could be routed to virtual bus A. Then, using the virtual bus A return gain control, a single adjustment can be used to 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. So the microphones could be routed directly to the output but the program material input could be routed to the virtual bus return where loudness contouring could be applied. The program material could then be routed to the same output as the microphones.

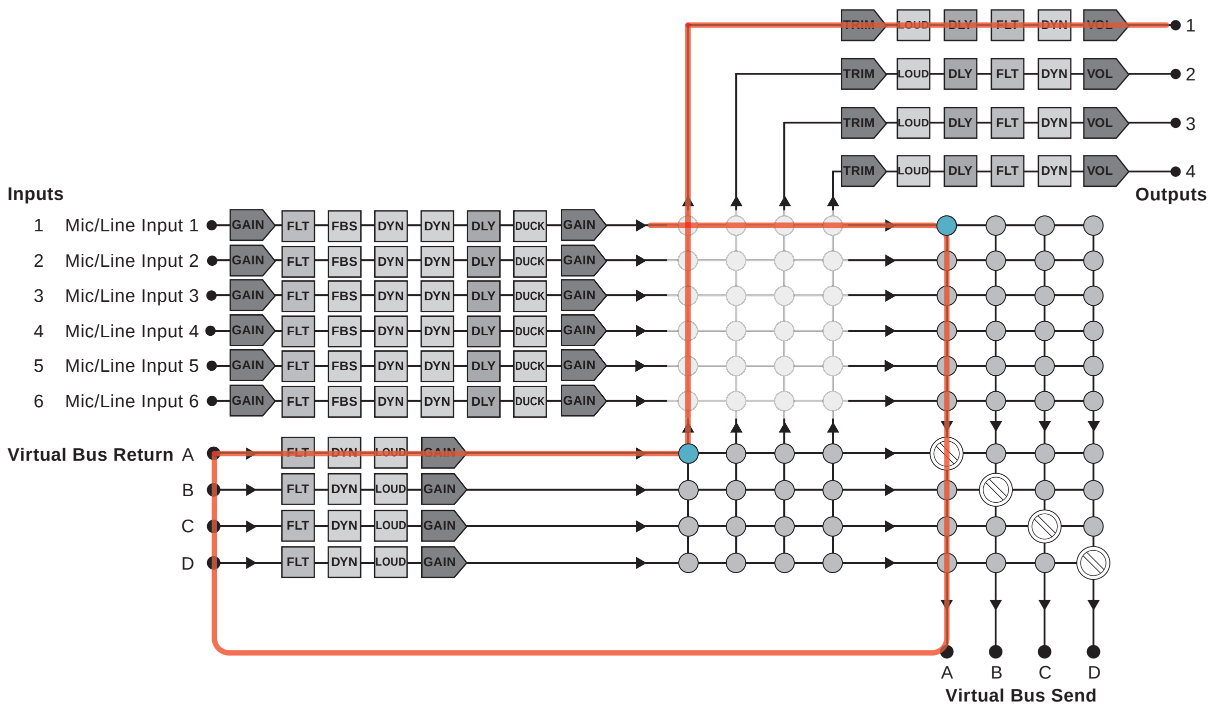


Figure 33. Input 1 to Virtual Bus A

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 compensates for changes in human perception to varying volume levels by applying a filter compensation curve to the signal in an inverse relationship to the gain control setting—the higher the gain setting, the less 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 will boost those diminished frequencies to the highest degree at low volume levels, decreasing the boost as volume increases.

Bypass must be disengaged for the loudness processor to function. The bypass button is red when engaged (loudness control defeated), 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.

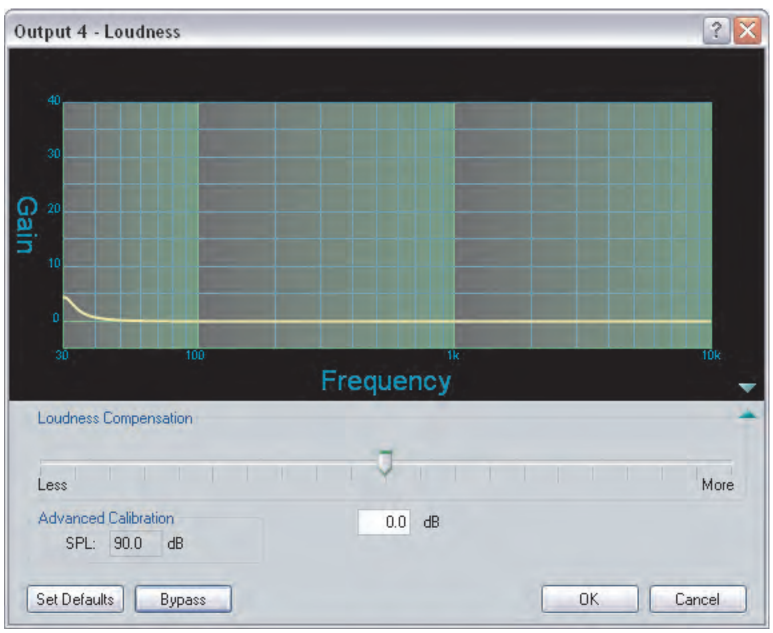


Figure 34. **Loudness Dialog Window**

The Loudness dialog window contains the following elements:

1. **Graph —** displays the compensation curve being applied to the signal. These curves are read-only, and are not adjustable from the graph.

2. **Compensation Adjustment 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.

3. **Advanced Calibration —** The 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 may 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**). A pre-recorded track of pink noise or pink noise from a signal generator is preferable for this purpose. Program material may 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**.
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.

Theoretically, calibration can be performed with the output channel volume and/or 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.

An alternate method is, with the compensation adjustment slider in its default center position and the output channel volume fader at 0 dB (100% volume), adjust the amplifier until the SPL meter reads 90 dB. Loudness is now calibrated. This method works if 90 dB is an acceptable amplifier/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. The **Calibrate** slider should be set to 0, the center point. Disengage the loudness **Bypass**. The result will be 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 less loudness compensation, move the loudness compensation slider to the left (less). For more loudness compensation, move the slider to the right (more).
5. Any adjustment made to the loudness compensation slider will 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 “A/B” the difference between loudness off and on. At a loud listening level, the difference should be minimal or barely perceivable.

Delay

Delay function and interface is identical to the line Input channel delay block, described in mic/line input section, **Delay**.

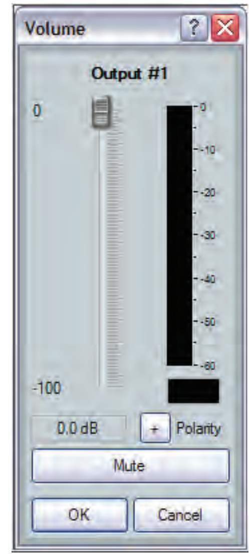
Filter

Filter function and interface is identical to the line input channel filter block, described in mic/line input section **Filter**. However, there are a total of nine filters allowed in the output signal processor chain.

Dynamics

Dynamics function and interface is identical to the line input channel dynamics block, described in mic/line input section **Dynamics**.

Volume



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 directly into the display window in 0.1 dB increments. Using the arrow buttons on the keyboard provides 1 dB increments.

An individual **Mute** button provides control of channel muting. Output polarity switching is also provided with a button that toggles between plus and minus 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 single click, while the **Cancel** button ignores changes and closes the dialog.

SIS Programming and Control

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

- **Connection Options**
- **Host-to-device communications**
- **Command/Response Table for Basic SIS Commands**
- **Command/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 the Extron SIS (Simple Instruction Set) commands, embedded Web pages, or DSP Configurator software. See chapter 2 for pin assignments and details on the configuration and control port connections. For information on DSP Configurator see **DMP Software** and for the embedded Web pages, see **HTML Operation**.

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

DMP 64 RS-232 protocol:

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

NOTE: Both rear panel configuration ports require 38400 baud communication. This is a higher speed than many other Extron Electronics products use. The DMP 64 control software automatically sets the connection for the appropriate speed. If using HyperTerminal or a similar application, make sure the PC or control system connected to these ports is set for 38400 baud.

See **RS-232 Ports**, for additional details on connecting the RS-232 port.

USB port details:
The Extron USB driver must be installed before use. See **Install the USB Driver** for driver installation instructions.

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 the DataViewer utility. The ports make serial control of the switcher possible. Use the protocol information listed above to make the connection. Once the connection is made, see **Host-to-device communications** later in this section for SIS programming details.

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. Once the connection is established, see **Host-to-device communications** later in this section for SIS programming details.

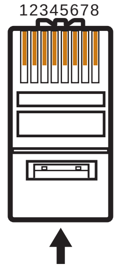
Ethernet (LAN) Port

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 (figure 37).

- **Crossover cable** — Direct connection between the computer and the DMP 64.
- **Patch (straight) cable** — Connection of the DMP 64 to an Ethernet LAN.



Crossover Cable

Pin	End 1 Wire color	End 2 Wire color
1	White-green	White-orange
2	Green	Orange
3	White-orange	White-green
4	Blue	Blue
5	White-blue	White-blue
6	Orange	Green
7	White-brown	White-brown
8	Brown	Brown
	T568A	T568B

A cable that is wired as T568A at one end and T568B at the other (Tx and Rx pairs reversed) is a "crossover" cable.

Straight-through Cable

Pin	End 1 Wire color	End 2 Wire color
1	White-green	White-orange
2	Orange	Orange
3	White-orange	White-green
4	Blue	Blue
5	White-blue	White-blue
6	Green	Green
7	White-brown	White-brown
8	Brown	Brown
	T568B	T568B

A cable wired the same at both ends is called a "straight-through" cable, because no pin/pair assignments are swapped.

Figure 35. RJ-45 Ethernet Connector Pin Assignments

Establishing a Connection

Establish a network connection to the DMP 64 as follows:

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/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.

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 on page 65.

> **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 ASCII and URL 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 already 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 via 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 via Telnet, but not via 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.

Using the Command/Response Tables

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										Esc	1B	CR	ØD	LF	ØA
	2Ø	!	21	"	22	#	23	$	24	%	25	&	26	'	27
(	28	)	29	*	2A	+	2B	,	2C	-	2D	.	2E	/	2F
Ø	30	1	31	2	32	3	33	4	34	5	35	6	36	7	37
8	38	9	39	:	3A	;	3B	<	3C	=	3D	>	3E	?	3F
@	4Ø	A	41	B	42	C	43	D	44	E	45	F	46	G	47
H	48	I	49	J	4A	K	4B	L	4C	M	4D	N	4E	O	4F
P	5Ø	Q	51	R	52	S	53	T	54	U	55	V	56	W	57
X	58	Y	59	Z	5A	[	5B	\	5C	]	5D	^	5E	_	5F
`	6Ø	a	61	b	62	c	63	d	64	e	65	f	66	g	67
h	68	i	69	j	6A	k	6B	l	6C	m	6D	n	6E	o	6F
p	7Ø	q	71	r	72	s	73	t	74	u	75	v	76	w	77
x	78	y	79	z	7A	{	7B	|	7C	}	7D	~	7E	DEL	7F

Figure 36. ASCII to Hex Conversion Table

The command/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 will be directed to the specified port and must be encoded (URL encoding) if it is non-alphanumeric. Change any non-alphanumeric character (%, +, |, ←, etc.) 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)
E12 - Invalid port number
E13 - Invalid parameter (number is out of range)
E14 - Not valid for this configuration
E17 - System timed out
E22 - Busy
E23 - Checksum error (for file uploads)
E24 - Privilege violation
E25 - Device is not present
E26 - Maximum connections exceeded
E27 - Invalid event number
E28 - Bad filename or file not found

Simple Control Port Commands - Telnet and Web-browser Accessible

Upper & 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 (i.e. a comma ',' would be represented as %2C). Characters such as '%' (percent), '+' (plus) and ' ' (space) should also be encoded in Hex.

Telnet	**Web Browser**
Escape (Hex 1B)	W [must **not** be encoded]
Carriage Return (Hex 0D)	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
(e.g. http://192.168.254.254/mypage.HTML).

To send commands using a Web browser, prefix them with the full URL followed by ?cmd=
(e.g. 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 03RS ← 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=W03RS|1*2!	URL?cmd=W03RS | 1%2A2%21	OUT 02•IN 01• ALL↵

X2 = Input number, 1 – 6

X3 = Outputs 1 – 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/ Response Table for Basic SIS Commands** that follows.

The second set of tables, "DSP SIS commands" uses the command structure outline beginning with **Command/Response Tables for DSP SIS Commands**. 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 will be used for global configuration such as setting IP addresses, date/time, while the Audio SIS commands allow functionality of the audio signal chain.

Command/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	0Q	0Q	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/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 ←	IpzX3 ↵	
View GMT offset	Esc CZ ←	X3 ↵	
Set Daylight Savings Time	Esc X34 CX ←	IpxX34 ↵	
Read Daylight Savings Time	Esc CX ←	X34 ↵	
Set IP address	Esc X14 CI ←	IpiX14 ↵	
Read IP address	Esc CI ←	X14 ↵	
Read hardware address (MAC)	Esc CH ←	X18 ↵	
Set subnet mask	Esc X19 CS ←	IpsX19 ↵	
Read subnet mask	Esc CS ←	X19 ↵	
Set gateway IP address	Esc X14 CG ←	IpgX14 ↵	
View gateway IP address	Esc CG ←	X14 ↵	
Set DHCP on	Esc 1DH ←	Idh1 ↵	
Set DHCP off	Esc 0DH ←	Idh0 ↵	
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 ←	VrbX22 ↵	
View verbose mode	Esc CV ←	X22 ↵	
Get connection listing	Esc CC ←	[number of connections] ↵	

NOTE:

X3 = Greenwich Mean Time offset	GMT offset value (-12:00 to 14:00) representing hours and minutes (HH:MM) local time is offset from GMT time
X5 = On/Off status	0=off/disable 1=on/enable
X12 = Unit name	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.
X13 = Local date/time	**Set:** MM/DD/YY-HH:MM:SS **Read:** day of week, date, month, year HH:MM:SS (e.g. Fri, 21 Jun 2002 10:54:00)
X14 = IP Address	default 192.168.255.255
X18 = Hardware MAC address	00-05-A6-xx-xx-xx
X19 = Subnet mask	Default 255.255.0.0
X22 = Verbose/Response mode	0=clear, 1=verbose, 2=tagged responses, 3=verbose + tagged responses
X34 = Daylight Saving time	0=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).
X49 = Alpha-numeric unit name	combination of unit name and last three pairs of MAC address

Command/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* ↵ *... space_remaining•Bytes Left* ↵↵

NOTE: LF has same response from unit as DF command, except path / directory will precede filenames for files from directories below current directory.

NOTE: 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 timeout1-65000 steps, (1 step=10 seconds)

Command/Response table for basic SIS commands (continued)

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 ↵

NOTE:

X1 = Port Number — 01-99 represented by 2 Bytes (ASCII).

X2 = Command data section — **NOTE:** For web encoding only: Data will be directed to specified port and must be encoded if non-alpha numeric. Since data can include either command terminator, they must be encoded as follows when used within the data section: Space (Hex: 20) would be encoded as %20 and Plus sign (Hex: 2B) would be encoded as %2B

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/Response table for basic SIS commands (continued)

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)	

NOTE:

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/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	EscX10,X11NG←	NmgX10,X11↵	
Example:	Esc1,Security 1NG←	Nmg01,Security 1↵	Name preset 1 "Security 1".
Read preset name	EscX10NG←	X11↵	
Example:	Esc2NG←	Security 2↵	
Recall a preset	X10.	RprX10↵	Command character is a period
Example	5.	RprX10↵	Recall preset 5, which becomes the current configuration.
Write input name	EscX3,X11NI←	NmiX3,X11↵	
Example:	Esc9,Podium cam1NI←	Nmi09,Podium cam↵	Name input 9 "Podium cam".
Read input name	EscX3NI←	X11↵	
Write output name	EscX2,X11NO←	NmoX2,X11↵	
Example:	Esc1,Main PJ1NO←	Nmo01,Main PJ1↵	Name output 1 "Main PJ1".
Read output name	EscX2NO←	X11↵	
Resets			
Reset presets and names	EscZG←	Zpg↵	Clear all presets and their names.
Reset an individual preset	EscX10ZG←	ZpgX10↵	Clear preset X10.
Reset a group	EscZX20GRPM←	GrpmZX20↵	Delete all members from group X20, reset parameters and soft limits.
NOTE: See **Group Masters**, for more information about audio group masters.			
Reset flash	EscZFFF←	Zpf↵	Reset flash memory (erase user-supplied files).
System Reset (factory defaults)	EscZXXX←	Zpx↵	Resets all processors, level controls and mixers to default.
Reset all device settings and delete files	EscZY←	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	EscZQQQ←	Zpq↵	Similar to **System Reset**, plus sets the IP address to 192.168.254.254 and the subnet mask to 255.255.0.0.

NOTE:

X3 = Input number	01 – 06
X2 = Output number	01 – 04
X10 = Preset #	32 maximum (0 = current configuration)
X11 = Name	12 characters maximum
X20 = Group master group number	01 - 32

Command/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 37 shows the specific DSP processors available for SIS commands.

> **NOTE:** The entire signal flow is described in more detail in the section, **Windows-based Program Control**.

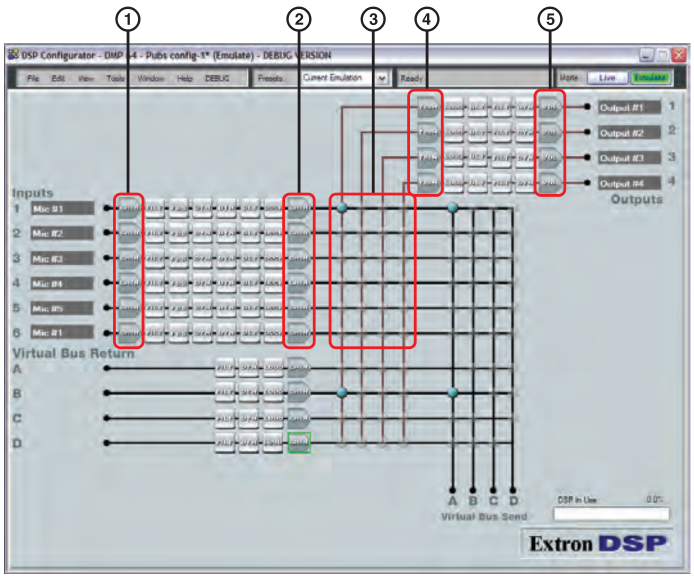


Figure 37. DSP Processors Addressable via SIS Commands

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. Post-mixer trim block (gain only)
5. Output volume (including gain and mute)

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 74.
X61	=	Level value; mix-point gain (③), and post-mixer trim(④)	See the table on page 75 through 77. -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 page 78 through 81. -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 page 82 through 85. 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's name, passwords, or locally created file names.

The DMP 64 rejects the following characters:
{space (spaces **are** oK for names)} + } ~ , @ = ' [ ] { } < > ' " semicolon (;) colon (:) | \ and ?.

Command/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			
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: • 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 is -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. All responses are shown with the DMP 64 in Verbose mode 2 or 3.			
Set a trim or gain (excluding mic/line inputs)	EscGX60*X61AU←	DsGX60*X61↵	Set trim or mix control X60 to a value of X61 dB.
Example 1 (pre-mixer gain):	EscG40105*2040AU←	DsG40105*2040↵	Set the #6 pre-mixer gain to a value of -0.8 dB.
Example 2 (mix-point gain):	EscG20001*2213AU←	DsG20001*2213↵	Mix +16.5 dB of mic 1 into output 2.
Set a mic/line gain	EscGX60*X62AU←	DsGX60*X62↵	Set mic/line gain control X60 to a value of X62 dB.
Example:	EscG40001*2288AU←	DsG40001*2288↵	Set the mic/line input 2 gain to a level of +24.0 dB.
Read a trim or mix (excluding mic/line inputs)	EscGX60AU←	DsGX60*X61↵	DSP trim or mix control X60 is set to a value of X61 dB.
Example 1 (post mixer gain control):	EscG60101AU←	DsG60101*2103↵	Output 2, post mixer trim is set to a value of +5.5 dB.
Example 2 (mix control):	EscG20203AU←	DsG20203*2140↵	+9.2 dB of mic 3 is mixed into output 4.
Read a mic/line gain	EscGX60AU←	DsGX60*X62↵	Mic/line gain control X60 is set to a value of X62 dB.
Example:	EscG40000AU←	DsG40000*2598↵	Mic/line input 1 gain is set to a value of +55.0 dB.
Audio mute			
NOTE: • The post-mixer trim cannot be muted. • All responses are shown with the mixer device in Verbose mode 2 or 3.			
Audio mute	EscMX60*1AU←	DsMX60*1↵	Mute audio point X60.
Example:	EscM20301*1AU←	DsM20301*1↵	Mute mix-point input 4 to output 2.
Audio unmute	EscMX60*0AU←	DsMX60*0↵	Unmute audio point X60.
Read audio mute or level	EscMX60AU←	DsMX60*X64↵	X64: 0 = mute off, 1 = mute on.

