DMX

Networked AoIP Broadcast Console

Installation Guide and User Manual

AUDIOARTS ENGINEERING

Manufactured by Wheatstone Corporation

Publication Information

Trademarks

Customer Service Contact Information

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Manual Revisions

A = Initial release, October 2017

B = technical corrections, IP Navigator additions, operational clarifications, October 2017

C = Appendix A on networking multiple consoles added; some technical corrections; consolidated software app info into chapter 3, July 2018

D = Added info on a new feature: saving/loading EQ Configuration files (DMX Surface Setup, build 1.0.29 and later), September 2018

D.1 = Added info about saving two Studio EXT selections, October 2018

D.2 = Changed exclamation point graphic to display properly in a PDF, October 2018

E = Clarified note on using mono signals on the Razor along with other minor changes and clarifications. January 2019

F = Updated manual with Audioarts Engineering logo and new DMX Setup & support for WNIP Navigator, January 2020

G = Updated Gigabit Ethernet switch model and information on changing the Mix Engine’s IP address and Device number, March 2020

H = DMX-8 Surface now uses a separate in-line supply, various manual updates, December 2021

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Safety Instructions

1. **Read All Instructions.** Read all safety and operating instructions before operating the product.
2. **Retain All Instructions.** Retain all safety and operating instructions for future reference.
3. **Heed All Warnings.** You must adhere to all warnings on the product and those listed in the operating instructions.
4. **Follow All Instructions.** Follow all operating and product usage instructions.
5. **Heat.** This product must be situated away from any heat sources such as radiators, heat registers, stoves, or other products (including power amplifiers) that produce heat.
6. **Ventilation.** Slots and openings in the product are for ventilation. They ensure reliable operation of the product and keep it from overheating. Do not block or cover these openings during operation. Do not place this product into a rack unless proper ventilation is provided and the manufacturer's recommended installation procedures are followed.
7. **Water and Moisture.** Do not use this product near water such as a bathtub, wash bowl, kitchen sink, or laundry tub, in a wet basement, or near a swimming pool or the like.
8. **Attachments.** Do not use any attachments not recommended by the product manufacturer as they may cause hazards.
9. **Power Sources.** You must operate this product using the type of power source indicated on the marking label and in the installation instructions. If you are not sure of the type of power supplied to your facility, consult your local power company.
10. **Grounding and Polarization.** This product is equipped with a polarized AC plug with integral safety ground pin. Do not defeat the safety ground in any manner.
11. **Power Cord Protection.** Power supply cords must be routed so that they are not likely to be walked on nor pinched by items placed upon or against them. Pay particular attention to the cords at AC wall plugs and convenience receptacles, and at the point where the cord plugs into the product.
12. **Lightning.** For added protection for this product, unplug it from the AC wall outlet during a lightning storm or when it is left unattended and unused for long periods of time. This will prevent damage to the product due to lightning and power line surges.
13. **Overloading.** Do not overload AC wall outlets, extension cords, or integral convenience outlets as this can result in a fire or electric shock hazard.
14. **Object and Liquid Entry.** Never push objects of any kind into this product through openings as they may touch dangerous voltage points or short out parts, which could result in a fire or electric shock. Never spill liquid of any kind on the product.
15. **Accessories.** Do not place this product on an unstable cart, stand, tripod, bracket, or table. The product may fall, causing serious injury to a child or adult and serious damage to the product. Any mounting of the product must follow manufacturer's installation instructions.
16. **Product and Cart Combination.** Move this product with care. Quick stops, excessive force, and uneven surfaces may cause the product and the cart combination to overturn.
17. **Servicing.** Refer all servicing to qualified servicing personnel.
18. **Damage Requiring Service.** Unplug this product from the wall AC outlet and refer servicing to qualified service personnel under the following conditions:
    a. When the AC cord or plug is damaged.
    b. If liquid has been spilled or objects have fallen into the product.
    c. If the product has been exposed to rain or water.
    d. If the product does not operate normally (following operating instructions).
    e. If the product has been dropped or damaged in any way.
    f. When the product exhibits a distinct change in performance. This indicates a need for service.
19. **Replacement Parts.** When replacement parts are required, be sure the service technician has used replacement parts specified by the manufacturer or that have the same characteristics as the original parts. Unauthorized substitutions may result in fire, electric shock, or other hazards.
20. **Safety Check.** Upon completion of any repairs to this product, ask the service technician to perform safety checks to determine that the product is in proper operating condition.
21. **Cleaning.** Do not use liquid or aerosol cleaners. Use only a damp cloth for cleaning.

Hazard and Warning Label Identification

C A U T I O N

RISK OF ELECTRIC SHOCK
DO NOT OPEN

WARNING: SHOCK HAZARD - DO NOT OPEN

AVIS: RISQUE DE CHOC ELECTRIQUE - NE PAS OUVRIR

CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK DO NOT REMOVE ANY COVER OR PANEL. NO USER SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.

WARNING: TO REDUCE THE RISK OF FIRE OR ELECTRIC SHOCK, DO NOT EXPOSE THE PRODUCT TO RAIN OR MOISTURE.

The **Exclamation Point symbol**, within an equilateral triangle, alerts the user to important operating and maintenance (servicing) instructions in product literature and instruction manuals.

The **Lightning Flash With Arrowhead** symbol, within an equilateral triangle, alerts the user to the presence of uninsulated dangerous voltage within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock.

NOTE: The DMX Family of 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. These 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.