NOTE:

X60	= Audio level control, or mix-point select	See the tables on page 74.
X61	= Level value; mix-point or post-mixer trim	See the table on page 75 through 77.
X62	= Mic/line gain level value	See the table on page 78 through 81.
X63	= Level value; pre-mixer gain and output volume	See the table on page 82 through 85.
X64	= Mute status	0 = unmute 1 = mute

Command/Response table for DSP SIS commands (continued)

Command	ASCII command (host to device)	Response (device to host)	Additional description
Audio group master commands			
NOTE: • See **Group Masters**, 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*GRPM ↵	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*GRPM ↵	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 lower GRPM ←	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 O X65 GRPM ←	GrpmO 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 ↵.

NOTE:	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/Response table for DSP SIS commands (continued)

Command	ASCII command (host to device)	Response (device to host)	Additional description
Protected configuration			
NOTE: The DMP 64 can save and recall a Personal Identification Number (PIN)-protected configuration, including all presets, 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.			
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 ↵	

NOTE: X70 = Personal Identification Number (PIN) — Four numeric digits, default = 0000
X71 = Protected configuration status — 0 = no protected configuration saved
1 = protected configuration saved

Command/Response table for DSP SIS commands (continued)

Table 1. X60 — Level control and mix-point selection

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 Output 1	40100
Mic/Line Output 2	40101
Mic/Line Output 3	40102
Mic/Line Output 4	40103
Mic/Line Output 5	40104
Mic/Line Output 6	40105

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 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 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

④ Post-mixer trim	X60
Output 1	60100
Output 2	60101
Output 3	60102
Output 4	60103

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

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	dB value	X61	dB value	X61	dB value	X61	dB value	X61	dB value	X61	dB value	X61	dB value	X61	dB value	X61	dB value	X61
																		-35.0	1698
-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)

Table 3. Post-mixer trim and mix-point gain

dB value	X61	dB value	X61	dB value	X61	dB value	X61	dB value	X61	dB value	X61	dB value	X61	dB value	X61	dB value	X61	dB value	X61
																		-12.0	1928
-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

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

Table 4. Mix-point gain only

dB value	X61	dB value	X61	dB value	X61	dB value	X61	dB value	X61	dB value	X61	dB value	X61	dB value	X61	dB value	X61	dB value	X61
																		12.0	2168
12.1	2169	12.2	2170	12.3	2171	12.4	2172	12.5	2173	12.6	2174	12.7	2175	12.8	2176	12.9	2177	13.0	2178
13.1	2179	13.2	2180	13.3	2181	13.4	2182	13.5	2183	13.6	2184	13.7	2185	13.8	2186	13.9	2187	14.0	2188
14.1	2189	14.2	2190	14.3	2191	14.4	2192	14.5	2193	14.6	2194	14.7	2195	14.8	2196	14.9	2197	15.0	2198
15.1	2199	15.2	2200	15.3	2201	15.4	2202	15.5	2203	15.6	2204	15.7	2205	15.8	2206	15.9	2207	16.0	2208
16.1	2209	16.2	2210	16.3	2211	16.4	2212	16.5	2213	16.6	2214	16.7	2215	16.8	2216	16.9	2217	17.0	2218
17.1	2219	17.2	2220	17.3	2221	17.4	2222	17.5	2223	17.6	2224	17.7	2225	17.8	2226	17.9	2227	18.0	2228
18.1	2229	18.2	2230	18.3	2231	18.4	2232	18.5	2233	18.6	2234	18.7	2235	18.8	2236	18.9	2237	19.0	2238
19.1	2239	19.2	2240	19.3	2241	19.4	2242	19.5	2243	19.6	2244	19.7	2245	19.8	2246	19.9	2247	20.0	2248
20.1	2249	20.2	2250	20.3	2251	20.4	2252	20.5	2253	20.6	2254	20.7	2255	20.8	2256	20.9	2257	21.0	2258
21.1	2259	21.2	2260	21.3	2261	21.4	2262	21.5	2263	21.6	2264	21.7	2265	21.8	2266	21.9	2267	22.0	2268
22.1	2269	22.2	2270	22.3	2271	22.4	2272	22.5	2273	22.6	2274	22.7	2275	22.8	2276	22.9	2277	23.0	2278
23.1	2279	23.2	2280	23.3	2281	23.4	2282	23.5	2283	23.6	2284	23.7	2285	23.8	2286	23.9	2287	24.0	2288
24.1	2289	24.2	2290	24.3	2291	24.4	2292	24.5	2293	24.6	2294	24.7	2295	24.8	2296	24.9	2297	25.0	2298

X62 — Mic/line gain (①)

dB Value	X62	dB Value	X62	dB Value	X62	dB Value	X62	dB Value	X62	dB Value	X62	dB Value		dB Value	X62	dB Value	X62	dB Value	X62
																		-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	-12.0	1928
-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	2023	-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	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

X62 — Mic/line gain (①), (continued)

dB Value	X62	dB Value	X62	dB Value	X62	dB Value	X62	dB Value	X62	dB Value	X62	dB Value		dB Value	X62	dB Value	X62	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	12.9	2177	13.0	2178
13.1	2179	13.2	2180	13.3	2181	13.4	2182	13.5	2183	13.6	2184	13.7	2185	13.8	2186	13.9	2187	14.0	2188
14.1	2189	14.2	2190	14.3	2191	14.4	2192	14.5	2193	14.6	2194	14.7	2195	14.8	2196	14.9	2197	15.0	2198
15.1	2199	15.2	2200	15.3	2201	15.4	2202	15.5	2203	15.6	2204	15.7	2205	15.8	2206	15.9	2207	16.0	2208
16.1	2209	16.2	2210	16.3	2211	16.4	2212	16.5	2213	16.6	2214	16.7	2215	16.8	2216	16.9	2217	17.0	2218
17.1	2219	17.2	2220	17.3	2221	17.4	2222	17.5	2223	17.6	2224	17.7	2225	17.8	2226	17.9	2227	18.0	2228
18.1	2229	18.2	2230	18.3	2231	18.4	2232	18.5	2233	18.6	2234	18.7	2235	18.8	2236	18.9	2237	19.0	2238
19.1	2239	19.2	2240	19.3	2241	19.4	2242	19.5	2243	19.6	2244	19.7	2245	19.8	2246	19.9	2247	20.0	2248
20.1	2249	20.2	2250	20.3	2251	20.4	2252	20.5	2253	20.6	2254	20.7	2255	20.8	2256	20.9	2257	21.0	2258
21.1	2259	21.2	2260	21.3	2261	21.4	2262	21.5	2263	21.6	2264	21.7	2265	21.8	2266	21.9	2267	22.0	2268
22.1	2269	22.2	2270	22.3	2271	22.4	2272	22.5	2273	22.6	2274	22.7	2275	22.8	2276	22.9	2277	23.0	2278
23.1	2279	23.2	2280	23.3	2281	23.4	2282	23.5	2283	23.6	2284	23.7	2285	23.8	2286	23.9	2287	24.0	2288
24.1	2289	24.2	2290	24.3	2291	24.4	2292	24.5	2293	24.6	2294	24.7	2295	24.8	2296	24.9	2297	25.0	2298
25.1	2299	25.2	2300	25.3	2301	25.4	2302	25.5	2303	25.6	2304	25.7	2305	25.8	2306	25.9	2307	26.0	2308
26.1	2309	26.2	2310	26.3	2311	26.4	2312	26.5	2313	26.6	2314	26.7	2315	26.8	2316	26.9	2317	27.0	2318
27.1	2319	27.2	2320	27.3	2321	27.4	2322	27.5	2323	27.6	2324	27.7	2325	27.8	2326	27.9	2327	28.0	2328
28.1	2329	28.2	2330	28.3	2331	28.4	2332	28.5	2333	28.6	2334	28.7	2335	28.8	2336	28.9	2337	29.0	2338
29.1	2339	29.2	2340	29.3	2341	29.4	2342	29.5	2343	29.6	2344	29.7	2345	29.8	2346	29.9	2347	30.0	2348
30.1	2349	30.2	2350	30.3	2351	30.4	2352	30.5	2353	30.6	2354	30.7	2355	30.8	2356	30.9	2357	31.0	2358
31.1	2359	31.2	2360	31.3	2361	31.4	2362	31.5	2363	31.6	2364	31.7	2365	31.8	2366	31.9	2367	32.0	2368
32.1	2369	32.2	2370	32.3	2371	32.4	2372	32.5	2373	32.6	2374	32.7	2375	32.8	2376	32.9	2377	33.0	2378
33.1	2379	33.2	2380	33.3	2381	33.4	2382	33.5	2383	33.6	2384	33.7	2385	33.8	2386	33.9	2387	34.0	2388
34.1	2389	34.2	2390	34.3	2391	34.4	2392	34.5	2393	34.6	2394	34.7	2395	34.8	2396	34.9	2397	35.0	2398
35.1	2399	35.2	2400	35.3	2401	35.4	2402	35.5	2403	35.6	2404	35.7	2405	35.8	2406	35.9	2407	36.0	2408
36.1	2409	36.2	2410	36.3	2411	36.4	2412	36.5	2413	36.6	2414	36.7	2415	36.8	2416	36.9	2417	37.0	2418
37.1	2419	37.2	2420	37.3	2421	37.4	2422	37.5	2423	37.6	2424	37.7	2425	37.8	2426	37.9	2427	38.0	2428
38.1	2429	38.2	2430	38.3	2431	38.4	2432	38.5	2433	38.6	2434	38.7	2435	38.8	2436	38.9	2437	39.0	2438
39.1	2439	39.2	2440	39.3	2441	39.4	2442	39.5	2443	39.6	2444	39.7	2445	39.8	2446	39.9	2447	40.0	2448
40.1	2449	40.2	2450	40.3	2451	40.4	2452	40.5	2453	40.6	2454	40.7	2455	40.8	2456	40.9	2457	41.0	2458
41.1	2459	41.2	2460	41.3	2461	41.4	2462	41.5	2463	41.6	2464	41.7	2465	41.8	2466	41.9	2467	42.0	2468
42.1	2469	42.2	2470	42.3	2471	42.4	2472	42.5	2473	42.6	2474	42.7	2475	42.8	2476	42.9	2477	43.0	2478

X62 — Mic/line gain (①), (continued)

dB Value	X62	dB Value	X62	dB Value	X62	dB Value	X62	dB Value	X62	dB Value	X62	dB Value		dB Value	X62	dB 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	43.9	2487	44.0	2488
44.1	2489	44.2	2490	44.3	2491	44.4	2492	44.5	2493	44.6	2494	44.7	2495	44.8	2496	44.9	2497	45.0	2498
45.1	2499	45.2	2500	45.3	2501	45.4	2502	45.5	2503	45.6	2504	45.7	2505	45.8	2506	45.9	2507	46.0	2508
46.1	2509	46.2	2510	46.3	2511	46.4	2512	46.5	2513	46.6	2514	46.7	2515	46.8	2516	46.9	2517	47.0	2518
47.1	2519	47.2	2520	47.3	2521	47.4	2522	47.5	2523	47.6	2524	47.7	2525	47.8	2526	47.9	2527	48.0	2528
48.1	2529	48.2	2530	48.3	2531	48.4	2532	48.5	2533	48.6	2534	48.7	2535	48.8	2536	48.9	2537	49.0	2538
49.1	2539	49.2	2540	49.3	2541	49.4	2542	49.5	2543	49.6	2544	49.7	2545	49.8	2546	49.9	2547	50.0	2548
50.1	2549	50.2	2550	50.3	2551	50.4	2552	50.5	2553	50.6	2554	50.7	2555	50.8	2556	50.9	2557	51.0	2558
51.1	2559	51.2	2560	51.3	2561	51.4	2562	51.5	2563	51.6	2564	51.7	2565	51.8	2566	51.9	2567	52.0	2568
52.1	2569	52.2	2570	52.3	2571	52.4	2572	52.5	2573	52.6	2574	52.7	2575	52.8	2576	52.9	2577	53.0	2578
53.1	2579	53.2	2580	53.3	2581	53.4	2582	53.5	2583	53.6	2584	53.7	2585	53.8	2586	53.9	2587	54.0	2588
54.1	2589	54.2	2590	54.3	2591	54.4	2592	54.5	2593	54.6	2594	54.7	2595	54.8	2596	54.9	2597	55.0	2598
55.1	2599	55.2	2600	55.3	2601	55.4	2602	55.5	2603	55.6	2604	55.7	2605	55.8	2606	55.9	2607	56.0	2608
56.1	2609	56.2	2610	56.3	2611	56.4	2612	56.5	2613	56.6	2614	56.7	2615	56.8	2616	56.9	2617	57.0	2618
57.1	2619	57.2	2620	57.3	2621	57.4	2622	57.5	2623	57.6	2624	57.7	2625	57.8	2626	57.9	2627	58.0	2628
58.1	2629	58.2	2630	58.3	2631	58.4	2632	58.5	2633	58.6	2634	58.7	2635	58.8	2636	58.9	2637	59.0	2638
59.1	2639	59.2	2640	59.3	2641	59.4	2642	59.5	2643	59.6	2644	59.7	2645	59.8	2646	59.9	2647	60.0	2648
60.1	2649	60.2	2650	60.3	2651	60.4	2652	60.5	2653	60.6	2654	60.7	2655	60.8	2656	60.9	2657	61.0	2658
61.1	2659	61.2	2660	61.3	2661	61.4	2662	61.5	2663	61.6	2664	61.7	2665	61.8	2666	61.9	2667	62.0	2668
62.1	2669	62.2	2670	62.3	2671	62.4	2672	62.5	2673	62.6	2674	62.7	2675	62.8	2676	62.9	2677	63.0	2678
63.1	2679	63.2	2680	63.3	2681	63.4	2682	63.5	2683	63.6	2684	63.7	2685	63.8	2686	63.9	2687	64.0	2688
64.1	2689	64.2	2690	64.3	2691	64.4	2692	64.5	2693	64.6	2694	64.7	2695	64.8	2696	64.9	2697	65.0	2698
65.1	2699	65.2	2700	65.3	2701	65.4	2702	65.5	2703	65.6	2704	65.7	2705	65.8	2706	65.9	2707	66.0	2708
66.1	2709	66.2	2710	66.3	2711	66.4	2712	66.5	2713	66.6	2714	66.7	2715	66.8	2716	66.9	2717	67.0	2718
67.1	2719	67.2	2720	67.3	2721	67.4	2722	67.5	2723	67.6	2724	67.7	2725	67.8	2726	67.9	2727	68.0	2728
68.1	2729	68.2	2730	68.3	2731	68.4	2732	68.5	2733	68.6	2734	68.7	2735	68.8	2736	68.9	2737	69.0	2738
69.1	2739	69.2	2740	69.3	2741	69.4	2742	69.5	2743	69.6	2744	69.7	2745	69.8	2746	69.9	2747	70.0	2748
70.1	2749	70.2	2750	70.3	2751	70.4	2752	70.5	2753	70.6	2754	70.7	2755	70.8	2756	70.9	2757	71.0	2758
71.1	2759	71.2	2760	71.3	2761	71.4	2762	71.5	2763	71.6	2764	71.7	2765	71.8	2766	71.9	2767	72.0	2768
72.1	2769	72.2	2770	72.3	2771	72.4	2772	72.5	2773	72.6	2774	72.7	2775	72.8	2776	72.9	2777	73.0	2778

X62 — Mic/line gain (①), (continued)

dB Value	X62	dB Value	X62	dB Value	X62	dB Value	X62	dB Value	X62	dB Value	X62	dB Value		dB Value	X62	dB Value	X62	dB Value	X62
73.1	2779	73.2	2780	73.3	2781	73.4	2782	73.5	2783	73.6	2784	73.7	2785	73.8	2786	73.9	2787	74.0	2788
74.1	2789	74.2	2790	74.3	2791	74.4	2792	74.5	2793	74.6	2794	74.7	2795	74.8	2796	74.9	2797	75.0	2798
75.1	2799	75.2	2800	75.3	2801	75.4	2802	75.5	2803	75.6	2804	75.7	2805	75.8	2806	75.9	2807	76.0	2808
76.1	2809	76.2	2810	76.3	2811	76.4	2812	76.5	2813	76.6	2814	76.7	2815	76.8	2816	76.9	2817	77.0	2818
77.1	2819	77.2	2820	77.3	2821	77.4	2822	77.5	2823	77.6	2824	77.7	2825	77.8	2826	77.9	2827	78.0	2828
78.1	2829	78.2	2830	78.3	2831	78.4	2832	78.5	2833	78.6	2834	78.7	2835	78.8	2836	78.9	2837	79.0	2838
79.1	2839	79.2	2840	79.3	2841	79.4	2842	79.5	2843	79.6	2844	79.7	2845	79.8	2846	79.9	2847	80.0	2848

X63 — Pre-mixer gain (②) and output volume (⑤)

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

dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63
																		-1000.0	1048
-99.9	1049	-99.8	1050	-99.7	1051	-99.6	1052	-99.5	1053	-99.4	1054	-99.3	1055	-99.2	1056	-99.1	1057	-99.0	1058
-98.9	1059	-98.8	1060	-98.7	1061	-98.6	1062	-98.5	1063	-98.4	1064	-98.3	1065	-98.2	1066	-98.1	1067	-98.0	1068
-97.9	1069	-97.8	1070	-97.7	1071	-97.6	1072	-97.5	1073	-97.4	1074	-97.3	1075	-97.2	1076	-97.1	1077	-97.0	1078
-96.9	1079	-96.8	1080	-96.7	1081	-96.6	1082	-96.5	1083	-96.4	1084	-96.3	1085	-96.2	1086	-96.1	1087	-96.0	1088
-95.9	1089	-95.8	1090	-95.7	1091	-95.6	1092	-95.5	1093	-95.4	1094	-95.3	1095	-95.2	1096	-95.1	1097	-95.0	1098
-94.9	1099	-94.8	1100	-94.7	1101	-94.6	1102	-94.5	1103	-94.4	1104	-94.3	1105	-94.2	1106	-94.1	1107	-94.0	1108
-93.9	1109	-93.8	1110	-93.7	1111	-93.6	1112	-93.5	1113	-93.4	1114	-93.3	1115	-93.2	1116	-93.1	1117	-93.0	1118
-92.9	1119	-92.8	1120	-92.7	1121	-92.6	1122	-92.5	1123	-92.4	1124	-92.3	1125	-92.2	1126	-92.1	1127	-92.0	1128
-91.9	1129	-91.8	1130	-91.7	1131	-91.6	1132	-91.5	1133	-91.4	1134	-91.3	1135	-91.2	1136	-91.1	1137	-91.0	1138
-90.9	1139	-90.8	1140	-90.7	1141	-90.6	1142	-90.5	1143	-90.4	1144	-90.3	1145	-90.2	1146	-90.1	1147	-90.0	1148
-89.9	1149	-89.8	1150	-89.7	1151	-89.6	1152	-89.5	1153	-89.4	1154	-89.3	1155	-89.2	1156	-89.1	1157	-89.0	1158
-88.9	1159	-88.8	1160	-88.7	1161	-88.6	1162	-88.5	1163	-88.4	1164	-88.3	1165	-88.2	1166	-88.1	1167	-88.0	1168
-87.9	1169	-87.8	1170	-87.7	1171	-87.6	1172	-87.5	1173	-87.4	1174	-87.3	1175	-87.2	1176	-87.1	1177	-87.0	1178
-86.9	1179	-86.8	1180	-86.7	1181	-86.6	1182	-86.5	1183	-86.4	1184	-86.3	1185	-86.2	1186	-86.1	1187	-86.0	1188
-85.9	1189	-85.8	1190	-85.7	1191	-85.6	1192	-85.5	1193	-85.4	1194	-85.3	1195	-85.2	1196	-85.1	1197	-85.0	1198
-84.9	1199	-84.8	1200	-84.7	1201	-84.6	1202	-84.5	1203	-84.4	1204	-84.3	1205	-84.2	1206	-84.1	1207	-84.0	1208
-83.9	1209	-83.8	1210	-83.7	1211	-83.6	1212	-83.5	1213	-83.4	1214	-83.3	1215	-83.2	1216	-83.1	1217	-83.0	1218
-82.9	1219	-82.8	1220	-82.7	1221	-82.6	1222	-82.5	1223	-82.4	1224	-82.3	1225	-82.2	1226	-82.1	1227	-82.0	1228
-81.9	1229	-81.8	1230	-81.7	1231	-81.6	1232	-81.5	1233	-81.4	1234	-81.3	1235	-81.2	1236	-81.1	1237	-81.0	1238
-80.9	1239	-80.8	1240	-80.7	1241	-80.6	1242	-80.5	1243	-80.4	1244	-80.3	1245	-80.2	1246	-80.1	1247	-80.0	1248
-79.9	1249	-79.8	1250	-79.7	1251	-79.6	1252	-79.5	1253	-79.4	1254	-79.3	1255	-79.2	1256	-79.1	1257	-79.0	1258
-78.9	1259	-78.8	1260	-78.7	1261	-78.6	1262	-78.5	1263	-78.4	1264	-78.3	1265	-78.2	1266	-78.1	1267	-78.0	1268
-77.9	1269	-77.8	1270	-77.7	1271	-77.6	1272	-77.5	1273	-77.4	1274	-77.3	1275	-77.2	1276	-77.1	1277	-77.0	1278
-76.9	1279	-76.8	1280	-76.7	1281	-76.6	1282	-76.5	1283	-76.4	1284	-76.3	1285	-76.2	1286	-76.1	1287	-76.0	1288
-75.9	1289	-75.8	1290	-75.7	1291	-75.6	1292	-75.5	1293	-75.4	1294	-75.3	1295	-75.2	1296	-75.1	1297	-75.0	1298
-74.9	1299	-74.8	1300	-74.7	1301	-74.6	1302	-74.5	1303	-74.4	1304	-74.3	1305	-74.2	1306	-74.1	1307	-74.0	1308
-73.9	1309	-73.8	1310	-73.7	1311	-73.6	1312	-73.5	1313	-73.4	1314	-73.3	1315	-73.2	1316	-73.1	1317	-73.0	1318
-72.9	1319	-72.8	1320	-72.7	1321	-72.6	1322	-72.5	1323	-72.4	1324	-72.3	1325	-72.2	1326	-72.1	1327	-72.0	1328
-71.9	1329	-71.8	1330	-71.7	1331	-71.6	1332	-71.5	1333	-71.4	1334	-71.3	1335	-71.2	1336	-71.1	1337	-71.0	1338