1 – INTRODUCING THE DMX CONSOLE

Thanks for joining the growing ranks of broadcasters employing Audioarts products designed, manufactured, and supported by Wheatstone. Throughout our long history we've endeavored to provide the finest quality products, documentation, and support. To obtain maximum benefit from the DMX console's capabilities please read, or at least skim over, this chapter and Chapters 2 and 3 prior to installing your DMX console. For those in a hurry a **DMX Quick Guide** and a **Mix Engine Hook-up Diagram** (PDF files on the USB flash drive that ships with the console) summarize the console's physical connections, using the software apps, and the Surface controls.

Your DMX console has these main components:

- **DMX-8 or DMX-16 Surface** – This part is commonly referred to as the "console" since it sits on the countertop and has all board operator controls. Figure 1-1 shows a DMX-8 Surface. Each DMX Surface has these features:
    - **Fader Channels** – There are eight identical fader channels on the DMX-8 and sixteen on the DMX-16. Each fader channel has eight push buttons with LED illumination (channel on, channel off, cue on/off, talkback (TB), and four PGM bus assigns), a color Organic Light Emitting Diode (OLED) display to show channel status, a 100 mm fader, and a rotary Channel Encoder to select the channel source. The Channel Encoders are also used to access three the Advanced Channel Features: audio mode, panning, and EQ & Dynamics—when not locked out from board operator access.
    - **Monitor Controls** – The console supports a Control Room and an associated Talk Studio with: CR monitor, CR headphone, Cue, and Studio Monitor level controls; CR and Studio monitor source selectors; timer controls; and four console Event buttons.
    - **Meter Bridge** – Three stereo LED meters display the levels for PGM 1, PGM 2, and a Switched signal: PGM 3, PGM 4, EXT, and CUE, when enabled. The Meter Bridge also has a four-digit timer, bus overload indicators, and an On-Air (Hot Mic) indicator.
    - **Headphone output** – A board operator headphone amp is included in the DMX Surface with a ¼" TRS jack on the right side of the chassis, in-line with the OLED displays.

Figure 1-1 DMX-8 Surface

Figure 1-2 Mix Engine, front panel

- **Mix Engine** – The 1RU Mix Engine (Figure 1-2) has the audio, logic, and network connections for the DMX console along with the DSP, signal mixing, and AoIP (Audio over IP) interface functions.

The Mix Engine has these main features:

- 5-port Gigabit Ethernet switch to network the Surface and Mix Engine, plus three other Ethernet AoIP devices, like the admin PC or a PC Blade running VoxPro, a Razor I/O Interface, or to connect a main AoIP gigabit switch to network multiple DMX studios together
- Two low-noise mic preamps, with female XLR inputs, each with switchable phantom power and input trim control
- Eight stereo/dual mono inputs (four analog and four AES)
- Four stereo program outputs, each with analog and AES output jacks
- Four stereo analog outputs for Control Room, Studio, Cue, and an operator headphone output for use with an outboard amplifier
- One 6-port GPIO logic connector

- **Installation Kit** – The DMX ships with a USB flash drive with PDF documentation files and the installers for Navigator and the DMX Surface Setup apps which are used to configure the DMX. Navigator is a licensed application so a separate license must be purchased to continue using that app beyond the 30-day demo license which is included with the DMX. Contact Wheatstone support (see page 59) to obtain your Navigator demo license.

  The DMX-8 and DMX-16 Surfaces use an in-line DC power supply so two IEC AC power cords are supplied for the Mix Engine and the Surface supply. All consoles also include a short CAT5 cable to connect the Mix Engine's Ethernet port to Port 5 on the built-in gigabit switch.

DMX OVERVIEW

The DMX console is a compact, self-contained, Audio over Internet Protocol (AoIP) radio broadcast and production console which uses the WheatNet-IP (WNIP) networking protocol. Designed for 24/7 operation, the console has two main hardware components: a 1RU rack mount **Mix Engine**, which has the console's audio and logic connections and a built-in 5-port gigabit Ethernet switch, and a tabletop board operator controller called the **DMX Surface** (which most users consider is the "DMX console"). Both Surface and Mix Engine are fully FCC and CE certified.

DMX consoles ship with pre-assigned IP addresses and a default configuration so they can be powered up and used right out of the box. These default settings are changed, to configure the console for use in an on-air studio, production room, newsroom, or other application, or to combine multiple DMX consoles into a larger network, using the two Windows applications included with the console. Using these software apps is covered in Chapter 3.

Razor I/O Interfaces (1RU, FCC and CE certified) are available to add audio and logic I/O to any DMX console. Razors use RJ45 jacks (StudioHub+ convention) for their eight inputs and eight outputs. Razors have three I/O styles: all analog I/O with two mic preamps (**Razor 16A**); all AES I/O (**Razor 16D**); or half analog and half-AES (**Razor 16AD**). Each Razor also has one RJ45 Logic jack with six GPIO ports. A Razor connects to a port on the Mix Engine's gigabit Ethernet switch using a single CAT5e or CAT6 cable of up to 330 feet (100 meters).

Your DMX can be networked with any **WheatNet-IP Blade System** to provide additional audio and logic I/O. This may require changes to the DMX settings (IP address and Device ID) or to the Mix Engine's Ethernet switch settings. See Appendix A about configuration settings to support networking multiple DMX consoles with an existing WNIP system.

DMX Surface

The Surface has eight or sixteen audio control channel strips or Fader Channels. Three of these fader channels are shown in Figure 1-3, along with the Monitor Controls which are in two columns of controls. Each fader channel has a large-knob rotary encoder so the board operator can select the audio source for each fader channel. A channel status display, just above the on and off buttons, shows the name of the audio source that's currently "dialed up" on that channel, along with other source and channel status information.

While the channel is off, the channel source may be using the channel encoder. When touched the channel display changes to show alternate source names set for that channel, with one name highlighted. Rotate the encoder until the desired source is highlighted then tap once, or "click" the rotary encoder to connect the highlighted source to that fader channel. The signal's name appears in green in the normal channel display.

Each fader channel can be assigned to any combination of Program buses using the four assignment buttons just below the channel encoder.

Each fader channel has a 100 mm fader for bus level control, plus eight illuminated switches (Off, On, Cue, Talkback (TB), PGM 1–PGM 4). The fader channel controls are identified in Figure 4-2 on page 46.

The two columns of controls at the right end of the Surface are the Monitor Controls, identified in Figure 4-4 on page 50. The left-hand column has the Control Room and headphone controls. The three top buttons select the source (PGM 3, PGM 4, or External) going to the *Switched Meter*.

A user-defined *Soft* button, which has no default settings, is just above the six Control Room monitor select buttons (PGM 1 – PGM 4, EXT 1, or EXT 2). These control the audio feeding the

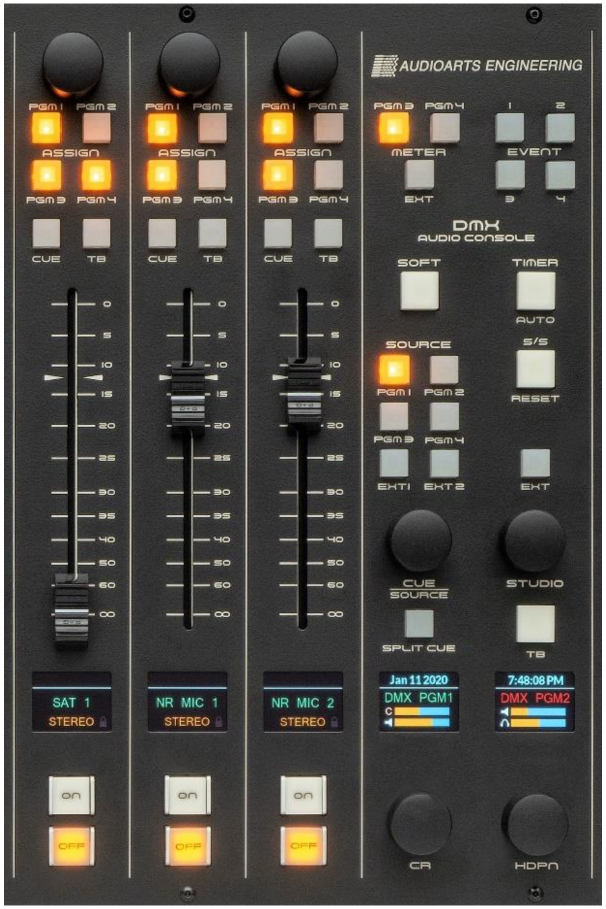


Figure 1-3 Surface Controls

monitor speakers in the Control Room and the board operator headphone jack. The Spilt Cue button sets how cue is fed to the headphones (whether cue even goes to the headphones is set in the DMX Surface Setup app).

The right column has the four *Event buttons*; two large *Timer control buttons*; a Studio Monitor Source Selector (*EXT*); and a large Talk to Studio button (*TB*).

The *CUE/SOURCE*, *CR*, *STUDIO*, and *HDPN* controls are rotary encoders used to control the level of the four dedicated monitor outputs on the Mix Engine. The two Monitor OLED displays show the current levels of those four outputs and their status (a red X over a bar graph icon indicates that output is muted). The *CUE/SOURCE* and *STUDIO* encoders are alternately used to assign sources to the various EXT buttons. They are also used to select and take "wild sources" for the CR and Studio monitor outputs, respectively.

All Surface controls are on a single field-replaceable *Control Panel* which connects to an internal host board using one (DMX-8) or two (DMX-16) plug-in ribbon cables. This allows for rapid field replacement, with minimal interruption to operations, in case of spills or other damage to the Surface controls. Because the program audio flows through the Mix Engine and not the Surface—except for the audio going to the side panel-mounted headphone jack, the Surface can be separately powered down from the Mix Engine without affecting the program audio.

Meter Bridge

An integrated Meter Bridge (Figure 1-4) sits above and behind the control panel. It has three stereo LED level meters, a four-digit Timer, and an On-Air indicator. The left meter and middle meters show the Program 1 and Program 2 bus levels, respectively. The right meter is switchable between showing the other program buses (PGM 3 or PGM 4) or an External signal (EXT) like an off-air tuner. The switched meter can also be configured to auto-switch to show the cue levels while cue is active.

The meters normally show both the average and peak levels, but software settings allow them to be changed to show only the average level or only the peak levels, for special functions.

Figure 1-4 DMX Meter Bridge

The four-digit Timer is controlled manually, using the Monitor section *S/S* button (Timer Start/Stop control), or automatically, when the *Auto Timer* button is lit and an audio source, set for Timer Reset (assigned to specific audio signals using the DMX Surface Setup app) is turned on.