X63 — Pre-mixer gain (②) and output volume (⑤), (continued)

dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	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	-70.1	1347	-70.0	1348
-69.9	1349	-69.8	1350	-69.7	1351	-69.6	1352	-69.5	1353	-69.4	1354	-69.3	1355	-69.2	1356	-69.1	1357	-69.0	1358
-68.9	1359	-68.8	1360	-68.7	1361	-68.6	1362	-68.5	1363	-68.4	1364	-68.3	1365	-68.2	1366	-68.1	1367	-68.0	1368
-67.9	1369	-67.8	1370	-67.7	1371	-67.6	1372	-67.5	1373	-67.4	1374	-67.3	1375	-67.2	1376	-67.1	1377	-67.0	1378
-66.9	1379	-66.8	1380	-66.7	1381	-66.6	1382	-66.5	1383	-66.4	1384	-66.3	1385	-66.2	1386	-66.1	1387	-66.0	1388
-65.9	1389	-65.8	1390	-65.7	1391	-65.6	1392	-65.5	1393	-65.4	1394	-65.3	1395	-65.2	1396	-65.1	1397	-65.0	1398
-64.9	1399	-64.8	1400	-64.7	1401	-64.6	1402	-64.5	1403	-64.4	1404	-64.3	1405	-64.2	1406	-64.1	1407	-64.0	1408
-63.9	1409	-63.8	1410	-63.7	1411	-63.6	1412	-63.5	1413	-63.4	1414	-63.3	1415	-63.2	1416	-63.1	1417	-63.0	1418
-62.9	1419	-62.8	1420	-62.7	1421	-62.6	1422	-62.5	1423	-62.4	1424	-62.3	1425	-62.2	1426	-62.1	1427	-62.0	1428
-61.9	1429	-61.8	1430	-61.7	1431	-61.6	1432	-61.5	1433	-61.4	1434	-61.3	1435	-61.2	1436	-61.1	1437	-61.0	1438
-60.9	1439	-60.8	1440	-60.7	1441	-60.6	1442	-60.5	1443	-60.4	1444	-60.3	1445	-60.2	1446	-60.1	1447	-60.0	1448
-59.9	1449	-59.8	1450	-59.7	1451	-59.6	1452	-59.5	1453	-59.4	1454	-59.3	1455	-59.2	1456	-59.1	1457	-59.0	1458
-58.9	1459	-58.8	1460	-58.7	1461	-58.6	1462	-58.5	1463	-58.4	1464	-58.3	1465	-58.2	1466	-58.1	1467	-58.0	1468
-57.9	1469	-57.8	1470	-57.7	1471	-57.6	1472	-57.5	1473	-57.4	1474	-57.3	1475	-57.2	1476	-57.1	1477	-57.0	1478
-56.9	1479	-56.8	1480	-56.7	1481	-56.6	1482	-56.5	1483	-56.4	1484	-56.3	1485	-56.2	1486	-56.1	1487	-56.0	1488
-55.9	1489	-55.8	1490	-55.7	1491	-55.6	1492	-55.5	1493	-55.4	1494	-55.3	1495	-55.2	1496	-55.1	1497	-55.0	1498
-54.9	1499	-54.8	1500	-54.7	1501	-54.6	1502	-54.5	1503	-54.4	1504	-54.3	1505	-54.2	1506	-54.1	1507	-54.0	1508
-53.9	1509	-53.8	1510	-53.7	1511	-53.6	1512	-53.5	1513	-53.4	1514	-53.3	1515	-53.2	1516	-53.1	1517	-53.0	1518
-52.9	1519	-52.8	1520	-52.7	1521	-52.6	1522	-52.5	1523	-52.4	1524	-52.3	1525	-52.2	1526	-52.1	1527	-52.0	1528
-51.9	1529	-51.8	1530	-51.7	1531	-51.6	1532	-51.5	1533	-51.4	1534	-51.3	1535	-51.2	1536	-51.1	1537	-51.0	1538
-50.9	1539	-50.8	1540	-50.7	1541	-50.6	1542	-50.5	1543	-50.4	1544	-50.3	1545	-50.2	1546	-50.1	1547	-50.0	1548
-49.9	1549	-49.8	1550	-49.7	1551	-49.6	1552	-49.5	1553	-49.4	1554	-49.3	1555	-49.2	1556	-49.1	1557	-49.0	1558
-48.9	1559	-48.8	1560	-48.7	1561	-48.6	1562	-48.5	1563	-48.4	1564	-48.3	1565	-48.2	1566	-48.1	1567	-48.0	1568
-47.9	1569	-47.8	1570	-47.7	1571	-47.6	1572	-47.5	1573	-47.4	1574	-47.3	1575	-47.2	1576	-47.1	1577	-47.0	1578
-46.9	1579	-46.8	1580	-46.7	1581	-46.6	1582	-46.5	1583	-46.4	1584	-46.3	1585	-46.2	1586	-46.1	1587	-46.0	1588
-45.9	1589	-45.8	1590	-45.7	1591	-45.6	1592	-45.5	1593	-45.4	1594	-45.3	1595	-45.2	1596	-45.1	1597	-45.0	1598
-44.9	1599	-44.8	1600	-44.7	1601	-44.6	1602	-44.5	1603	-44.4	1604	-44.3	1605	-44.2	1606	-44.1	1607	-44.0	1608
-43.9	1609	-43.8	1610	-43.7	1611	-43.6	1612	-43.5	1613	-43.4	1614	-43.3	1615	-43.2	1616	-43.1	1617	-43.0	1618
-42.9	1619	-42.8	1620	-42.7	1621	-42.6	1622	-42.5	1623	-42.4	1624	-42.3	1625	-42.2	1626	-42.1	1627	-42.0	1628
-41.9	1629	-41.8	1630	-41.7	1631	-41.6	1632	-41.5	1633	-41.4	1634	-41.3	1635	-41.2	1636	-41.1	1637	-41.0	1638

X63 — Pre-mixer gain (②) and output volume (⑤), (continued)

dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	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	-40.1	1647	-40.0	1648
-39.9	1649	-39.8	1650	-39.7	1651	-39.6	1652	-39.5	1653	-39.4	1654	-39.3	1655	-39.2	1656	-39.1	1657	-39.0	1658
-38.9	1659	-38.8	1660	-38.7	1661	-38.6	1662	-38.5	1663	-38.4	1664	-38.3	1665	-38.2	1666	-38.1	1667	-38.0	1668
-37.9	1669	-37.8	1670	-37.7	1671	-37.6	1672	-37.5	1673	-37.4	1674	-37.3	1675	-37.2	1676	-37.1	1677	-37.0	1678
-36.9	1679	-36.8	1680	-36.7	1681	-36.6	1682	-36.5	1683	-36.4	1684	-36.3	1685	-36.2	1686	-36.1	1687	-36.0	1688
-35.9	1689	-35.8	1690	-35.7	1691	-35.6	1692	-35.5	1693	-35.4	1694	-35.3	1695	-35.2	1696	-35.1	1697	-35.0	1698
-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	-12.0	1928
-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

X63 — Pre-mixer gain (②) and output volume (⑤), (continued)

dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	dB Value	X63	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	-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	2023	-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
Pre-mixer gain 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

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**
- **Digital I/O Ports**
- **Emulate Mode vs. Live Mode**
- **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**, 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**, and **Connection Options** 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 via 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 left 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 left 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** 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 via any of the three control ports, RS-232, USB, or LAN.

Updates to this program can be downloaded from the Extron Web site 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
2. 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".

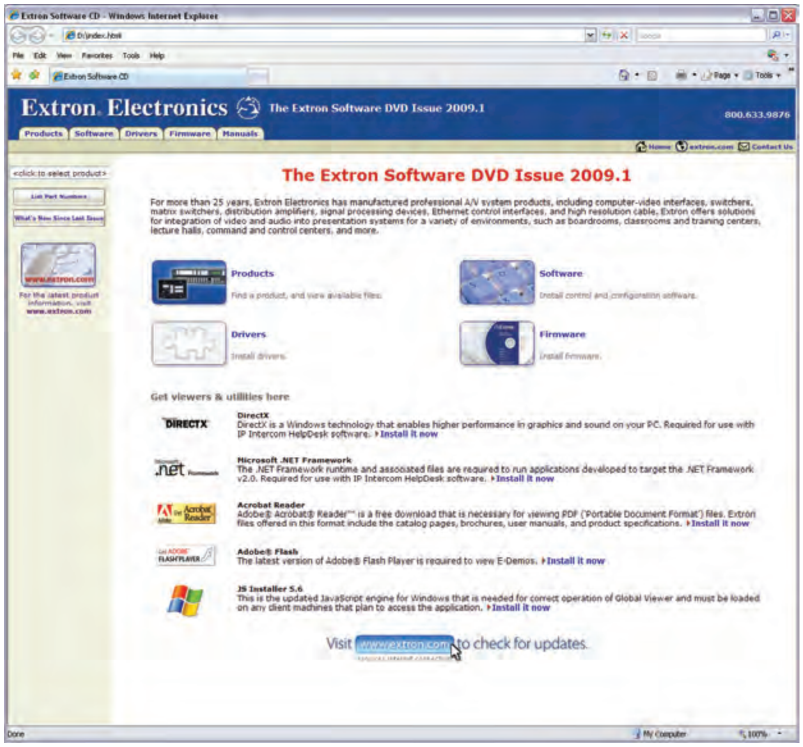


Figure 38. DVD Software Menu

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

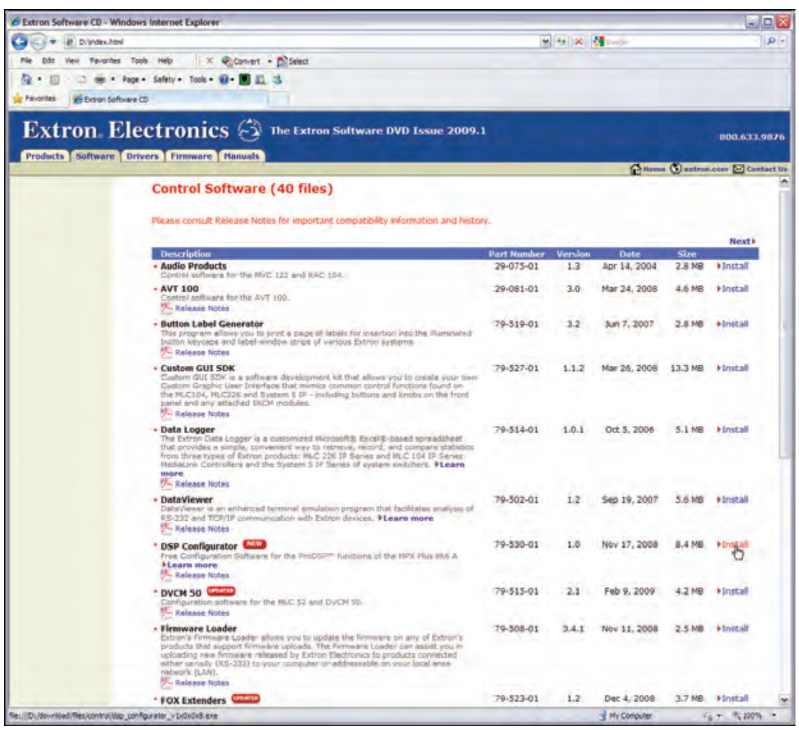


Figure 39. DVD Control Software Menu

4. Follow the on-screen instructions. By default, the Windows installation creates a C:\Program Files\Extron\DSP_Configurator folder for the DSP Configurator program files.

5. When the DSP Configurator installation is complete, the USB Installer starts automatically (figure 40). It is recommended to install the USB drivers whether they are used immediately or not.

Install the USB Driver

To install the USB driver, follow these instructions.

Figure 40. USB Installer Splash Screen

1. After the DMP Configurator program installation is complete, click **Next** to proceed.

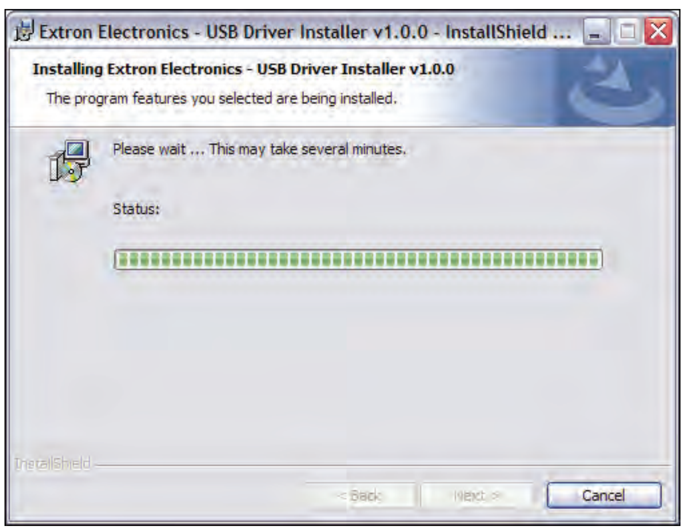


Figure 41. USB Installation

2. The USB driver installer is launched. When the installer has completed the installation of the USB drivers, the following screen appears:

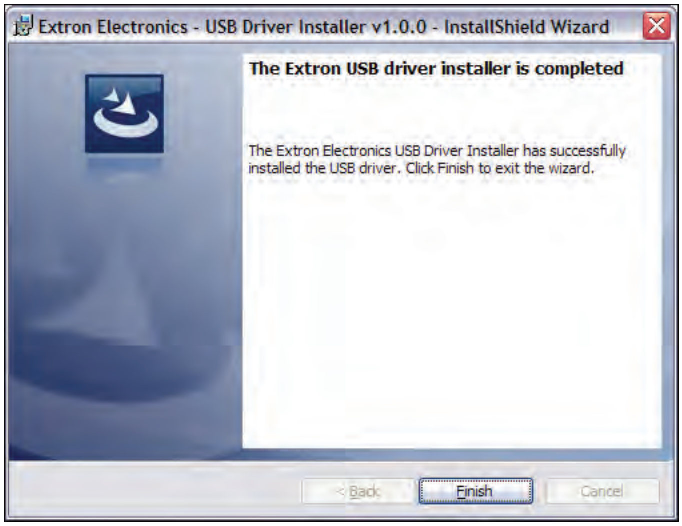


Figure 42. **Successful USB Driver Installation**

3. Click **Finish**.

USB driver installation is complete.

DSP Configurator Program

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 (figure 44). (Refer to **Emulate Mode vs. Live Mode**.)

Using the Program

In the DSP Configurator window ***Emulate*** mode, audio parameters may be selected, then transferred to the DMP 64 by going 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 vs. Live Mode**, later in this chapter.

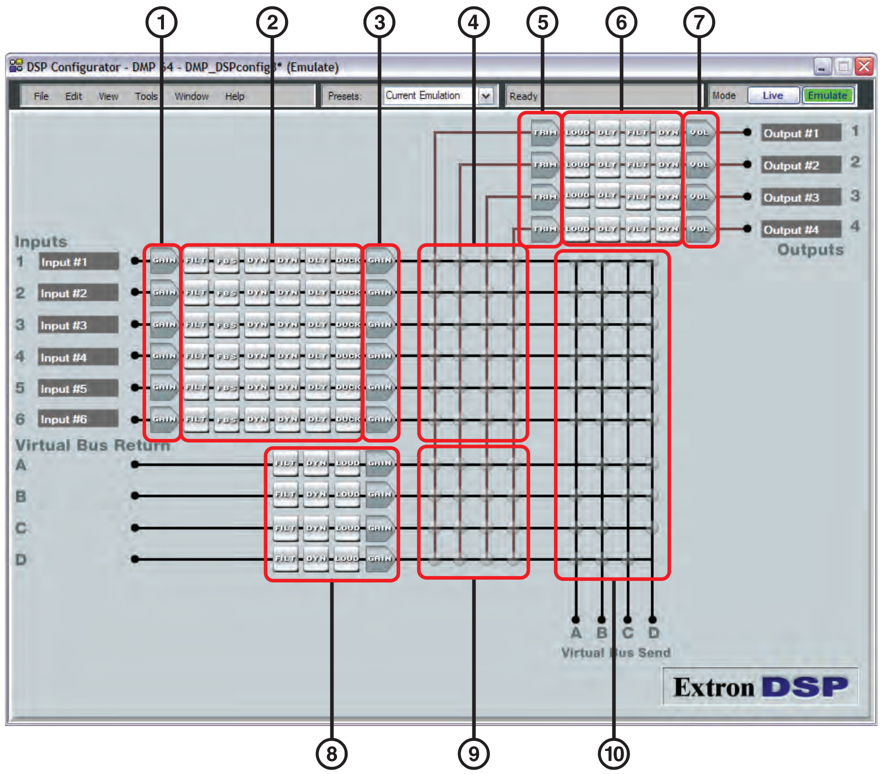


Figure 43. DMP 64 Configurator Program

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

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)

⑥ Output signal processor chain

⑦ Output volume control

⑧ Virtual Bus signal processor chain

⑨ Virtual Bus Return (primary) mix-points

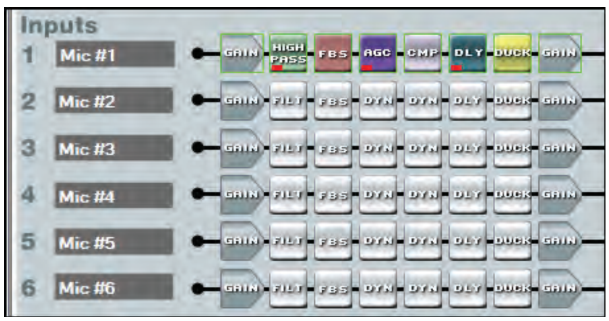
⑩ Virtual Bus Send (secondary) mix-points

Cut, Copy, or Paste Functions

The user may cut, copy, or paste a GUI processor. These actions can be performed from a context menu accessed by a right-click of the GUI element, using the Edit menu, or using the standard Windows keystrokes: <**Ctrl+X**> = cut; <**Ctrl+C**> = copy; <**Ctrl+V**> = paste.

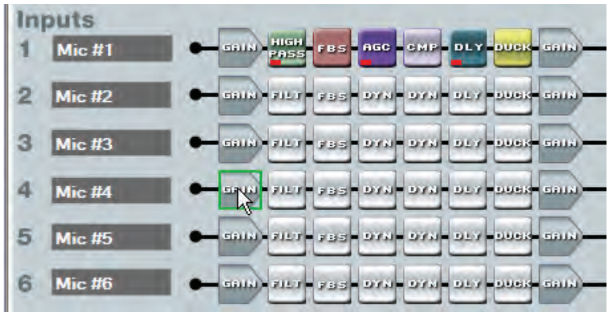
Multiple GUI elements may be acted upon but the blocks copied must be compatible with the desired paste blocks. A highlighted group of elements can be cut or copied to a clipboard. The clipboard contents may then be pasted, but will only succeed if there is an exact one-to-one relationship between the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

In the following example, the Mic #1 input signal path is copied to Mic #5. First the mouse is clicked and dragged 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.

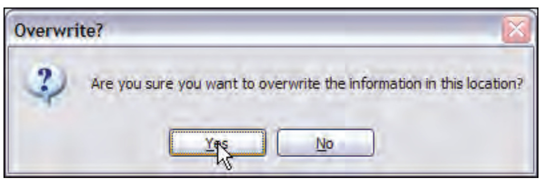


As shown below, the starting point for the paste, (the upper/leftmost element), must first be focused by left-clicking the mouse on it. Note the green focus outline that appears on the Mic #5 Gain block. The clipboard elements are pasted using the context menu **Paste** command, **Edit** menu **Paste** command, or <**Ctrl+V**>.

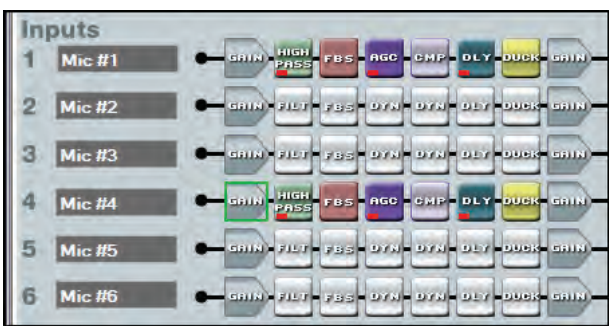
NOTE: A cut/copy of elements may be pasted to multiple locations. To copy the clipboard to an additional location, click on the leftmost block and paste again.