Mix Engine

Audio and logic jacks are on the rear of the Mix Engine (Figure 1-5). This 1RU device has a **5-port Gigabit** switch; two high-quality low-noise **Mic Preamps** with gain control and 48-volt phantom powering; **Signal Processing** with EQ and dynamics applied to any fader channel; **eight audio inputs** (four analog and four AES); **four Program audio outputs** (four analog and four AES outputting the same set of signals); **four analog monitor outputs** to feed powered Control Room and Studio monitors, powered cue speakers, and an outboard board operator headphone amp; **six GPIO logic contacts**, each independently set to function as a logic input or logic output, and an **Ethernet** jack to connect the Mix Engine to the built-in 5-Port Gigabit switch.

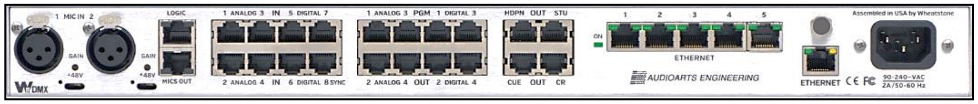


Figure 1-5 Mix Engine, rear panel

Input and Output Connectors

All audio, logic, and network connections on the Mix Engine and on the optional Razor I/O Interface (Figure 1-6)—other than microphone preamp inputs which have female XLR connectors, use RJ45 connectors and category wiring (CAT5, CAT5e, or CAT6). Figure 2-4 on page 12 has wiring pin outs for the RJ45 jacks that connect analog audio, digital audio, and logic to/from the DMX console. Analog and AES wiring conforms to the StudioHub+ convention with two balanced analog, or one AES/EBU (AES3) signal, per RJ45 jack.

The RJ45 Logic jack has six GPIO logic ports plus a +5V and a GND connection which use the WheatNet-IP logic jack wiring convention.

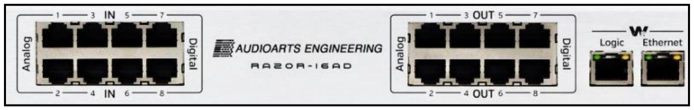


Figure 1-6 Razor 16AD Interface, rear panel RJ45 Connector Detail

DMX SPECIFICATIONS

Test Conditions:

- FSD = Full Scale Digital, 0 dBFS = +24 dBu analog
- 0 dBu corresponds to 0.775 volts RMS—regardless of the circuit impedance, as measured using a 600-ohm circuit.
- Noise specs measured using a 22 Hz – 20 kHz bandwidth. A 30k Hz bandwidth increases the noise measurement by 1.7 dB.

Mic Preamp

Source Impedance: 150 – 600 Ω, balanced
Nominal Input Level: -50 dBu
Input Range: -70 to -31 dBu (using trim control to reach nominal level)
Equivalent Input Noise: -131 dBu

Logic GPIO

Six per RJ45 connector: Connection assignments set using **Navigator** (type of logic command and whether it is a logic input or logic output)
Logic Input: Current-limited using internal pull-up. Supports +5 to +12 voltage logic. For a logic low, input voltage must fall below +2.5 volts.
Logic Output: Active low logic which supports +5 to +24 VDC logic, 50 mA nominal, 100 mA max

Analog I/O (Inputs & Outputs: +4 dBu, balanced)

Input Impedance: >10 k Ω, bridging
Optimal Source Impedance: <1 k Ω
Analog In > Analog Out Frequency Response:
+0.0, -0.25 dB, 20 Hz to 20 kHz @ +4 dBu
Analog In > Analog Out THD & Noise:
<0.0025%, 20 Hz-20 kHz @ +4 dBu
Nominal / Maximum Input Level: +4 dBu / +24 dBu (±18 dB of level gain/trim control available in Navigator)
Output Source Impedance: <10 Ω balanced
Output Load Impedance: 600 Ω optimal
Nominal / Maximum Output Level: +4 dBu / +24 dBu (±18 dB of output level gain/trim control in Navigator)
A > D Conversion: 24-bit resolution
D > A Conversion: 24-bit, advanced Delta-Sigma

Digital I/O (AES/EBU Inputs & Outputs)

Reference Level: -20 dB FSD (equivalent to analog +4 dBu)
Nominal / Maximum Input or Output Level: -20 dB FSD / 0 dB FSD (±18 dB of level gain/trim control available in Navigator)
Digital In > Digital Out Frequency Response:
±0.0 dB, 20 Hz to 20 kHz @ -20 dB FSD
Digital Input > Analog Output THD & Noise:
<0.00017%, 20 Hz-20 kHz @ -20 dB FSD

Digital I/O (continued)

Signal Format: AES-3, S/PDIF (inputs only)
AES-3 Input Compliance: 24-bit (uses SRC to support incoming sample rates of 32 – 96 kHz, 16- to 24-bit resolutions)
AES-3 Output Compliance: 24-bit
Output Sample Rate: 44.1 or 48 kHz
Processing Resolution: 24-bit

Surface Dimensions

DMX-08: 4.25" x 17.625" x 17" (H, W, D)
DMX-16: 4.25" x 29.625" x 17" (H, W, D)

Mix Engine Dimensions

A 1RU rack-mounted device:
1.75" x 19.0" x 13.25" (H, W, D)

Razor I/O Interface Dimensions

A 1RU rack-mounted device:
1.75" x 19.0" x 9.25" (H, W, D)

Power Supply

Type: Internal switching supply on the Razor and the Mix Engine; the DMX-8 and DMX-16 control surfaces use a separate in-line +16 VDC power supply
AC input: Detachable IEC cord
AC input: 90-240 VAC, 50/60 Hz
Output: +16 VDC @ 2.67 amps (Surface in-line supply)

Power Requirements

Mix Engine: <27 watts at 120 VAC / 60 Hz
DMX-08 Surface: <10 watts at 120 VAC / 60 Hz
DMX-16 Surface: <20 watts at 120 VAC / 60 Hz
Razor: <15 watts at 120 VAC / 60 Hz

Environment

Ambient Temperature: Less than 40°C
Cooling: Convection cooled, no fans

Wheatstone reserves the right to change the specifications without notice or obligation.