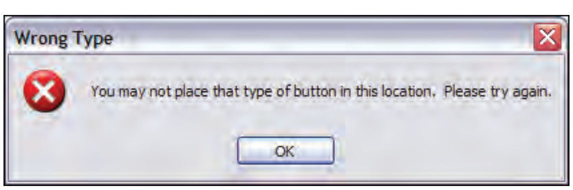
The program prompts to be certain the user understands all settings in the pasted to section will be overwritten:



Upon clicking **Yes**, the entire Mic #5 input path is now identical to the Mic #1 input path including signal levels, parameters settings and mute/bypass selections.



Any single processor block may 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 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.



Navigation

There are two methods of navigation through the GUI:

- Keyboard
- Mouse

One element in the GUI will always retain focus. When a new DSP Configurator file is opened, the upper left element (Input #1 Gain) will be focused by default.

Keyboard Navigation

All GUI 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** later in this chapter.

Mouse Navigation

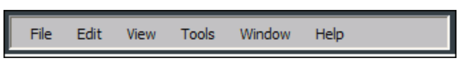
Left-click. A single left click brings focus to a processor block, as well as other GUI elements such as tabs, sliders, check boxes. Other left-click actions follow the Windows standard.

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 will open a dialog window from either the focused or unfocused state of a GUI element.

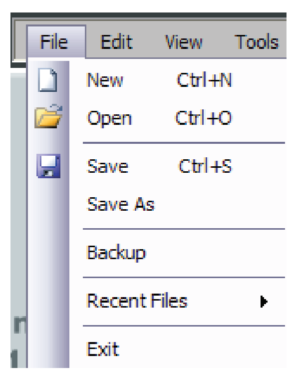
DSP Configurator Toolbar Menus

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



- File
- Edit
- View
- Tools
- Window
- Help

File

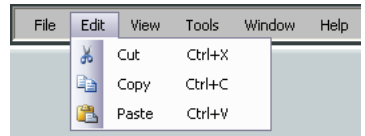


> **NOTE:** **New**, **Open**, and **Recent Files** are unavailable in Live mode.

- **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.

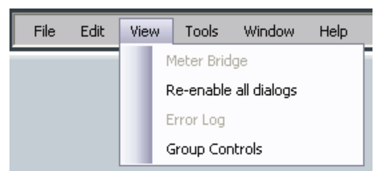


> **NOTE:** Processor blocks are not removed from the processor stream after a **Cut** and a subsequent **Paste** operation. Only the parameters are moved. Processor blocks and their parameters can be pasted only into another block of the same type. For example, the input 1 filter block and all of its 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 DSP Configurator program at the location selected.

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 via the LAN port.

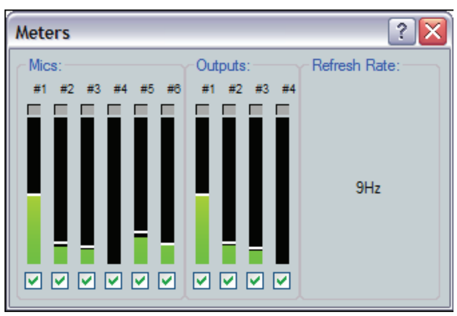
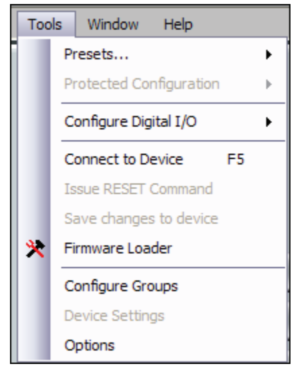


Figure 44. Meter Bridge

- **Re-enable all dialogs —** Re-enables all dialog boxes, the pop-up windows that allow changes to block parameters.
- **Error log —** A troubleshooting tool, error log lists error messages.
- **Group Controls —** Opens the Group Controls dialog box. See **Group Masters**.

Tools



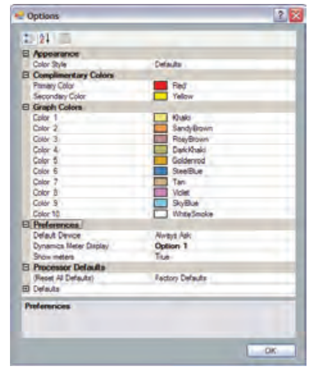
The Tools menu contains the following items and sub-menu:

- **Presets —** Provides three options:
  - o **Mark All Items —** Mark (select) all parts of the current configuration (excluding presets), including processors and mix-points to save as a partial preset.
  - o **Save Preset —** Save the currently marked processors, and mix-points as a partial preset.
  - o **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 the all 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. The 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:

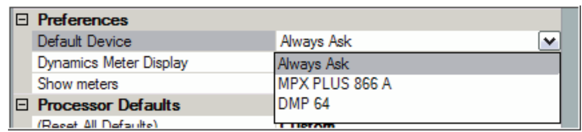
  > **NOTE:** The default PIN is 0000.

  - o **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.
  - o **Recall —** Recall the protected configuration.
  - o **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 may be used to trigger external events from DMP 64 actions, or for external events to trigger DMP actions. See **Digital I/O Ports** for details.
- **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 EscZXXX← SIS command (see **SIS Programming and Control**).
- **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 updating the firmware without taking the DMP 64 out of service. See **Firmware Loader**.
- **Configure Groups —** Opens the configure groups dialog box. See **Group Masters**.
- **Device Settings —** Live mode only. Opens a dialog box that provides a means to change the IP address, set administrator and user passwords, and to select the serial port baud rate.

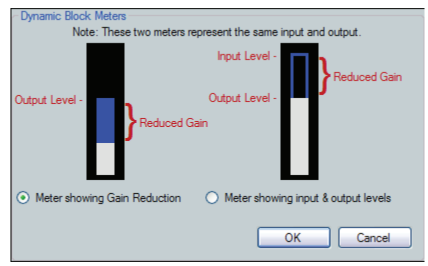
- **Options —** Opens a tabbed dialog box that allows customization of the DSP Configurator appearance and operation.



  - **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 for selection of the devices to connect to, or to “Always ask” on startup. That selection can be changed using **Default Device**.



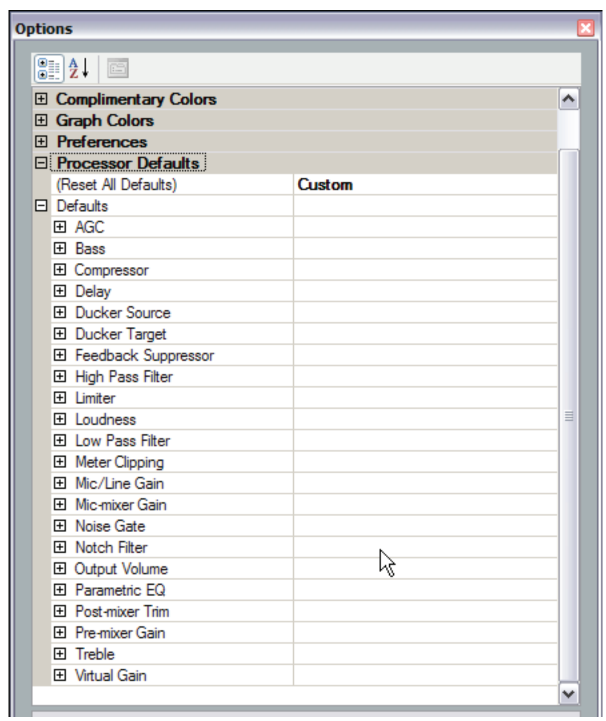
  - If **Show Meters** is set to **True**, **Dynamic Block Meters** may be used to tailor the appearance of the dynamics meters to use the full meter to show input and gain reduction, or to show the level based on the output and gain reduction.



  - **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.

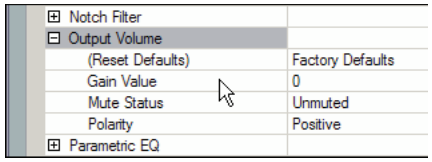
    Each row item contains default settings customized for the processor, filter, trim, or gain block it represents.

    Gain and volume blocks can be initially muted, while filter and dynamics processor blocks can be initially bypassed.

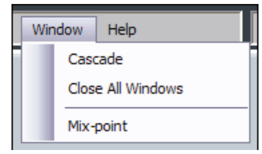


NOTE: The bypass function is labeled “Enable”.

- 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.

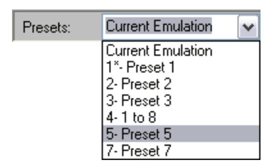
Help selection

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 via the F1 key from any main screen or dialog (which accesses context sensitive Help).

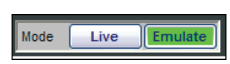
Presets Drop-down



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



Allows selection between **Live** mode and **Emulate** mode. See **Emulate Mode vs. Live Mode** 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.

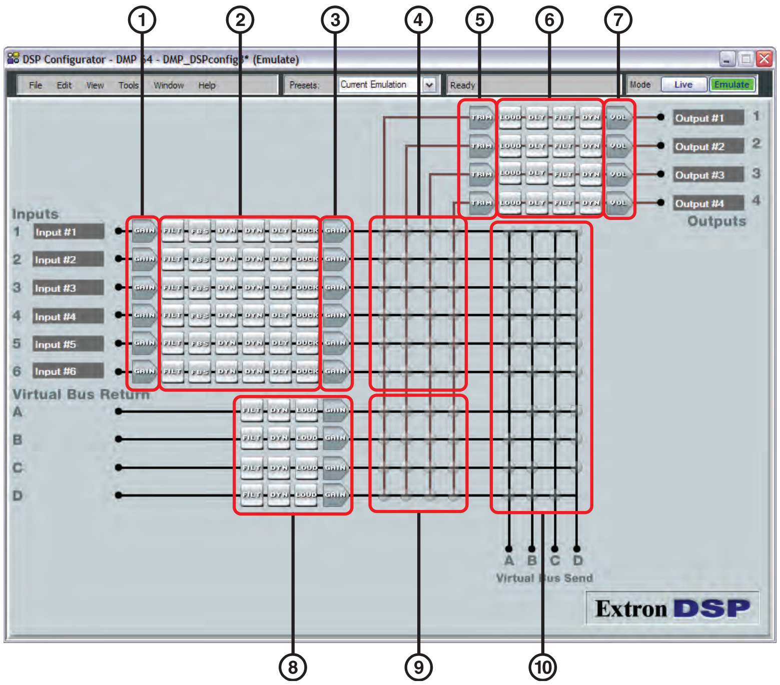


Figure 45. Control Blocks and Processor Chains

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

All blocks on the main DSP control screen have one of three main functions:

- Level control (gain/volume/trim)
- Mix-point (signal routing)
- Signal processing (filter/feedback/dynamic/delay/duck/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 (input gain), filter, feedback suppression, two dynamics, and a delay processor, then ducking and a pre-mixer gain control. The output chain begins with a level control (post-mixer trim), loudness, delay, filter, and dynamics processing blocks, and an output volume control. Each virtual bus chain has a filter, a dynamic processing block, loudness and output trim control. 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) shown in figure 45 above, consist of a series of control blocks of two basic types specific to that chain: level control (gain, trim, and volume control blocks), and signal processors (frequency filters, feedback suppression, dynamics, delay, ducking, and loudness). Both types of blocks are always present in the signal 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 46. Input Gain Control Muted, Dynamics Processor Bypassed

Gain 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 (figure 47). This action opens a dialog box that contains the fader for that control.

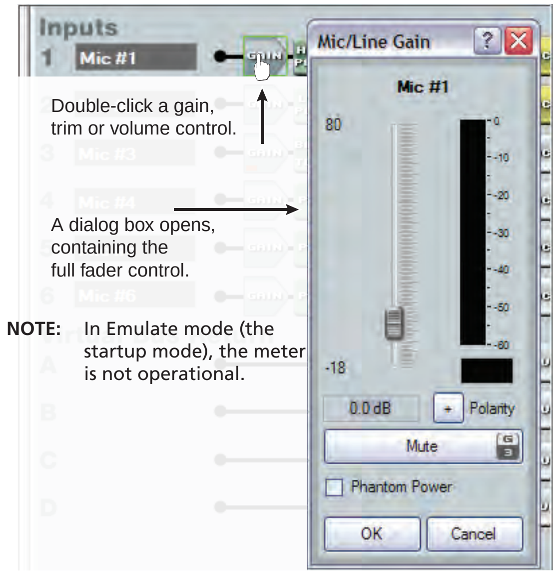


Figure 47. **Accessing a Typical Gain Control Dialog Box**

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 (figure 48). Once 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** earlier in this chapter.

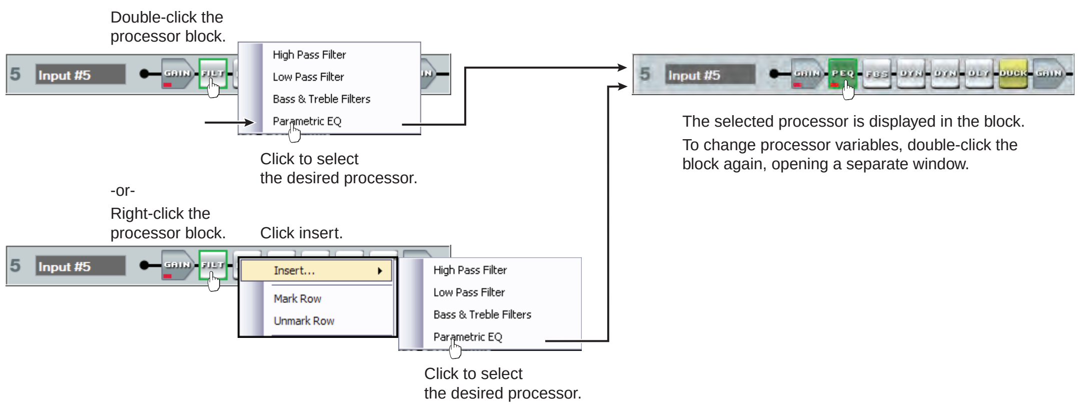


Figure 48. **Selecting a Processor Block**

Once inserted, to see 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 process (figure 49).

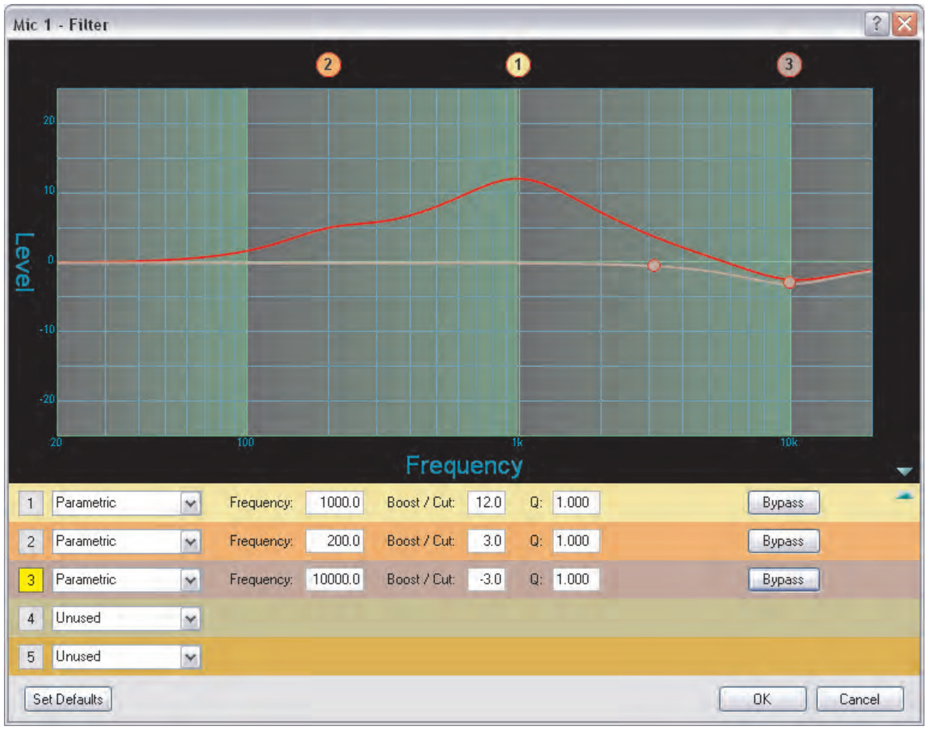
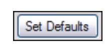
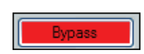


Figure 49. **Sample Processor Dialog Box**

- 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.

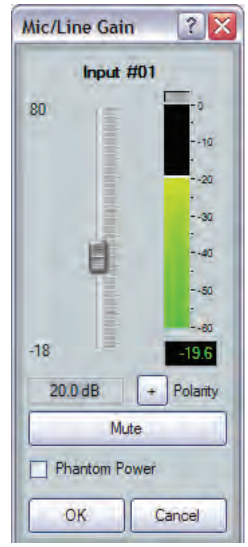
> **NOTE:** Figure 49 is a sample of one type of dialog box. Contents and appearance of each dialog box are unique to the processor type.

The block can be removed from the audio stream by selecting it with a single mouse click and depressing the keyboard Delete key or by right-clicking and selecting **Delete**.

Input Signal Controls

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

Gain Control



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 level setting 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).

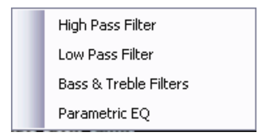
The **Phantom Power** checkbox, accessible in the dialog box, toggles the +48 VDC phantom power on and off. Phantom power is typically used to power a condenser microphone.

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

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

Also see **Mic/Line (Input) Gain**.

Filter Block

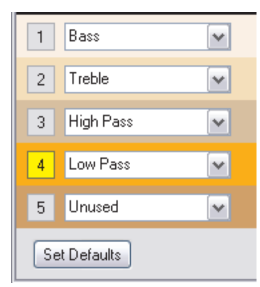


The filter processor block, when first inserted, provides one of four filter selections. Click the desired filter to select it. See **Filter** for additional information.

NOTE: Selecting "Bass & Treble Filters" inserts two separate filters.

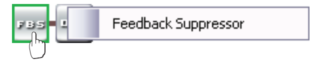
- Once inserted, double-click the processor block to change parameters of the filter.
- After the first filter is added, up to four additional filters may also be added to the filter block using the dialog box. Select the desired filter in a new row using the drop-down boxes.

The following filters are available:



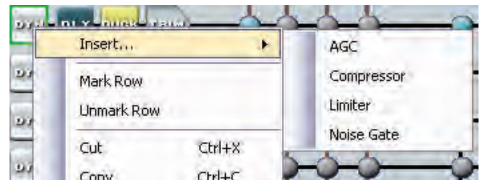
- **High pass filter —** A high pass filter passes a band of frequencies extending from a specified cutoff frequency (greater than zero) up toward the high end of the frequency spectrum. All frequencies above the specified cutoff frequency are allowed to pass, while all frequencies below are attenuated.
  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. This allows the user to control the amplitude, shift the center frequency, and determine the range of frequencies of each band.

Feedback Suppressor Block



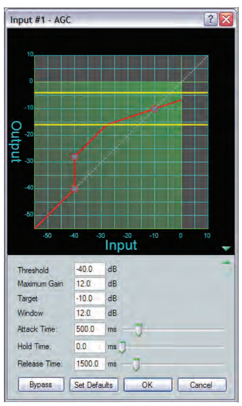
The feedback suppressor processor block, when inserted, detects feedback on a live microphone channel, and uses a set of fixed and dynamic filters to counteract the detected feedback frequencies. It is possible to achieve an additional 3 dB to 9 dB of mic gain in conditions where feedback may have otherwise prevented these levels. See **Feedback Suppressor** for additional information.

Dynamics Block (2)



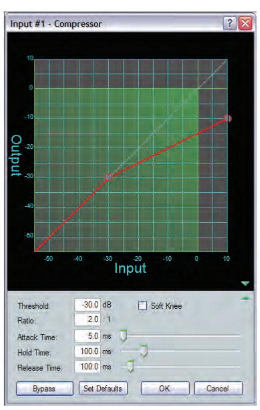
The two dynamics processor blocks, when inserted, each provide one of four dynamic processors. A dynamic processor alters the dynamic range of an audio signal, the difference between the loudest to the quietest portions of the signal. Click on the desired dynamics processor to select it. See **Dynamics** for additional information.

After selection, parameters can be changed in the dialog box, customized to the processor, by double-clicking the processor block.



- **Automatic Gain Control (AGC) —** AGC adjusts signal gain relative to the strength of the incoming signal to achieve 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 level approaches the target level it receives less gain or no gain at all. Once the signal level reaches the target level all gain is removed.
- A **window** range, indicated by the two yellow lines, is also applied 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.

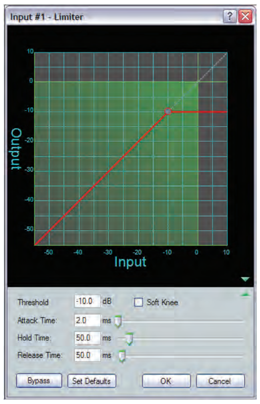
  The default threshold is -40 dB. The default target level is -10.0 dB. The default gain and window are 12.0 dB.



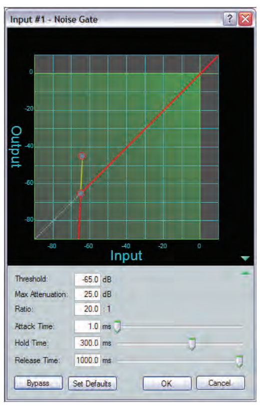
- **Compressor —** The compressor regulates signal level by reducing, or compressing, the dynamic range above a specified threshold. The signal input level to output level ratio determines the reduction in the dynamic range above the threshold setting. In the adjacent example, with a ratio setting of 2.0:1, once the threshold (-30 dB) level is exceeded, for every 2 dB of input above that level, the compressor outputs 1 dB.

  Compression is commonly used to keep mic levels within an acceptable range for maximum clarity. A compressor can also make softer sounds louder by reducing the dynamic range and raising the output level (referred to as "make-up gain"), or by increasing the input signal and reducing the louder portions of the signal. This has the effect of making louder portions of a signal softer. Similar to a limiter, compression also can be used to protect a system or a signal chain from overload.

  The default threshold is -30 dB and default ratio is 2.0:1.



- **Limiter —** The limiter regulates the level of the input signal by severely restricting its dynamic range above a specified threshold. Doing so prevents clipping, protecting a system against component or speaker damage. The limiter is closely related to the compressor but applies a much higher compression ratio, expressed as ∞:1, at a much higher signal level (default is -10 dB). The ratio cannot be changed.



- **Noise Gate —** The noise gate is an expander working on signal levels below the set threshold level. It applies an expansion ratio to all signals below the threshold making those signals softer. Picking a threshold level just above the noise floor of a system reduces background noise and allows stronger signals, (above the threshold), to pass untouched. Using a high ratio of 20:1, the noise gate closes the audio path below the threshold, eliminating background noise, and opens the path above the threshold allowing the signal to pass; hence the term **noise gate**.

  The default threshold is -65 dB. The ratio is 20.0:1.

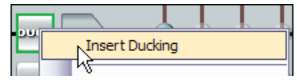
Delay Block

The delay processor block, when inserted, provides a means to delay the audio signal. The processor can delay the audio using either time or distance as a determiner. The default delay, when inserted, is 100 ms, using the time function. The delay settings can be changed by double-clicking the processor block. Distance can be specified in feet or meters. Temperature can be set in either degrees Fahrenheit or degrees Centigrade. The processor calculates changes in the speed of sound for the specified temperature. See **Delay** for additional information.

Pre-mixer Gain Control

The post-input processing gain control, also called the pre-mixer gain, provides boost or cut adjustment of the signal level with a range of -100 dB to +12 dB in 0.1 dB increments.
The default setting is unity gain (0.0 dB).

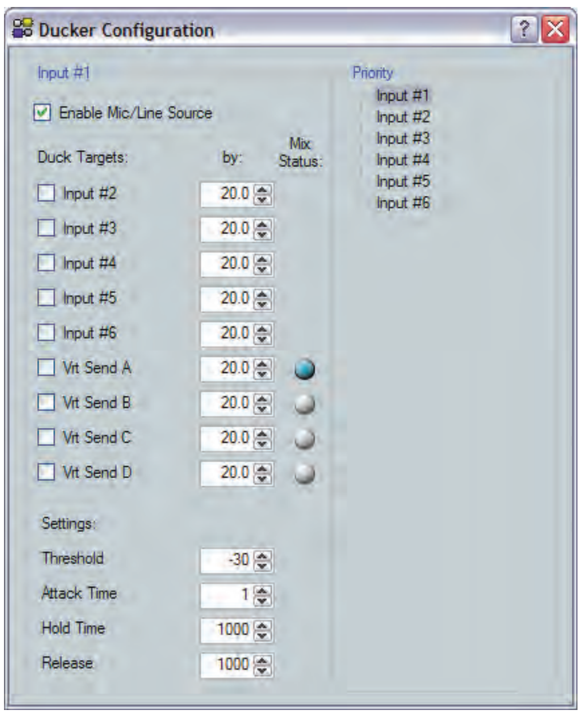
Ducking Block



The ducking processor block, when inserted, provides a means to **duck**, or lower the level of one or more microphones and/or 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 ducked mic original level once the other signal has ceased. Ducking is 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 and/or program material.
- A paging mic needs to attenuate all other signals.

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



When the first ducking source is inserted (shown above), no ducking targets are selected. The teal buttons next to the virtual send rows indicate the input signal has been routed to the virtual sends, which can also be ducking targets. All ducking targets must be selected manually.

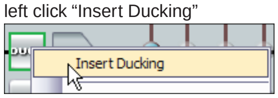
> **NOTE:** Signal reduction is not cumulative. Ducking will only reduce an input by the amount set in the **by:** text box even if it is being ducked (see **Ducking and Priority Ducking** later in this section) by another ducking source.

Also see **Ducking** for additional information.

Ducker Tutorials

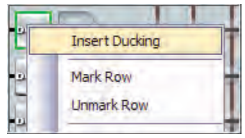
The examples below are based on different input configurations. Insert a ducker from a ducker processor block using one of the following methods:

Double-click the block, then left click “Insert Ducking”



-or-

Right-click the box to open context menu, then left-click “Insert Ducking”



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

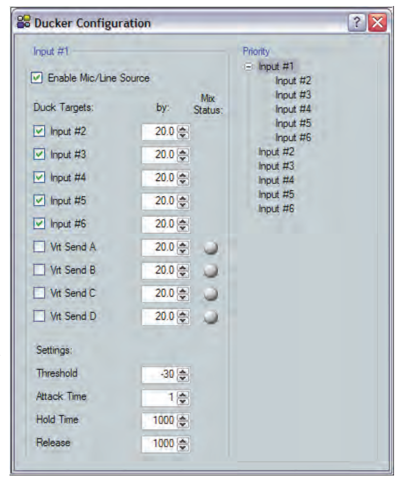
Ducking and Priority Ducking

The first inserted microphone will duck 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 #2-6 are the ducking targets.
Any signal on input #1 that exceeds the ducking threshold will now duck inputs 2-6.

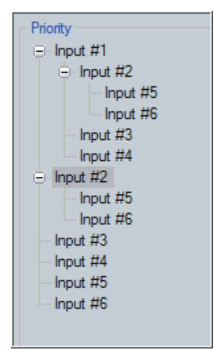


The ducking processor also provides a means to have an additional input duck other targets using the Priority feature. The second input ducks its selected duck targets, and can also be ducked by the first ducking source.

To set an additional ducking source:

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

> **NOTE:** Since it was previously selected as a ducking target, Input #1 will not be available as a target.



2. Open the ducking dialog window for the input and select the desired duck targets. In this example inputs #5-6 are the ducking targets of input #2.
Any signal on input #2 that exceeds the ducking threshold will now duck inputs 5-6.
The ducking targets may 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 2-6 will still be ducked regardless of whether the signal on input #2 exceeds its ducking threshold.
No input will be ducked more than 20 dB.

Output Signal Processing Block

The output signal processor chain, post mic mixer, makes adjustments to program audio material tied to a specific output.

Post-mixer Trim Control

The post-mixer trim control provides a fader for fine adjustment of the program material prior to the output signal chain. The trim control has a range of -12 dB to +12 dB in 0.1 dB increments. The default setting is unity gain (0.0 dB).

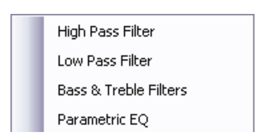
Loudness Block

The loudness processor block, when inserted, applies a filter compensation curve to the signal in an inverse relationship to the output volume control setting; the higher the output volume setting, the less compensation is applied. See **Calibrating Loudness** to fine-tune the loudness compensation.

Delay Block

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**. Typically the near speakers would be delayed so that audio delivery time matches the speakers further away.

Filter Block

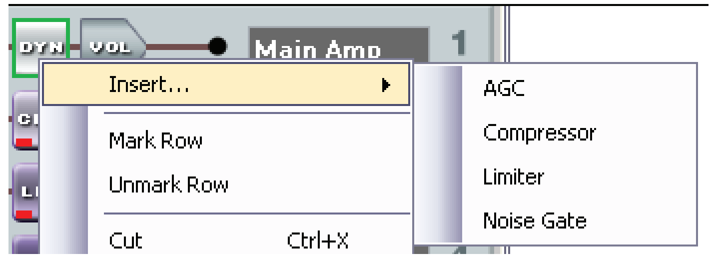


The filter processor block, when first inserted, provides one of four filter selections; High Pass, Low Pass, Bass & Treble filters and Parametric EQ. Up to nine filters can be added to each filter block. The output filter block is identical to the input filter processor block except that up to nine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

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

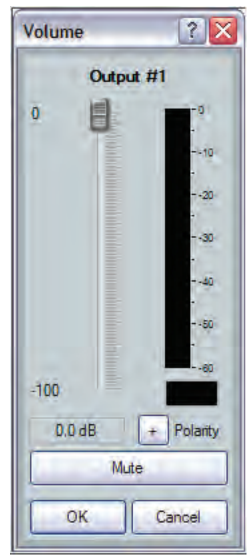
Dynamics Block

A dynamics processor block, when inserted, provides one of four dynamics processors. The available processors are identical to the processors available on the input in the dynamics processor block and described in **Dynamics**.



Volume Control

The output volume control provides output 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 audio output to be silenced. This control is identical to the input audio mute control. When the audio output is muted, 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.

See **Line Output Channels** for additional information.

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 are 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. (See below, **Group Controls | Tools** for information on using increment/decrement controls within the DSP Configurator software.)

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 still 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 50, next page). 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.

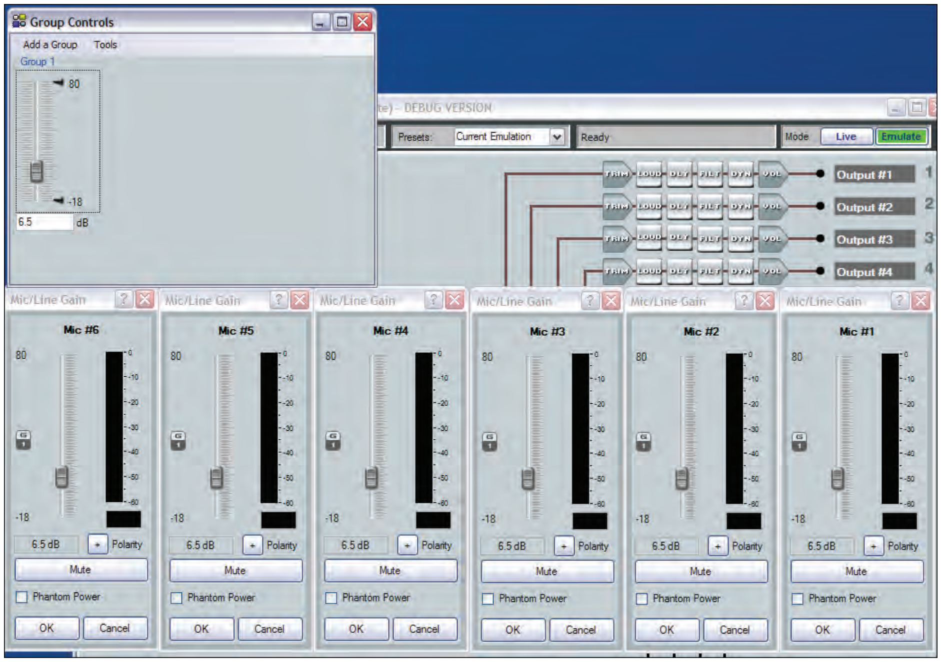


Figure 50. Sample Fader Group Master and Associated Gain Controls

Mute controls within the blocks can also be grouped (figure 51).

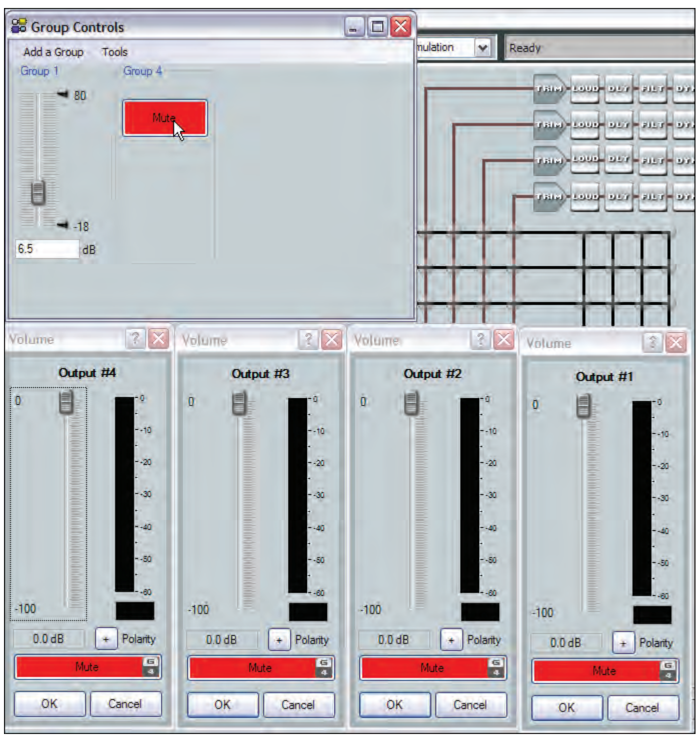


Figure 51. Sample Mute Group Master and Muted Outputs

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 (figure 52). The list defaults to the first empty group. Select an empty group if necessary, or select an existing group to overwrite.

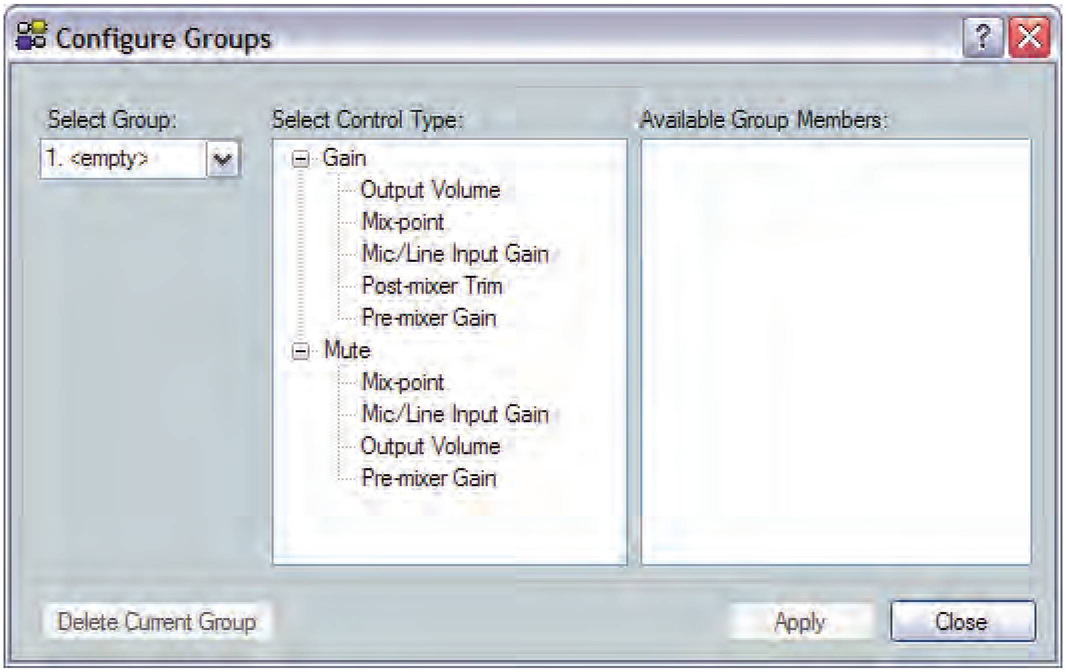


Figure 52. Configure Groups Add Group Dialog Box

> **NOTE:** **<empty>** groups have no group members assigned. Numbered groups (such as **<Group #1>**) have controls assigned that may 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 **4** that are already assigned to a different group are displayed in **blue**.

4. In the **Available Group Members** section, make appropriate selections by clicking the checkbox(es). When a + sign exists, click to expand the tree and select individual controls. Up to 16 group members may 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 **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 (figure 53) which displays all current group master controls. This window can be resized for convenience. The group controls dialog contains two menu items:

Add a Group **Tools**

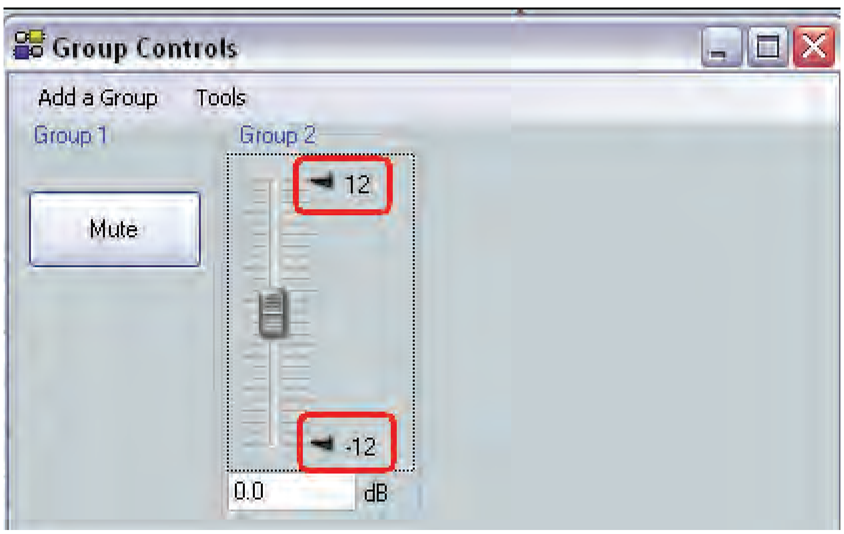


Figure 53. Group Controls Dialog Box

- 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/or floor for the group.

> **NOTE:** The 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.

- Click the **Mute** button in a mute group to mute or unmute all blocks in the group.

> **NOTE:** If a block is muted, 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. All soft limits will also be 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 this value is not remembered in the device.

To use the Increment/Decrement Simulator:

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

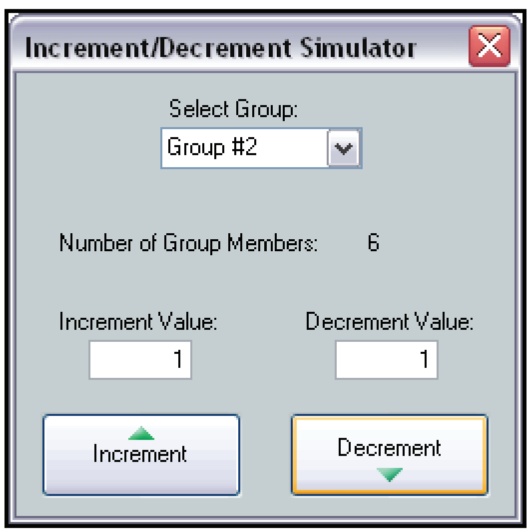


Figure 54. Increment/Decrement Simulator Dialog Box

> **NOTE:** The **Number of Group Members:** readout indicates the number of controls to be 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:** If set, soft limits cannot be exceeded.

Group Details Report

Click **Tools > Group Details Report** to create a Microsoft Word file that details all created groups (figure 55).

GROUP DETAILS REPORT

Group #1

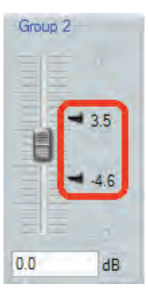
Processor Type: Output Volume
Current Mute status: Unmuted
Current Group Members:
Main Amp (Output#1) Left Channel
Stage Mixer (Output#2) Right Channel
House Video (Output#3) Left Channel
Prgm Record (Output#4) Right Channel

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 55. **Sample Group Details Report**

Soft Limits



Each gain type control provides upper and lower soft limits that can be used to 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 clicking on, then 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 + Home>`** moves limit to the current fader position. **`<Ctrl + End>`** moves limit to lower default.

Digital I/O Ports

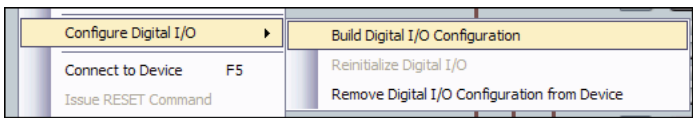
The DMP 64 provides six digital I/O ports that may be used to trigger external events from DMP 64 actions, or for external events to trigger DMP 64 actions. The DSP Configurator software provides pre-configured scripts with a fixed set of common trigger/event combinations. When selected, the script is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

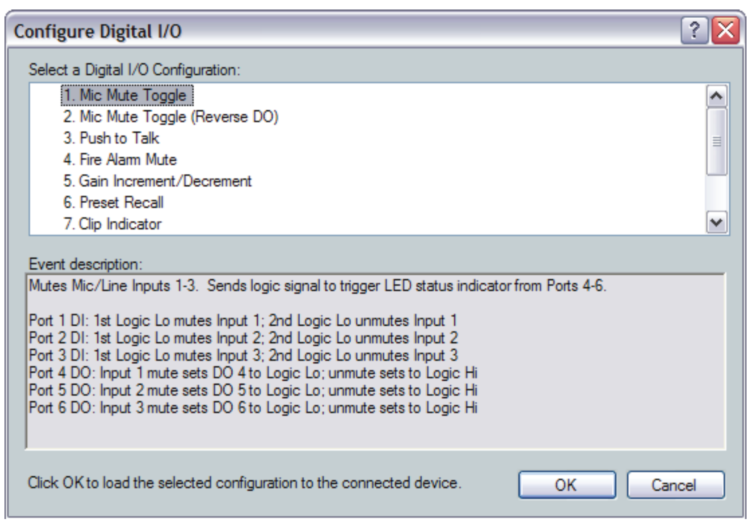
A DO (digital output) sends a Logic Lo as less than +1 VDC and a Logic Hi as +5 VDC. For every script that involves a DO, two versions are available to provide either a Logic Hi or a Logic Lo response to any action. The alternate script is designated as **"Reverse DO."**

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

1. From the tools menu, click **Configure Digital I/O**, then select **Build Digital I/O Configuration**.



2. This brings up a dialog that allows selection from a list of pre-configured scripts.
3. 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**.