WARRANTY STATEMENT

LIMITED WARRANTY BY WHEATSTONE CORPORATION

1. All equipment sold and shipped to final destinations within the USA and its possessions warranted for one (1) full year from the date of purchase against defects in material and workmanship. All equipment sold and shipped to final destinations outside the U.S.A. and its possessions warranted for one (1) full year from the date of purchase against defects in material and workmanship.

All repairs to maintain the unit at original specification will be made at no charge to the original purchaser, except for shipping and insurance costs to be prepaid by the owner to the factory in the event the unit cannot be serviced by an authorized Wheatstone Corporation dealer.

2. This Warranty is subject to the following restrictions and conditions:

a) The owner must have filled out the enclosed Warranty Card and returned it to Wheatstone Corporation; or at the time of servicing the owner must provide proof of purchase from an authorized Wheatstone Corporation distributor or dealer.

b) This Warranty is valid for the original purchaser on the unit. Parts used for replacement are warranted for the remainder of the original warranty period. Repair or replacement is in the discretion of Wheatstone Corporation and is the exclusive remedy hereunder.

c) This Warranty DOES NOT apply to damage or defects resulting from abuse, careless use, misuse, improper installation, electrical spikes or surges, or alteration, repair, or service of the unit or equipment by anyone other than Wheatstone Corporation or its authorized dealer.

d) This Warranty is void if the serial number has been removed, altered or defaced.

e) This Warranty DOES NOT cover loss or damage, direct or indirect, arising out of the use or inability to use this unit or for shipping or transportation to any dealer.

f) Wheatstone Corporation reserves the right to modify or change any unit in whole or in part at any time prior to return delivery in order to incorporate electronic or mechanical improvements deemed appropriate by the Wheatstone Corporation but without incurring any responsibility for modifications or changes of any unit previously delivered or to supply any new equipment in accordance with any earlier specifications.

g) THERE ARE NO OTHER WARRANTIES, EXPRESSED, IMPLIED, OR STATUTORY, INCLUDING ANY WARRANTIES OF MERCHANTABILITY OR FITNESS FOR A PARTICULAR PURPOSE. IF FOR ANY REASON, ANY IMPLIED OR STATUTORY WARRANTY CANNOT BE DISCLAIMED, THEY ARE LIMITED TO THIRTY (30) DAYS FROM THE DATE OF PURCHASE. WHEATSTONE COPORATION IS NOT RESPONSIBLE FOR ELECTRICAL DAMAGE, LOSS OF USE, INCONVENIENCE, DAMAGE TO OTHER PROPERTY, OR ANY OTHER INCIDENTAL OR CONSEQUENTIAL, WHETHER DIRECT OR INDIRECT, AND WHETHER ARISING IN CONTRACT, TORT, OR OTHERWISE. NO REPRESENTATIVES, DEALERS, OR WHEATSTONE PERSONNEL ARE AUTHORIZED TO MAKE ANY WARRANTIES, REPRESENTATIONS, OR GUARANTIESS OTHER THAN THOSE EXPRESSLY STATED HEREIN.

Attention!

Federal Communications Commission (FCC) Compliance Notice:
Radio Frequency Notice

NOTE: 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. These 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.

This is a Class A product. In a domestic environment, this product may cause radio interference, in which case, the user may be required to take appropriate measures.

This equipment must be installed and wired properly in order to assure compliance with FCC regulations.

Caution! ***Any modifications not expressly approved in writing by Wheatstone could void the user's authority to operate this equipment.***

2 – DMX HARDWARE INSTALLATION

Each DMX Surface is 17” (43.2 cm) deep and 4 1/4" (11 cm) tall. The DMX-8 Surface is 17 5/8" (45 cm) wide. The DMX-16 Surface is 29 5/8” (75.3 cm) wide.

The Mix Engine (1RU) typically mounts within the studio cabinetry. DMX components are convection cooled allowing them to be mounted into any studio, including ones with active microphones, without adding any environmental noise. When locating the DMX Surface and Mix Engine, avoid proximity to devices which may generate electromagnetic fields, such as large power transformers, motors, and audio amps using switching supplies.

LOCATING THE DMX COMPONENTS

The DMX Surface is intended to set on top of the studio furniture countertop with its palm rest between six and twelve inches (15 to 30 cm) from the edge of the countertop (Figure 2-1). This “setback space” allows keyboards and mice, a VoxPro controller, copy, log sheets, etc. to be set in front of the console.

The **Mix Engine** (1RU) is typically rack mounted into a 19” rack located below the countertop. We recommend adding a 1RU or 2RU vented panel above and at least one 1RU panel below the Mix Engine for ventilation as it has a lot of op amps and thus generates a fair amount of heat. Mount the Mix Engine so its rear panel is easily accessible since that’s where the DMX audio, logic, and network connections are made.

The DMX Surface comes with an in-line 16 volt DC supply with a captive DC cable with a locking plug and a detachable IEC AC cord. An isolated-ground AC outlet must be located below the countertop to allow the six-foot DC cable to plug into the Surface and the six-foot AC cord to plug into an isolated-ground AC outlet.

The Surface’s Ethernet port connects to Port 4 on the Ethernet switch included within the Mix Engine. Use a straight-thru category cable (**customer supplied CAT5e or CAT6 cable**).

The DMX Surface is typically just set onto the countertop since its weight (16 lbs. for DMX-8, 26 lbs. for DMX-16) and four large rubber feet should hold it in place. If the Surface needs to be fastened to the countertop for security reasons, a step-by-step procedure is listed in the section *Fastening the Surface to the Countertop*, on page 11.

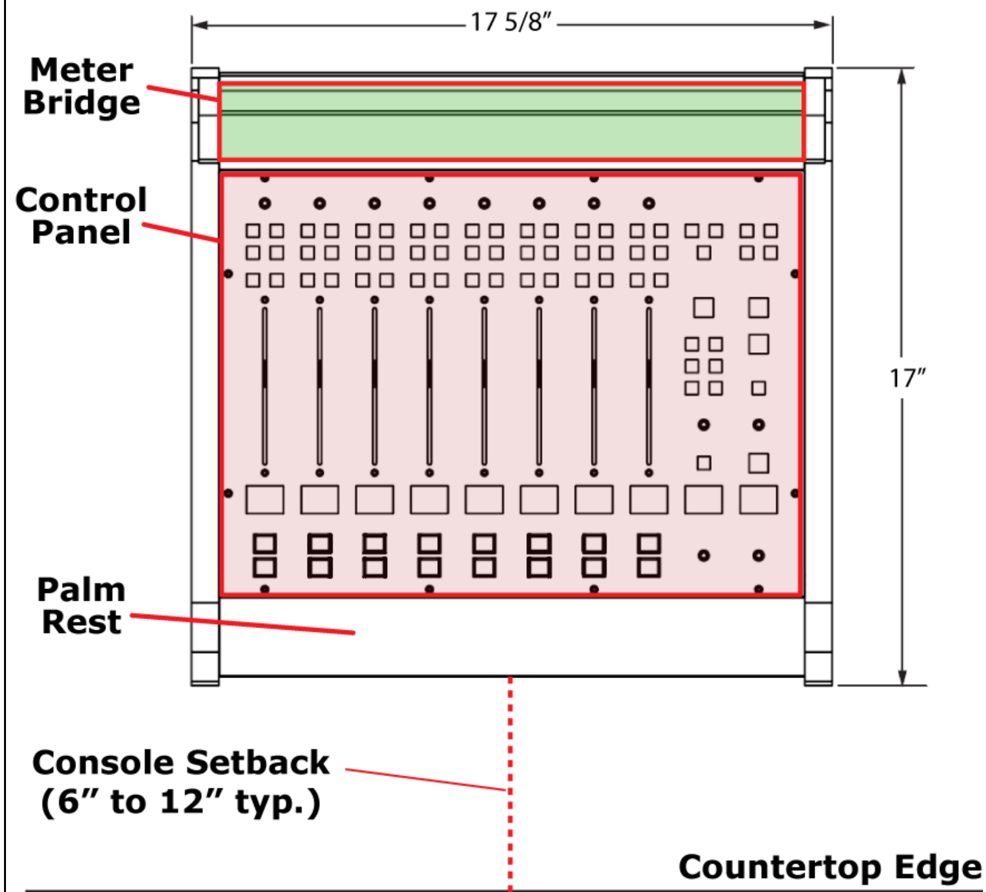


Figure 2-1 DMX-8 Surface, Countertop Positioning

For the cleanest installation, we recommend drilling a one- to two-inch cable access hole through the countertop for the Surface wiring. Only the customer supplied CAT5e/CAT6 cable, and the DC power cable/connector must pass through this hole, so hole size is not critical. The hole could be sized to match an available hole grommet, or it could be left raw since it’ll be covered by the Surface.

Here are the steps to drilling the cable access hole:

1. Position the Surface on the countertop so that the palm rest is parallel to the countertop edge and is setback the desired amount (typically, between six and twelve inches).

2. Use a pencil to mark the countertop at the center of the front edge of the palm rest (for the DMX-8, this is the center line between fader channels 5 and 6, as shown in Figure 2-1).
3. Move the Surface out of the way and mark a point 13” behind the palm rest mark, perpendicular to the countertop edge. This marks where the one- to two-inch cable access hole will be drilled.
4. Drill the cable access hole through the countertop.

Surface Placement & Cable Connections

Before setting the Surface back into position, remove the upper rear cosmetic cover by removing the black #1 Phillips screws. The top of the cover fits into a slot in the Meter Bridge top cover, so it may have to be pulled down slightly to remove it from that slot. When removed, the host control board, with the DC power jack and RJ45 (RJ45) Ethernet jack is revealed, as shown in Figure 2-2.

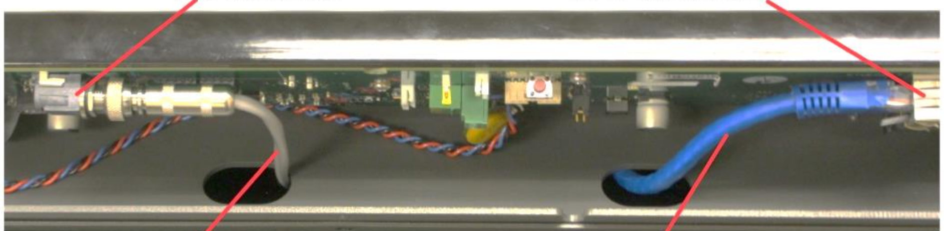


Figure 2-2 Surface rear view (upper rear cover removed)

Route the DC and category cables up through the countertop hole and through one of the lower chassis access holes, as shown in Figure 2-2. Plug and latch the category cable into the **RJ45 Ethernet jack**. Plug the DC cable into the **DC power jack** tightening its ring nut onto the threaded jack. Reattach the upper rear cover then set the Surface into position on the countertop.