4. 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 will appear between step **3** and step **4**. The DSP Configurator will connect and then disconnect during the procedure, returning to Emulate mode when completed.

Reinitialize Digital I/O

Should the script stop running for any reason, go to **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 press **OK**.
2. If the DSP Configurator is connected to a device, the I/O configuration will be removed. If it is not connected, a connection dialog box will appear.
3. Make certain the connection information is correct, then press **OK**. The I/O configuration script will be removed and a confirmation dialog box will appear.

Emulate Mode vs. *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. 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. 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 vs. Push

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 via the Ethernet LAN port when using DSP Configurator.

2. Click the Mode **Live** button (see ② in figure 56). The **Communication Type Selection** window appears.

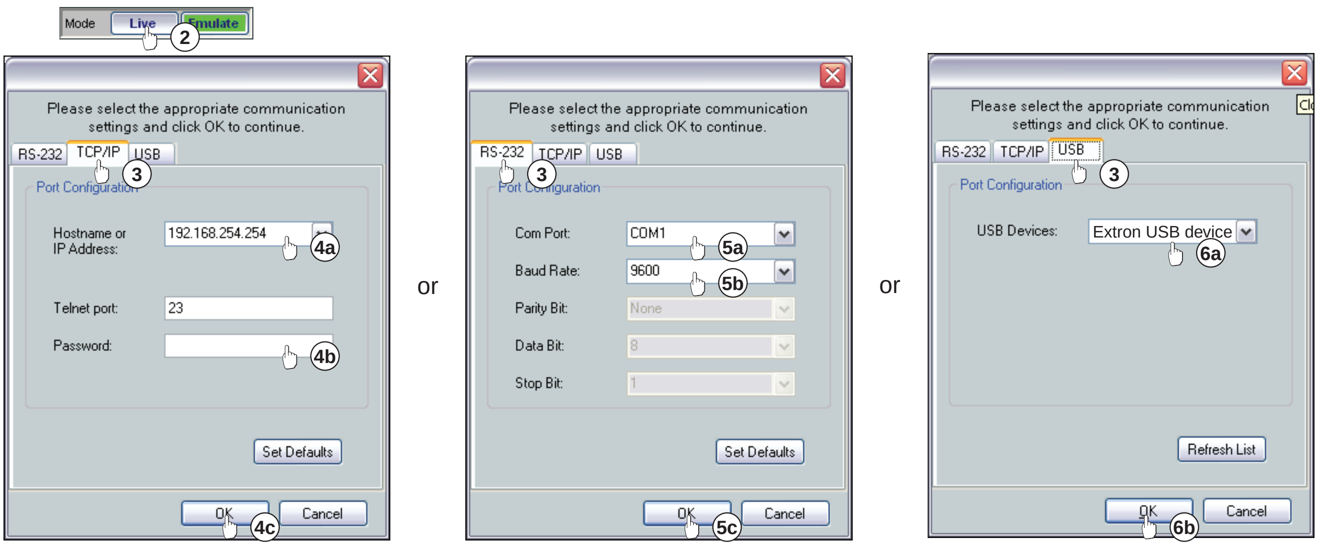


Figure 56. Selecting Live Mode

3. **If necessary** and as desired, click either the:
   - **TCP/IP tab** (for connection via the LAN port (**preferred**) — proceed to step **4**,
   - **RS-232 tab** (for connection via either of the rear panel RS-232 ports — proceed to step **5**,
   - **USB tab** (for connection via the front panel configuration port — proceed to step **6**.
4. **If TCP/IP was 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 field is correct**, proceed to step **4b**.
      - **If the address is not correct,** either click in the IP Address field and enter the IP address or click on the scroll down button (▾) to open a drop-down list and select from among the recently used addresses. Proceed to step **4b**.

      > **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.

   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 (figure 57) appears. Proceed to step **7**.

5. **If RS-232 was selected in step 3**:

   a. Click the com port drop-down menu and select the PC's comm port that is connected to the rear panel RS-232 port.

   b. Check the baud rate displayed in the comm 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 (figure 57) appears. Proceed to step **7**.

6. **If USB was 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 (figure 57) 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**

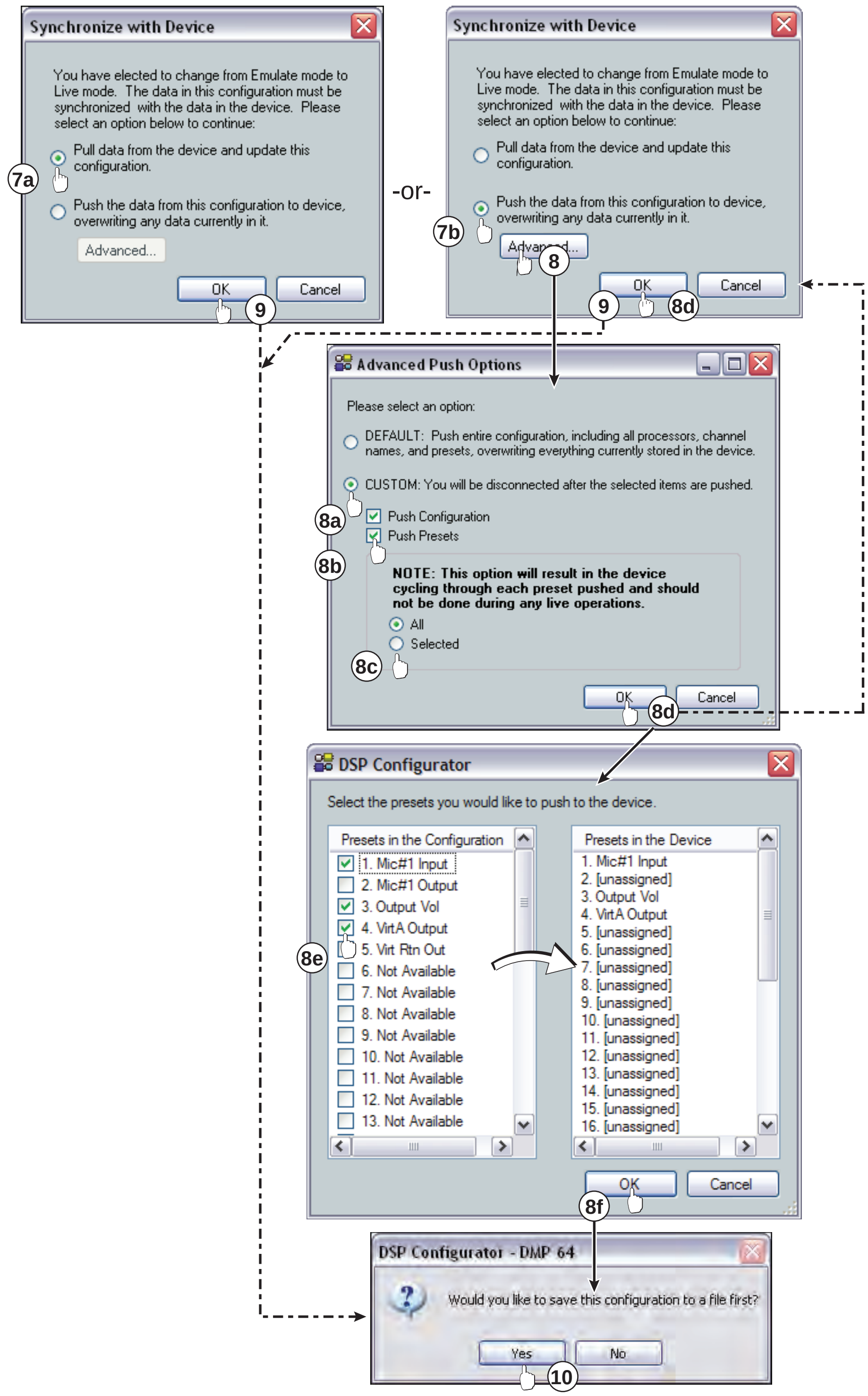


Figure 57. Selecting Live Mode, continued

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 checkbox(es); **Push Configuration** and/or **Push Presets**. 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(es).
   f. Click **OK**. — Proceed to step **10.**
9. Click **OK**. The DSP Configurator program is connected live to the device and the processors, and presets are pushed or pulled as selected, completing the selection of **Live** mode.
10. If changes were made to the DSP parameters (including mix-point, gain and/or processor blocks) since the last file save, the DSP Configurator prompts to save the file. Click **Yes** or **No**, as desired,

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

    The configuration and/or presets will be uploaded to the DMP 64.

Presets

Presets are used to recall a group of frequently used settings. Presets created by the DSP Configurator may 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, then uploaded as a set and stored to the device and/or stored to 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/rename or save to disk.

When recalled, a preset will only overwrite 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.

An additional preset, Preset 0, contains current state information if reading from the device in Live mode, or the state of the DSP Configurator if in Emulate mode (titled current emulation). Current emulation can be a configuration not yet saved as a preset (work in progress), the last preset or combination of presets recalled within the DSP Configurator, or the current state of the device as a result of switching from Live to Emulate mode.

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

When a **pull data** synchronization method 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**). 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 may also be saved and recalled via the embedded web page. Presets saved via the web page contain input gain, output volume, and the primary mix-point settings.

Previewing/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.

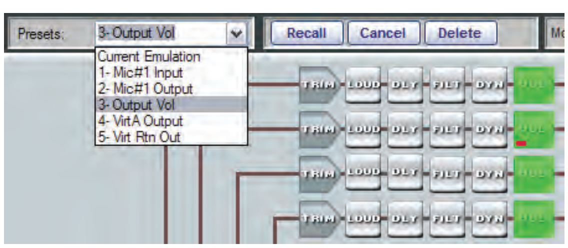


Figure 58. **Preset Preview**

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, and 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 will not be reflected in the GUI while previewing a preset, and the user cannot alter GUI elements. To apply the preset, the user clicks **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. The user clicks **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 that are highlighted (given focus) will be saved as a preset. Ctrl + A will highlight all elements within the DSP Configurator.

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

1. **<Left click>** on the desired block to select a single block,
2. **<Ctrl + left click>** to select multiple blocks that are not adjacent,
3. **<Shift/hold + click>** on the first block and click on the last block in either a vertical column or horizontal row to select multiple blocks, and
4. Click and drag a selection rectangle to select multiple adjacent blocks in either the vertical or horizontal direction.
5. Go to **Tools | Presets** and select **Mark All Items** or press <**Ctrl + A**>. This will mark all elements within the DSP Configurator, which will save a “full” preset,
6. To save the selection(s) see “Save Preset” below.

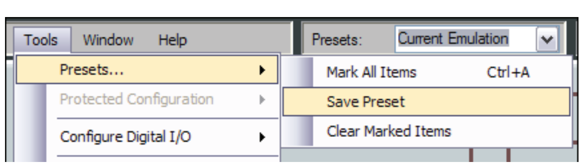
Save Preset

A preset may be saved in either emulate mode or live mode.

Saving a preset in emulate mode stores that preset in the currently open file. The DSP Configurator file must then be saved to disk via **File menu | Save** (recommended), and/or pushed to the device after a connection is established. This differs from live mode where the 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 block(s) by using left click, <**Ctrl + left click**>, <**shift + left click**> or drag around the desired blocks.
2. Select **Tools | Presets | Save Preset** in the main structural menu.



3. Select a preset number. In the Preset Name box, 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."

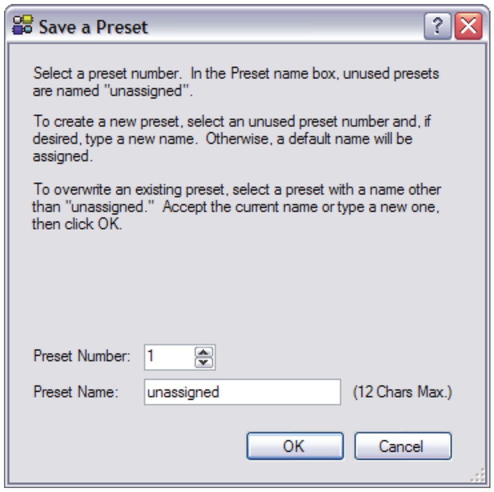
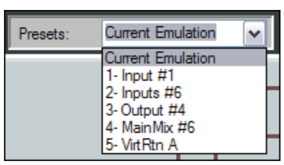


Figure 59. **Save Preset**

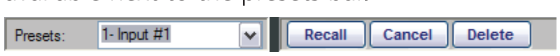
4. Click **OK** to save the preset, or **Cancel** to stop the save preset operation.

Managing Presets in the GUI

Once the preset is created (whether or not the DSP Configurator file is saved) it will appear in the preset list, available from the DSP Configurator screen.



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** will first apply the currently displayed preset elements ("marked" elements) from the stored preset and overwrite that portion of the current state, then switch the drop-down list to read "Current State."

In emulate mode, the **Recall** action button will apply the currently displayed preset elements ("marked" elements) from the file and 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 being previewed, in either live or emulate mode, the **Delete** button is available. In live mode, the preset is deleted from the hardware, which will be reflected in software (it will be 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. 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 live or emulate mode, the **Cancel** button will defeat the preview action and return 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 has been recalled. The DSP Configurator pulls the names of the presets only. 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 the DSP Configurator in live mode have no asterisk. Presets with no asterisk can be saved to disk.

Protected Configuration

Protected Configuration is a configuration secured with PIN protection. The protected configuration can be recalled by any user, but can only be written/overwritten using the assigned (up to) 4-digit PIN. Utilities for save/recall/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, and are unavailable in Emulate 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 to create a new one.

Recall Protected Configuration

The dialog informs the user, "Recalling the protected configuration will overwrite all audio and video settings currently in the device. Are you sure you want to continue?" Click **OK** to continue or **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

Keyboard Navigation

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

When the program is started, the cursor focus is in the mic/line input gain block.(① on figure 62). The input 1 gain block is highlighted green [GAIN]. When the tab button is pressed, the cursor focus toggles to the next area outlined by red boxes in figure 62.

At any time the navigation arrows may be used to move one block right, left, up or down with no boundaries.

NOTE: The callout numbers for figure 60 are not the same as figure 43.

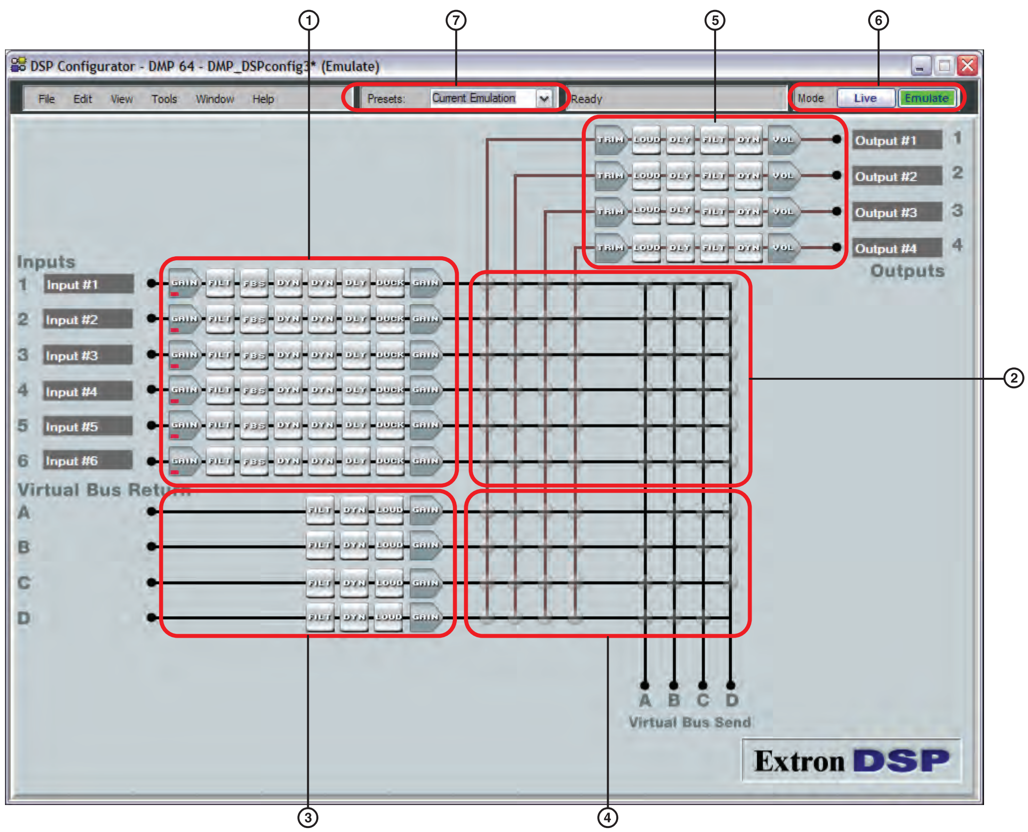


Figure 60. **DSP Configurator Program Window**

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 62.
- **Enter Key —** Performs the same action as a mouse double-click. For example, opens the context menu from which a processor type may be selected or opens 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 A/V matrix block nodes.
- **Alt key —** The Alt key 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 (**F**ile, **E**dit, **V**iew, **T**ools, **W**indow, or **H**elp) become underlined. Press the underlined letter key to open that menu.
- Once a task bar menu is open, use the up and down arrow keys to move up and down in the menu or submenu, use the right 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 ↓) key(s) 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 (←, →, ↑, and ↓) key(s) to navigate away from the selected block.
   b. To highlight additional sequential blocks, **continue to hold** the Shift key, then use the arrow (←, →, ↑, and ↓) key(s) to navigate away from the selected block. Additional blocks will be highlighted as long as the 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** the **Ctrl** key, then use the **arrow** (, , , and) key(s) 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 unhightlighted.

   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** > **Enter** (figure 61).

7. The **Save a Preset** dialog box appears.

   a. Tab to the preset number field and type a specific preset number.

   b. Tab to the preset name field and type a preset name.

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

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

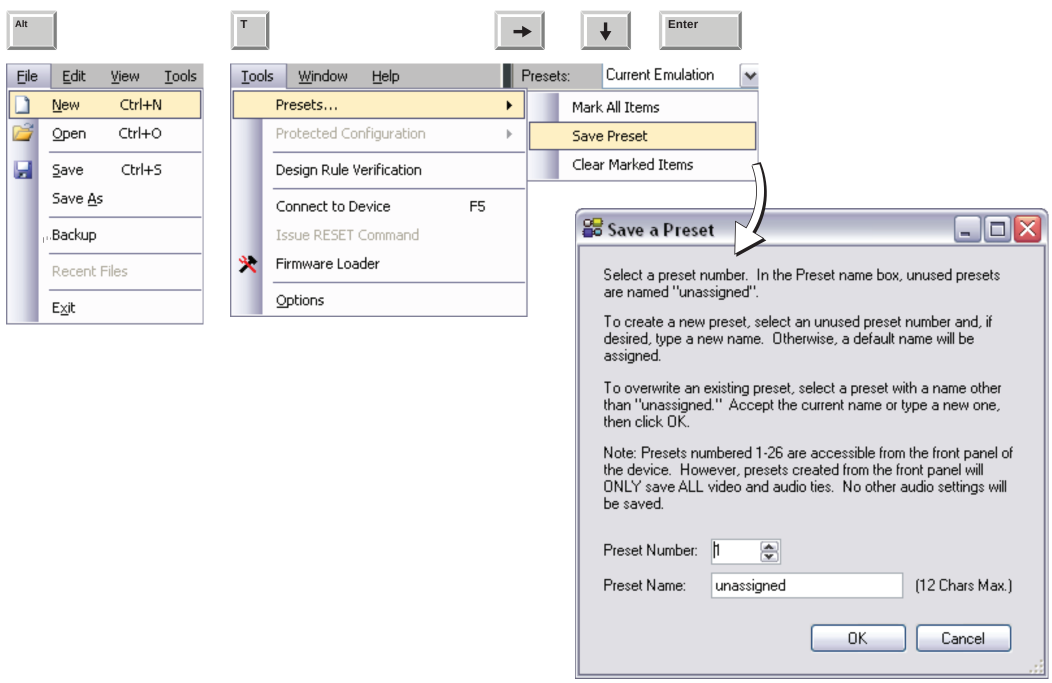


Figure 61. Saving a Preset using Keyboard Navigation

Optimizing Audio Levels

The DMP 64 uses floating point DSP technology, processing data using a combination of 32- and 64-bit algorithms. The ADCs (analog to digital converters) and DACs (digital to analog converters) sample at 48kHz, 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 and before the DAC. 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 actual clipping provides a safety net, allowing the user to reduce input gain before clipping actually occurs. This “safety net” may 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 the figure 62.