Route the category cable to the back of the Mix Engine (Figure 2-3) and plug it into **Port 4** of the built-in **ETHERNET** switch. If you are not fastening the Surface to the countertop the in-line DC power supply’s IEC AC cord can be plugged into an isolated AC outlet.

Note: When the Surface is powered up without the Mix Engine also being powered, the peak meter LEDs slowly scan up each meter to indicate the Surface is not communicating with its host Mix Engine. Once the Mix Engine is powered, and connects to the Surface, this meter scanning stops.

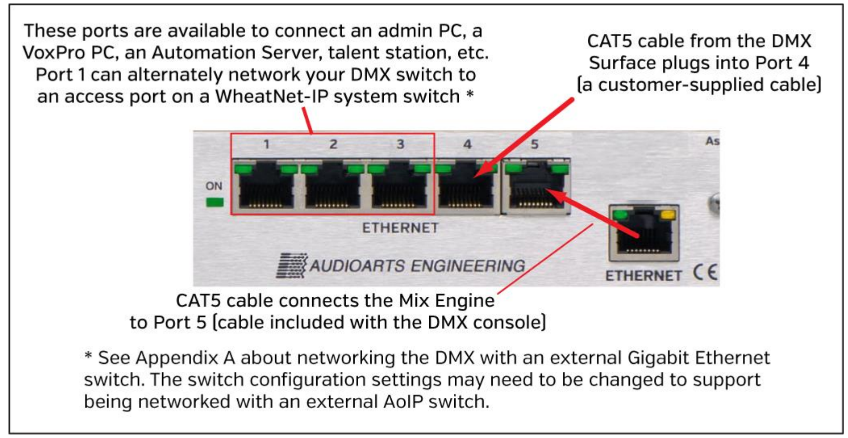


Figure 2-3 Mix Engine, Rear Panel Partial View, Ethernet and DC Power Connections

If not connected already, connect the short category cable supplied with the Mix Engine, from the **ETHERNET** jack to **Port 5** on the Ethernet switch, as illustrated in Figure 2-3

Fastening the Surface to the Countertop

When the DMX Surface needs to be fastened to the countertop for security, the screws fastening the two front feet must be removed. The Surface’s control panel (see Figure 2-1 on the previous page) must also be removed. Two #6 screws with oversize washers and lock washers, are required. They must go thru the countertop from below and go through the two front feet to screw into the chassis. When tightened, the two screws must not extend over ¼” into the chassis to avoid contacting the control panel circuit board or internal cabling.

The cable access hole should already have been drilled (per the four steps listed previously) and the two Surface cables should already been run through the hole and plugged into the Surface. The Surface must be powered down during this procedure:

1. Unscrew the two front feet. Retain the two #6 screws, lock washers, and flat washers for possible future reuse.
2. Use a 1/16” hex driver to remove the Control Panel screws. Lift the panel up slightly to unplug one ribbon cable on a DMX-8 or two ribbon cables on a DMX-16. Set the control panel aside onto padded material.
3. With the Surface set into position, mark the countertop through the front two feet and their chassis mounting holes using a pencil or a thin center punch, or even the 1/16” hex driver.
4. Move the Surface aside and drill a 3/16” hole at each mark completely through the countertop material.

5. *This step requires two persons.* One person inserts and holds the two long #6 screws (each with a lock and oversize washer) through the countertop from below, with the appropriate screwdriver for the two screws.

   A second person sets the two rubber feet onto the screws and then aligns the Surface over the screws, holding it in position as the two long #6 screws are tightened. Verify the Surface remains parallel to the countertop edge as the screws are tightened. Do not overly compress the two rubber feet. Verify the end of the screws do not extend more than ¼” above the chassis.

6. Set the back edge of the control panel onto the Surface chassis and plug in the ribbon cable(s). Set the control panel back down onto the chassis and reinstall the mounting screws removed in step 2.

MIX ENGINE CONNECTIONS

To facilitate wiring the Mix Engine, create a wire list, listing all connections to and from peripheral devices. Identify and create labels for each end of each audio, logic, and network cable. List these connections in a master facility wiring logbook to ease installation, future system wiring and equipment changes, and system troubleshooting.

Route all audio cabling to/from the Mix Engine with the maximum practical distance from all AC mains wiring.

Line level analog, AES/EBU, and logic wiring connect using RJ45 plugs (aka 8P8C plugs) on both the Mix Engine and Razor. Analog line-level inputs and outputs are balanced +4 dBu signals which only have connections to the plus (+) and minus (-) signals from peripheral devices.

Shielded twisted pair (STP) and unshielded twisted pair (UTP) wiring can be used interchangeably with balanced signals and with logic signals. When crimping CAT5 cables, UTP has the advantage since it’s a lot easier to hand crimp than STP wiring.

Each type of audio and logic signal using RJ45 jacks is identified in Figure 2-4. Note that the wire colors listed are for EIA/TIA T568B wiring (straight-thru patch cable wiring) but category cables may be wired using the T568A standard, which swaps the Orange and Green wire pairs with WHT/GRN on terminal 1, GRN on terminal 2, WHT/ORG on terminal 3, and ORG on terminal 6. The other wire connections are identical.