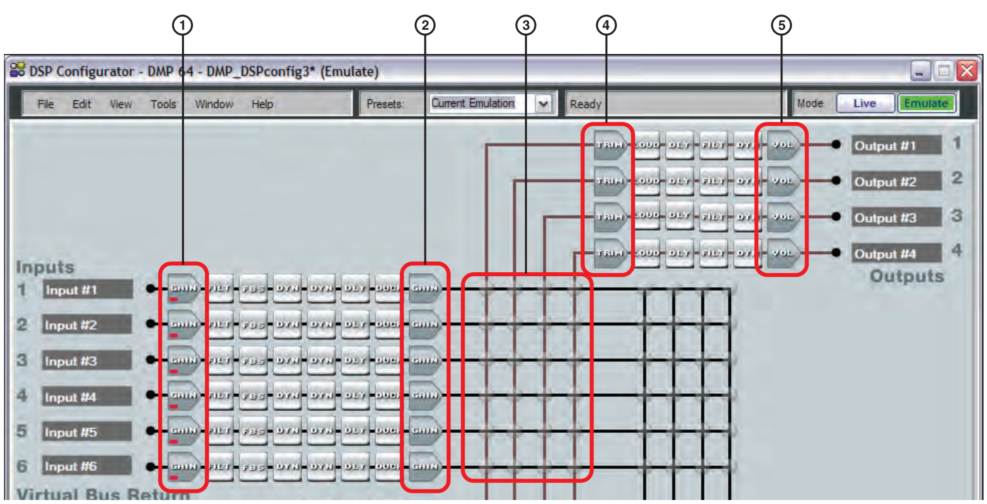


Figure 62. Gain, Trim and Volume Controls

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 will be controlled. The output volume of the DMP 64 may be controlled by either of the following two gain blocks:

- 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 will maintain 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. 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 level (①) so the meters reach approximately -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 possible 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. If the maximum output setting provides gain, then back off 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 may be advantageous. Route the audio that will carry program material from the source to the speakers in the room being set up. With the output volume control (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 will set the amplification/volume nominal level of the system, and if desired, allow listening while making adjustments. Adjust or mute the volume control as necessary. See **Setting Volume Control for the Amplifier Stage**.

Adjusting Pre-mixer Gain

After setting input gain, add desired processors into the input signal chain. The pre-mixer gain control (②) may be used to compensate for level changes due to processing. Adding a compressor generally reduces the signal level, while a filter may boost or cut the overall signal level. If 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.

Open the line input gain (①), output volume (⑤), and pre-mixer gain (②) dialog boxes. Set output volume to 100% (mute if necessary). With program material (or pink noise) present at the input, adjust the pre-mixer gain so the meter level on the input gain dialog matches the meter level the output volume dialog. This will maintain the audio at an optimal level in the input signal chain.

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

When using the pre-mixer gain for output volume control, the procedure may 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 a filter may boost or cut the overall signal level and adding a compressor generally reduces the signal level. Inserting either or both may 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 will boost the overall signal level, but at lower volume settings

After adding processors to the output signal chain, the output volume level may clip when set to 100% (or less). Floating point DSP allows clipping to be overcome by lowering the output volume (⑤) 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 will also contribute to possible clipping at the outputs, and may need to be 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. Double-click the mix-point (③) for mic/line 1 – Output 1 to open the dialog for that mix-point. Unmute the mix-point to place that signal into the mix. The default level is 0 dB, or unity gain.

Open the Input 1 gain (①) dialog. With a microphone attached to input #1 and gain set to 0 dB (turn on phantom power if the mic requires it), unmute the channel. While testing the mic, raise the fader level until the mic is clearly audible. The amount of gain and the meter level reading will vary at this point, but as a general guideline the input gain level should be at approximately 40 to 50 dB, and the meter averaging somewhere around -20 dBFS. Ideally, audio should 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 may be necessary.

Adjusting Trim

This is where setting gain structure becomes a balancing act. The following sections provide guidelines, but it may 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.

Open the output volume (⑤) and post-matrix trim (④) dialog. Set output volume to 100% (mute if necessary). With program material (or pink noise) present at the input, 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 when the input attenuator is fully open. If using the output Volume control (⑤) of the DMP 64 to control volume levels to ensure clipping does not occur at the amplifier, turn down the input attenuator of the power amp the equivalent of 18 dB (22 - 4 = 18). That puts the amplifiers input level at -14 dB (+4 – 18 = -14). If the amplifier setting (when the output volume controls of the DMP 64 are at maximum) is too loud for the room, it may need to be reduced further. If it is not loud enough for the room, a more powerful amplifier may be required.

It is recommended to use the output volume or post-mixer trim control on the DMP 64 for controlling output volume. If using loudness processing on the unit, it will only work in conjunction with these controls.

If 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 -18 dB. This is another way that clip points of the two devices will be 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.

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

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 (figure 63).

Figure 63. **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 above conditions is true**, the device downloads the factory-installed default startup page, "nortxe_index.HTML" (figure 64), also known as the System Status page.

Status Tab

System Status Page

The System Status page (figure 64) 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.

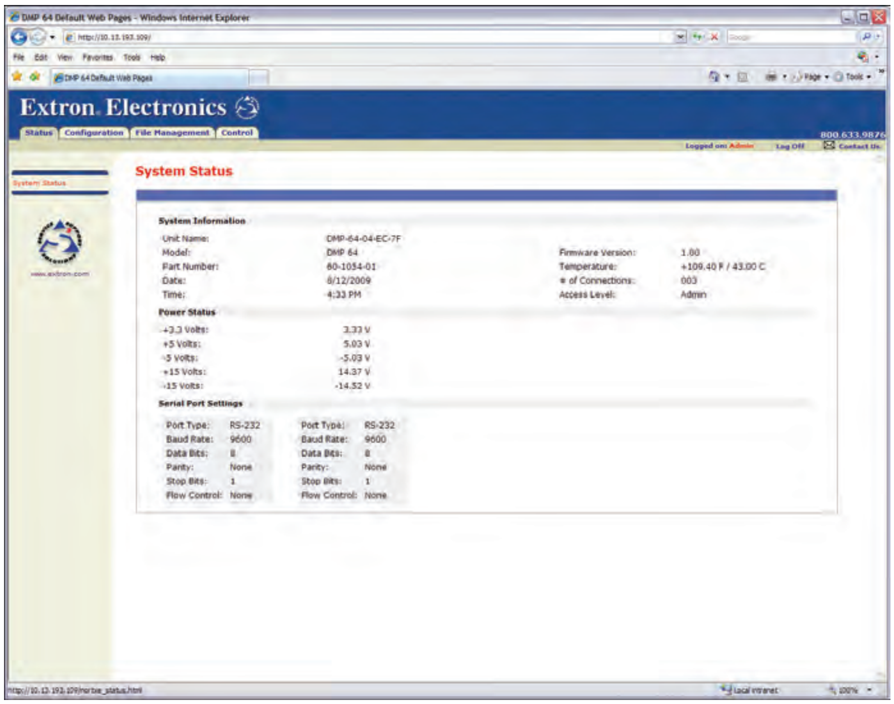


Figure 64. System Status Page

Configuration Tab

System Settings Page

Click the **Configuration** tab to download the System Settings page (figure 65). 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. See **Ethernet (LAN) Port**, for basic information about IP addresses and subnetting.

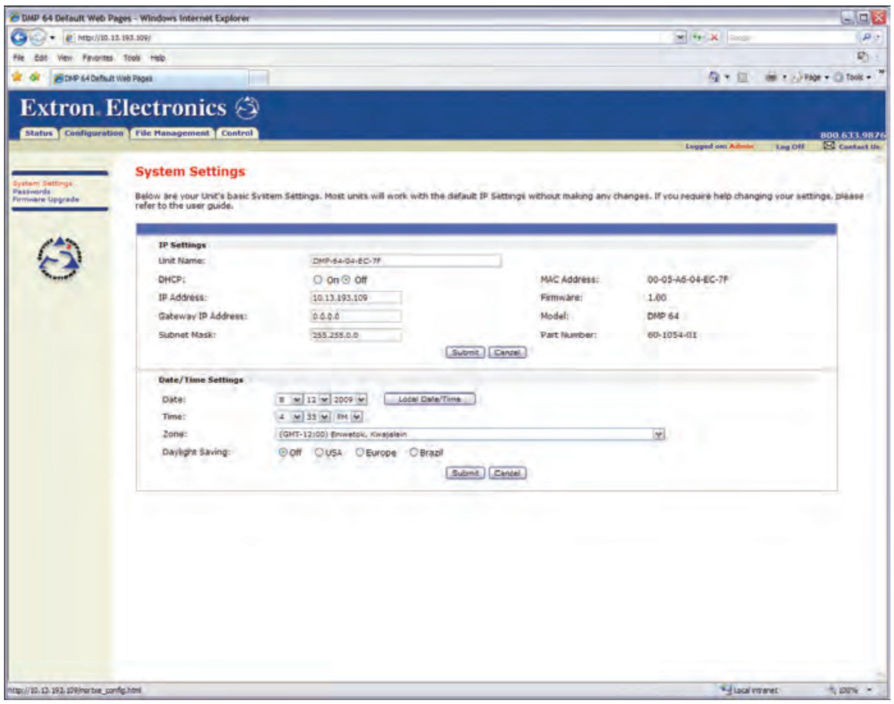


Figure 65. System Settings Page

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

- Ethernet connection to the device, either entering SIS commands (see **SIS Programming and Control**) or using the Extron DSP Configurator Program (see **DMP Software**) 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.

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 Radio Buttons

The **DHCP On** radio button directs the device to ignore any entered IP addresses and to obtain its IP address from a Dynamic Host Configuration Protocol (DHCP) server (if the network is DHCP capable). The **DHCP Off** radio button 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 3 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 (figure 66) provide a location for viewing and setting the time functions.

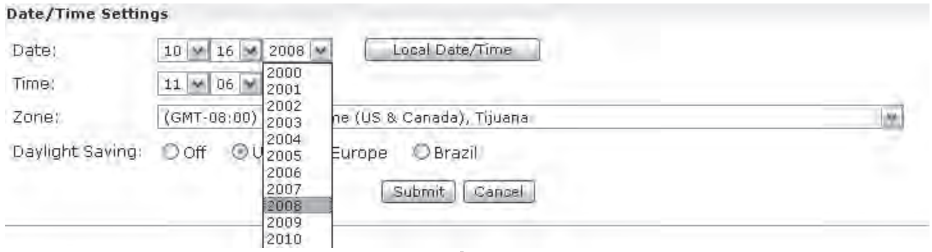


Figure 66. **Date/Time Settings Fields**

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 scroll box appears (the year drop box is shown selected in figure 66).

2. If all variable selections are not visible, click and drag the slider or click the scroll up button or scroll 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 in the **Daylight Savings** radio button 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 the **Submit** button.

Passwords Page

Access the passwords page (figure 67) by clicking the **Passwords** link on the system settings page.

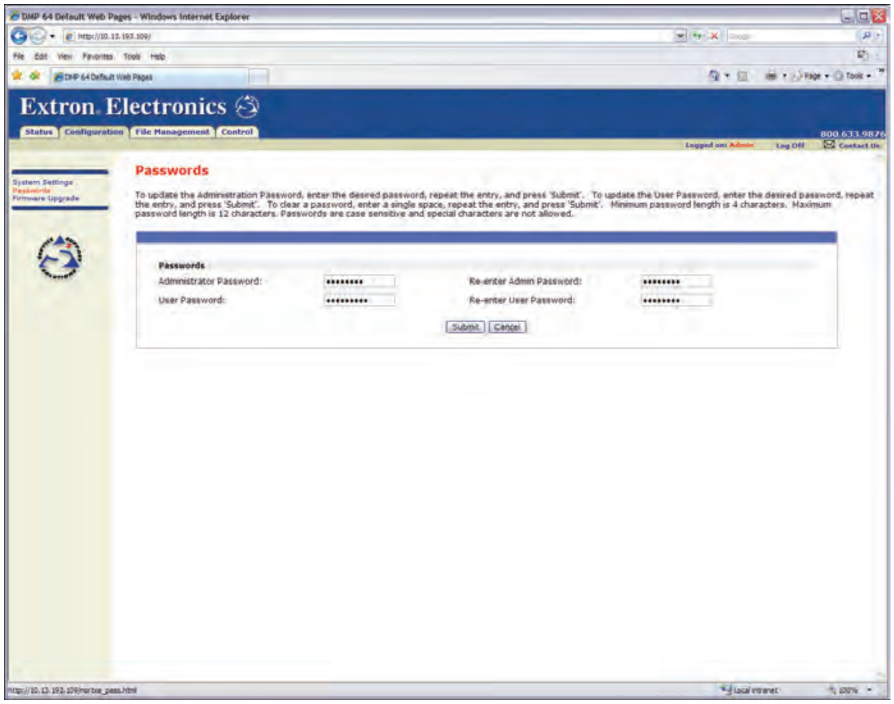


Figure 67. Passwords Page

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

> **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 password and re-enter password fields and click the **Submit** button.

Clear a Password

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

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. Access the Firmware Upgrade page (figure 68) by clicking the **Firmware Upgrade** link on the System Configuration page.

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

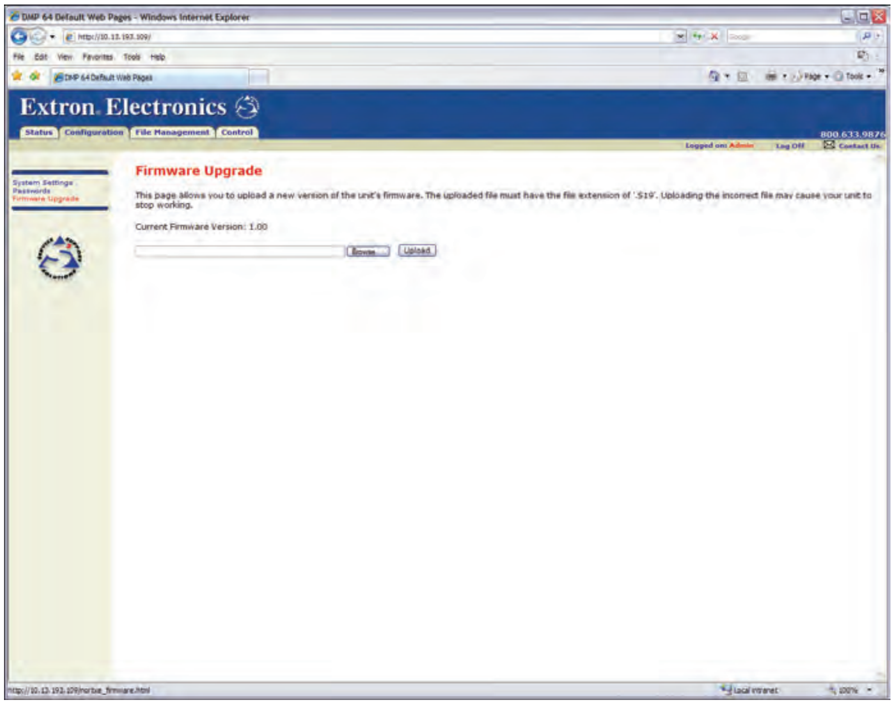


Figure 68. Firmware Upgrade Page

Update the device firmware as follows:

> **NOTE:** The Firmware Upgrade page is ***only*** for replacing the firmware that controls device operation. To insert custom HTML pages, see **File Management Page**.

1. Visit the Extron Web site, **www.extron.com**, and click the **Download Center** tab.
2. Click the **Firmware** link (figure 69 on the next page).
3. Select the appropriate firmware file to download and click **Download**.
4. Enter the requested information.
5. Click **Download** to copy the firmware to your computer.

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

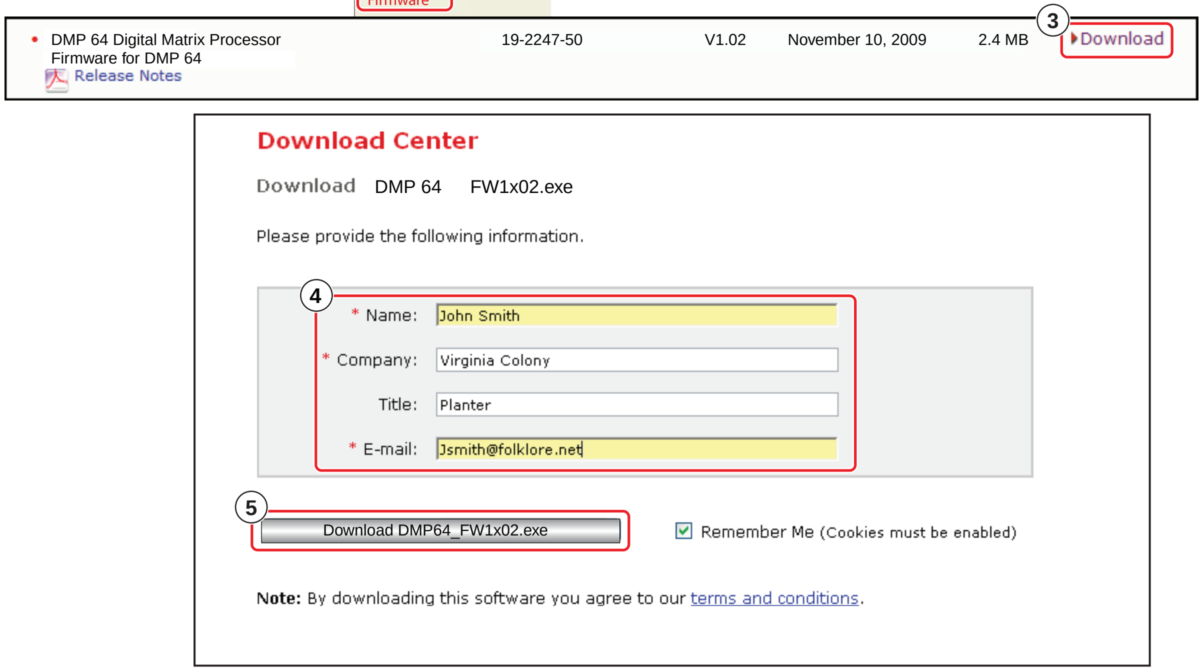


Figure 69. Location of Firmware Upgrade Files on the Web Site

6. Click **Run** twice (⑥ in figure 70). The PC downloads the firmware update from the Extron Web site and starts the installation program to extract the firmware file.

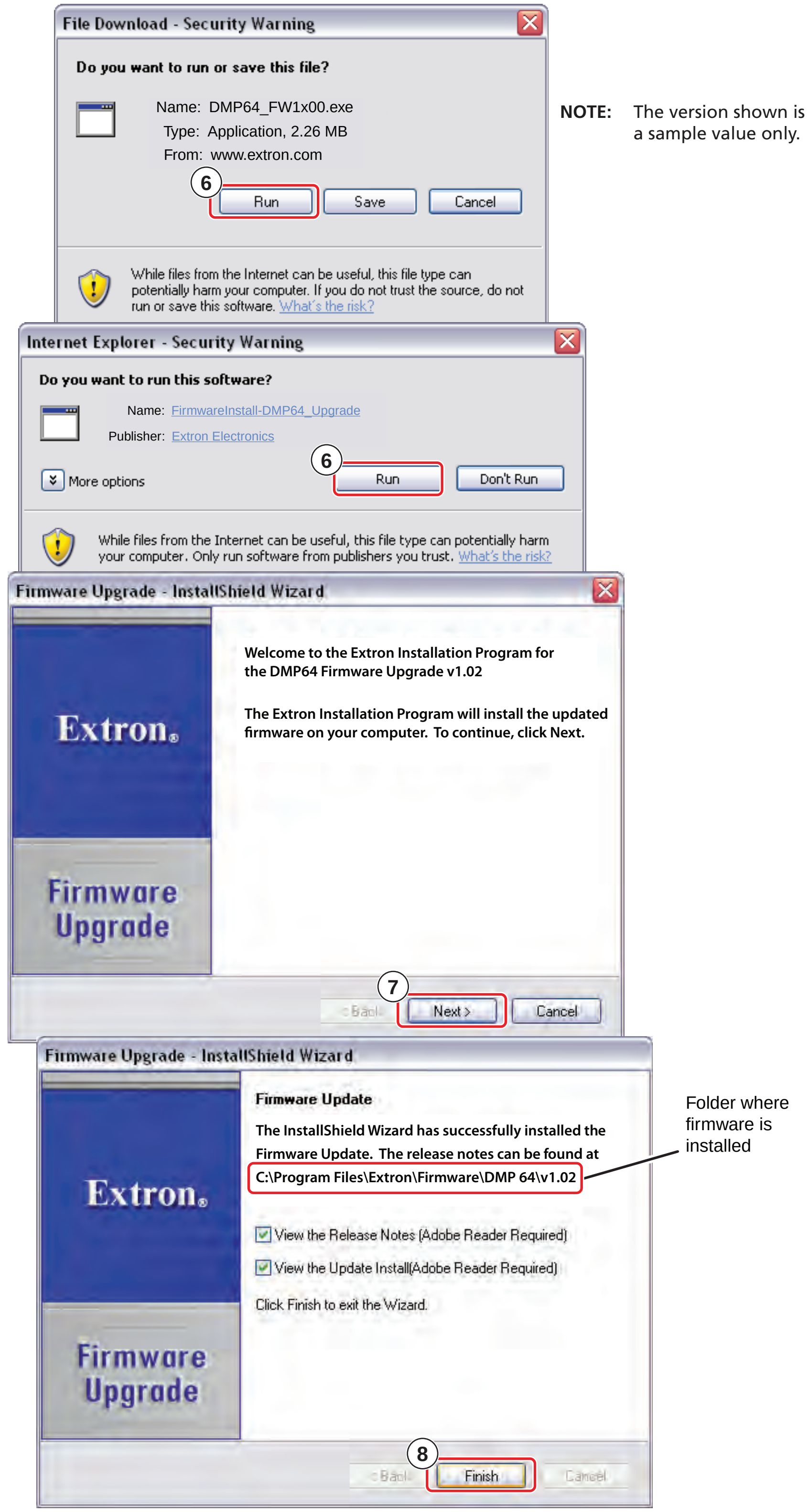


Figure 70. **Downloading Firmware Upgrade Files**

7. Click **Next** (⑦ in figure 70 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** (⑧ in figure 70) 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** .)
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 the **Open** button.
16. Click the **Upload** button. The firmware upload to the device may 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 (figure 71).

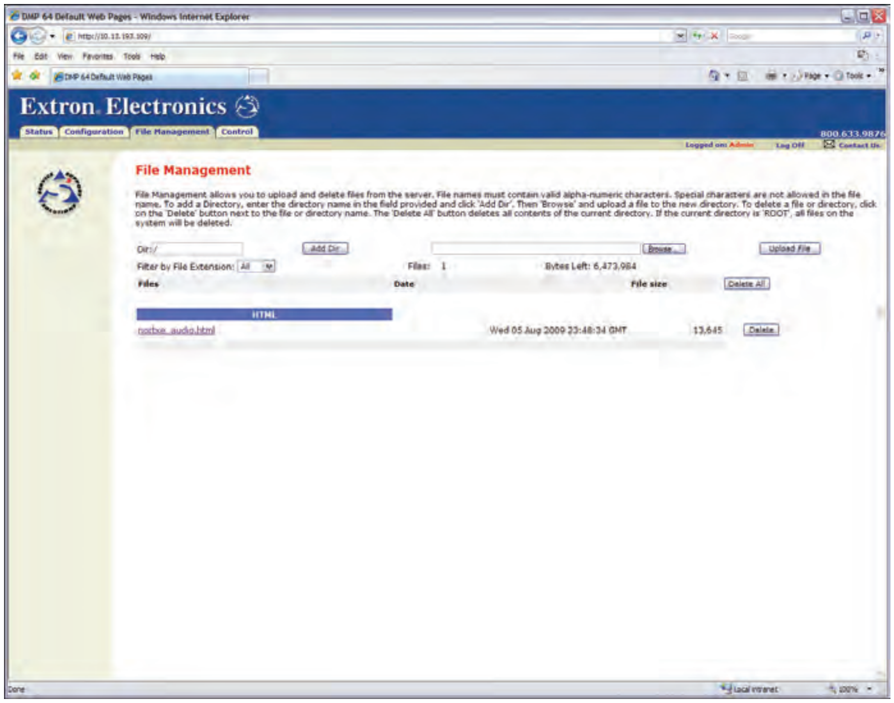


Figure 71. **File Management Page**

NOTE: The files listed in figure 71 are shown for example only.

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

Upload your own files 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 file(s).

   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 file(s) 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/unmute. Access the **Audio Settings** page (figure 72) by clicking the **Audio Settings** link on the control page.

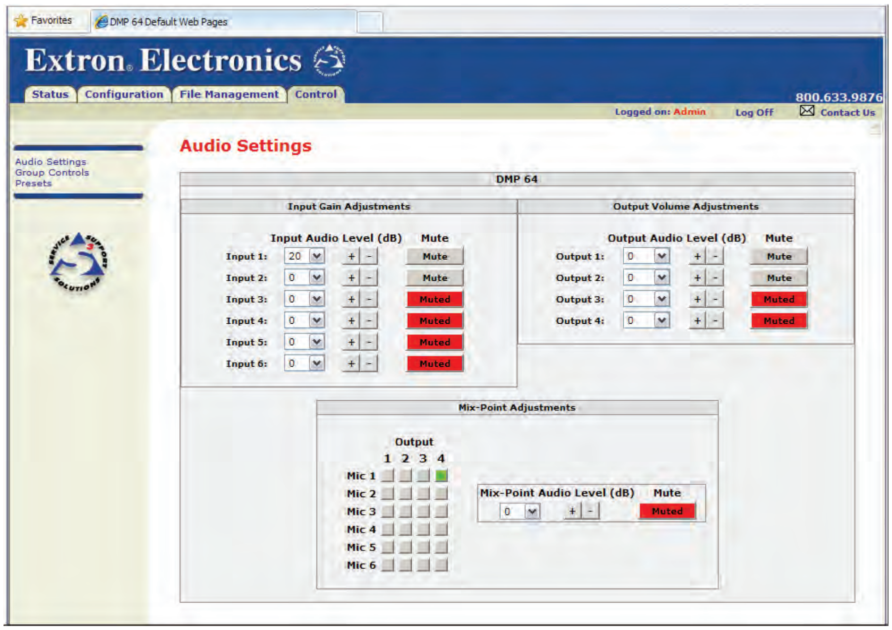
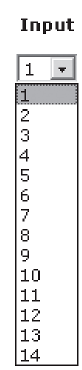


Figure 72. Video and Audio Settings Page

Change the Input Gain and Attenuation

Users can set each input's level of audio gain or attenuation (-18 dB to +80 dB) 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 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 1-3 for each input.

Mute and Unmute Inputs and Outputs

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

Mute and umute 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/output combination. The selected mix-point turns green. (Mic 1 to Output 4 is shown in figure 73).

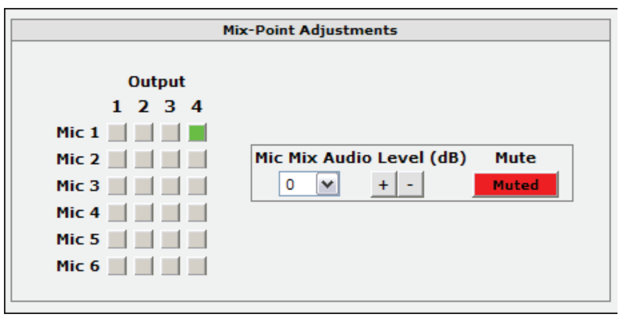


Figure 73. **Output Selection Drop Box**

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/output combination. The selected mix-point turns green. (Mic 1 to Output 4 is shown in figure 73).
2. Click the **Mic Mix Audio Level** drop box. A drop-down scroll 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 1-3 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 box. A drop-down scroll 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 1-3 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. 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.

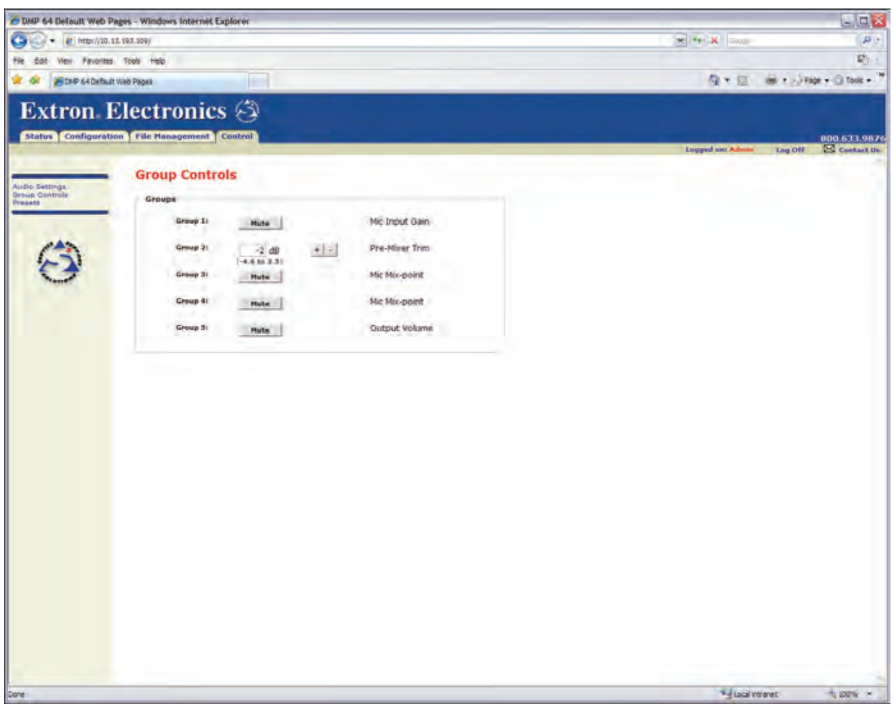


Figure 74. Group Controls Page

To adjust a group control:

1. 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.
2. Repeat step 1 for each group.

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

Recall a Preset

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

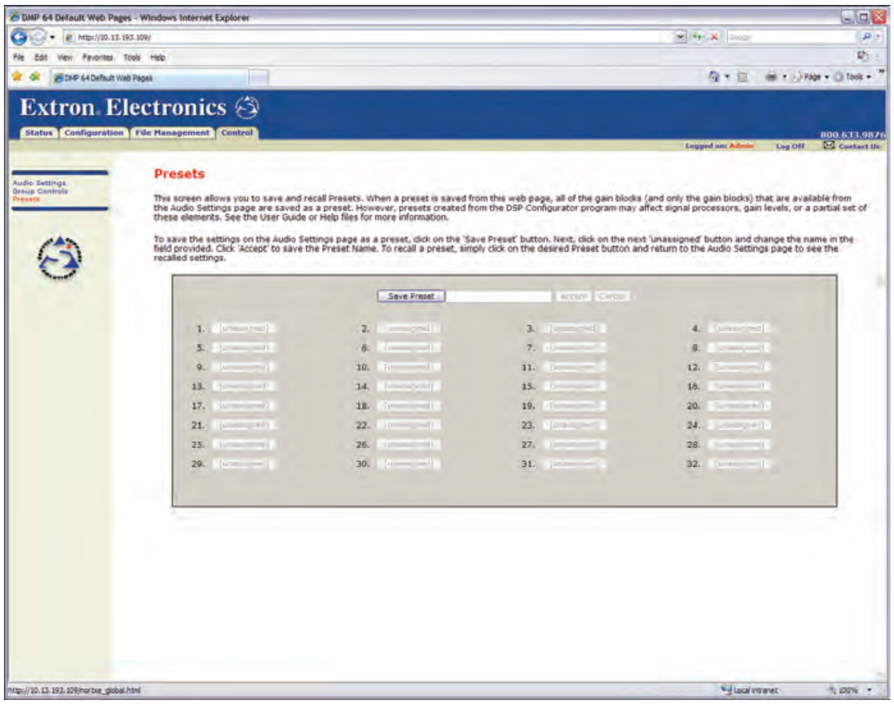


Figure 75. Presets

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

> **NOTE:** When presets are **recalled** via the HTML page, any signal processing adjustments that were 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 rest 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's name, passwords, or locally created file names.

The device rejects the following characters:
{space} + ~ , @ = ' [ ] { } < > ' " semicolon (;) colon (:) | \ and ?.

Reference Information

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

- **Specifications, Part Numbers, Accessories**
- **Firmware Loader**
- **Hardware Reset Modes**

Specifications

Audio

Gain	Unbalanced output: -6 dB; balanced output: 0 dB
Frequency response	20 Hz to 20 kHz, ±0.1 dB
THD + Noise	<0.01% @ 1 kHz, at maximum output level
S/NOTE:	>105 dB, 20 Hz to 20 kHz, at maximum output, unweighted
Crosstalk	<-90 dB @ 1 kHz, fully loaded
CMRR	>70 dB @ 1 kHz

Audio input

Number/signal type	6 mono, mic/line, balanced/unbalanced
Connector	(6) 3.5 mm captive screw connectors, 3 pole
Impedance	>10k ohms unbalanced/balanced
Nominal level	+4 dBu when level is set to 0 dB gain; adjustable from -60 dBu to +4 dBu
Maximum level	+24 dBu, balanced, when input gain is set to -3 dB
Noise level	<-120 dBV (1 µVrms) at 40 dB gain
Volume range	-18 dB to +80 dB, mic/line input, adjustable per input
Mic phantom power	+48 VDC, which can be switched on or off

NOTE: 0 dBu = 0.775 Vrms, 0 dBV = 1 Vrms, 0 dBV ≈ 2 dBu

Audio processing

D/A conversion	24 bit, 48 kHz sampling

Audio output

Number/signal type	4 mono, balanced/unbalanced
Connectors	(4) 3.5 mm captive screw connectors, 3 pole
Impedance	50 ohms unbalanced, 100 ohms balanced
Maximum level (Hi-Z)	>+21 dBu balanced, >+15 dBu unbalanced

Control/remote — audio processor

Serial host control port	1 bidirectional RS-232, 3.5 mm captive screw connector, 3 pole
Baud rate and protocol	38400 baud; 8 data bits, 1 stop bit, no parity
Serial control pin configuration	Pin 1 = TX, 2 = RX, 3 = GND
USB control ports	1 front panel female mini USB B
USB standards	USB 2.0, low speed
Ethernet host port	1 RJ-45 female
Ethernet data rate	10/100Base-T, half/full duplex with autodetect
Ethernet default settings	Link speed and duplex level = autodetected IP address = 192.168.254.254 Subnet mask = 255.255.0.0 Default gateway = 0.0.0.0 DHCP = off
Web server	Up to 200 simultaneous sessions 6.5 MB nonvolatile user memory
Program control	Extron control/configuration program for Windows® Extron Simple Instruction Set (SIS™) Microsoft® Internet Explorer®, Telnet

General

External power supply	100 VAC to 240 VAC, 50-60 Hz, to +12 VDC, 2 A, regulated
Power input requirements	+12 VDC, 1.5 A
Temperature/humidity	Storage: -40 to +158 °F (-40 to +70 °C) / 10% to 90%, noncondensing Operating: +32 to +122 °F (0 to +50 °C) / 10% to 90%, noncondensing
Cooling	Convection, no vents
Mounting	
Rack mount	Yes, with optional 1U rack shelf
Furniture mount	Yes, with optional under-desk mounting kit
Enclosure type	Metal
Enclosure dimensions	1.7" H x 8.75" W x 9.5" D (1U high, half rack wide) (4.3 cm H x 22.2 cm W x 24.1 cm D) (Depth excludes connectors.)
Product weight	1.9 lbs (0.9 kg)
Shipping weight	4 lbs (2 kg)
Vibration	ISTA 1A in carton (International Safe Transit Association)
Regulatory compliance	
Safety	CE, c-UL, UL
EMI/EMCAUTION:	CE, C-tick, FCC Class B, ICES, VCCI
Environmental	Complies with the appropriate requirements of RoHS, WEEE
MTBF	30,000 hours
Warranty	3 years parts and labor

NOTE: All nominal levels are at ±10%

NOTE: Specifications are subject to change without notice.

Part Numbers and Accessories

Included Parts

These items are included in each DMP 64 order:

Included parts	**Replacement part number**
DMP 64 Digital Matrix Processor	60-1054-01
Software Products DVD	
3.5 mm, 3-pole captive screw connectors w/strain relief (12)	10-703-11LF
3.5 mm, 4-pole captive screw connectors (2)	10-319-17LF
12 VDC, 2 amp external power supply	28-181-05LF
Power Cord 10 A/125 VAC, 7.5’	
Rubber Feet (4)	
Nylon tie wraps (12)	
Tweeker	
DMP 64 Set up Guide	

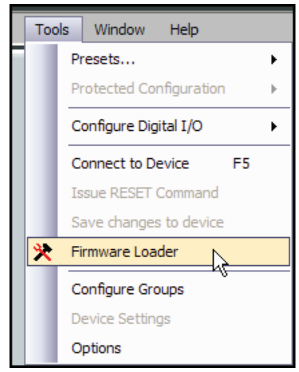
Accessories

These items can be ordered separately:

Adapters, power supplies, labels	**Part number**
USB A Male to USB Mini B Male Configuration Cable	26-654-06
PS 124, Multiple Output 12 Volt DC Power Supply	60-1022-01
CSR 6, Captive Screw to RCA Female Audio Adapter	26-575-01
MBU 125, Under-Desk Mount Kit	70-077-01
MBD 129, Through-Desk Mount Kit	70-077-02
RSB 129, Basic Rack Shelf Kit for 9.5” deep products	60-604-02
RSU 129, Universal Rack Shelf Kit for 9.5” deep products	60-190-01

Firmware Loader

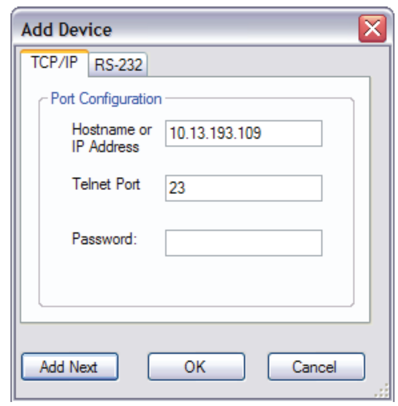
The DSP Configurator program includes a firmware loader program which allows replacing the firmware without taking the DMP 64 out of service. Download the desired firmware file from the Extron website, (see **Firmware Upgrade Page** for instructions.)



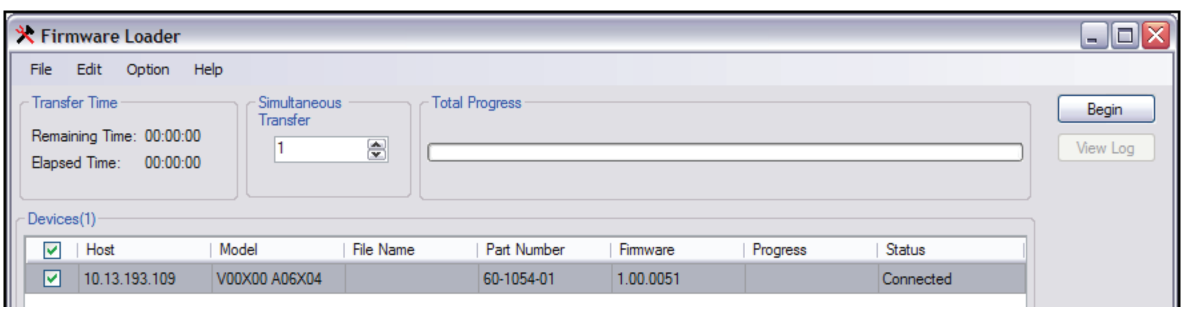
To access the firmware uploader:

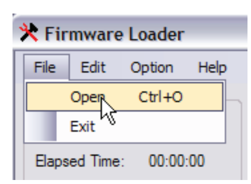
1. Select **Tools**, then **Firmware Loader**.
2. The **Add Device** dialog box appears. Type the IP address of the DMP 64, then press **OK**.

   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, exit the program by selecting **File | Exit**.

The firmware upload is complete.

DMP 64 Hardware Reset Modes

<table>
<tr><th colspan="5">DMP 64 Reset Mode Summary</th></tr>
<tr><th></th><th>Mode</th><th>Mode Activation</th><th>Result</th><th>Purpose/Notes</th></tr>
<tr><td>Use Factory Firmware</td><td>1</td><td>Hold the reset button while applying 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.</td><td>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 settins, etc.) 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.</td><td>This mode reverts to the factory default firmware version if incompatibilty issus arise with user-loaded firmware.
NOTE: User-defined Web pages may not work correctly if using an earlier firmware version.</td></tr>
<tr><td>Run/Stop Events</td><td>3</td><td>With 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: Nothings happens if the momenary press does not occur within 1 second.</td><td>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.</td><td>Useful for troubleshooting</td></tr>
<tr><td>Reset all IP Settings</td><td>4</td><td>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.</td><td>Mode 4:
• 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.</td><td>Enables resetting IP address information using ARP and MAC address.</td></tr>
<tr><td>Reset to Factory Defaults</td><td>5</td><td>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.</td><td>Mode 5 performs a complete reset to factory defaults, except for firmware:
• Does everything mode 4 reset does.
• All mix-points set 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.</td><td>Useful to start over with configuration or uploading, and to replace events.</td></tr>
</table>

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
1001 East Ball Road
Anaheim, CA 92805
U.S.A.

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

Europe, Africa, and the Middle East:
Extron Europe
Hanzeboulevard 10
3825 PH Amersfoort
The Netherlands

China:
Extron China
686 Ronghua Road
Songjiang District
Shanghai 201611
China

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

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 modification 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	**Europe**:	31.33.453.4040
Asia:	65.383.4400	**Japan**:	381.3.3511.7655

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 **USA - West**
Headquarters

+800.633.9876
Inside USA / Canada Only

+1.714.491.1500
+1.714.491.1517 FAX

Extron **USA - East**

+800.633.9876
Inside USA / Canada Only

+1.919.863.1794
+1.919.863.1797 FAX

Extron **Europe**

+800.3987.6673
Inside Europe Only

+31.33.453.4040
+31.33.453.4050 FAX

Extron **Asia**

+800.7339.8766
Inside Asia Only

+65.6383.4400
+65.6383.4664 FAX

Extron **Japan**

+81.3.3511.7655
+81.3.3511.7656 FAX

Extron **China**

+400.883.1568
Inside China Only

+86.21.3760.1568
+86.21.3760.1566 FAX

Extron **Middle East**

+971.4.2991800
+971.4.2991880 FAX