To simplify your audio wiring, use standard lengths of straight-thru CAT5 cables along with RJ45 Audio Adapters to plug into peripheral device jacks. Angry Audio and other vendors make a wide variety of RJ45 Audio Adapters, most of which are carried by your Audioarts and Wheatstone dealers. For more information on RJ45 cable adapters for audio wiring refer to this web page: [Angry Audio]

AUDIO WIRING

RJ45 PIN (WIRES*)	SIGNAL
1 (WHT/ORG)	LEFT + / AES +
2 (ORG)	LEFT - / AES -
3 (WHT/GRN)	RIGHT +
6 (GRN)	RIGHT -
4, 5, 7, 8	UNUSED

LOGIC WIRING

RJ45 PIN (WIRES*)	SIGNAL
1 (WHT/ORG)	GROUND
2 (ORG)	LIO 1
3 (WHT/GRN)	LIO 2
4 (BLU)	LIO 3
5 (WHT/BLU)	LIO 4
6 (GRN)	LIO 5
7 (WHT/BRN)	LIO 6
8 (BRN)	+5 VDC

* EIA/TIA T568B WIRE COLORS

Figure 2-4 Audio & Logic RJ45 Pin Outs & Signals

To simplify logic wiring, NotaBotYet has GPIO Logic Adapters for the RJ45 Logic jacks on the DMX Mix Engine, Razor, and WNIP Blades. A straight-thru CAT5 cable connects the RJ45 Logic jack to the GPIO Logic Adapter which breaks-out the six GPIO connections to screw terminals for wiring hard-wired peripherals like warning lights and hot mic LEDs.

When wiring directly from audio RJ45 jacks to individual XLR, TRS, or other audio connectors, single-pair UTP (DataMax 5100 or similar) can be used to wire AES or mono analog connections. Dual-pair UTP (DataMax 5200 or similar) can be used for wiring to stereo connections when a single connector (like a D-Sub) is used on the peripheral device.

When wiring a stereo analog device with two connectors, use two single-pair UTP wires connected to a common RJ45 for the cleanest connection to the two audio plugs. Alternately, separate out the ORG and GRN wire pairs from a standard CAT5 cable to wire to the two audio plugs. For a clean installation, cover each wire pair with tubing held in place with a short piece of shrink tubing over the ends of the tubing and the CAT5 jacket.

The two female XLR jacks on the Mix Engine are designed for microphone cables wired using shielded wiring specifically for low-level balanced mic signals. The monitor outputs, when going to a powered monitor speaker may also require using shielded cable that terminates the shield at the monitor speaker ends since there’s no shield connection on the Mix Engine’s RJ45 jacks.

Note: The StudioHub+ wiring convention uses pin 4 on the RJ45 plug as a DC Ground to support PoE (Power-over-Ethernet). The Mix Engine, Razor, and WNIP Blade RJ45 audio connectors **do not tie pin 4 to ground** since these connections do not support PoE. Thus, a shield or ground connection can only be made at the peripheral device end. If the peripheral device only has unbalanced inputs using a Tip-Sleeve or RCA connector, a signal matchbox, or an audio balun transformer (balanced in, unbalanced out) must be placed in-line to properly unbalance the balanced output signals.

An unbalanced output device can directly connect to a **DMX analog input** by connecting its low (-) signal wires (pin 2, ORG, and/or pin 6, GRN) to GND on the unbalanced device. Connect the hot wire (the unbalanced signal) to the WHT/ORG and/or WHT/GRN wires (terminals 1 and 3). Since unbalanced devices have lower output levels—as compared to a balanced signal, use **Navigator's Device > Sources** tab to raise the input. Any DMX input can be increased by up to +18 dB from its nominal input level (0 dB of gain or trim).

When a **DMX analog output** needs to connect to an unbalanced device a signal matchbox or an audio balun transformer (a balanced in-to-unbalanced out device) must be used.

Note: Do not connect the low (-) signals (the ORG and GRN wires) on any output going to an unbalanced input device to ground. This can not only lead to phase and crosstalk issues it can also lead to component failure in the Mix Engine or Razor over time since that action shorts an active op amp output to ground.

Ethernet Connections

Ethernet network wiring also uses RJ45 jacks but, because a WNIP network is running at 1000Base-T (gigabit or 1 GB) only CAT5e or better cabling should be used. For a short connection, like connecting the Mix Engine to Port 5 or the Surface to Port 4 on the **Ethernet switch** (Figure 2-3 on page 11), CAT5 cabling can be used but all longer cable runs should only use CAT5e or CAT6 cables.

Ports 1, 2, and **3** on the five-port Gigabit Ethernet switch are available to connect a Windows PC running the **DMX Surface Setup** and **Navigator**. This PC won't need to be connected during normal use unless **Navigator** is needed to change signal connections. Alternately, if a VoxPro or other audio editor PC is networked with the DMX, the two setup apps could be installed on that PC. Any open switch ports are available to network a WNIP-compatible media server, a Razor I/O Interface, an accessory control panel, or any WNIP Blade.

Port 1 is preset so that the DMX switch can connect to a WNIP system AoIP switch to network the DMX with additional WheatNet-IP consoles, Blades, Razor I/O Interfaces, and WNIP-compatible media severs in a larger WNIP network. Appendix A has details on checking the built-in Ethernet switch's configuration settings to ensure it is setup to support connecting the DMX to a WNIP network.

DMX POWER UP

The Mix Engine, Razor I/O Interfaces, and Blades do not have power switches since all are designed for continuous 24/7 operation. All are designed to work with AC mains supplying 90-240 VAC at 50/60 Hz. For the most reliable operation, all DMX components should only plug into isolated ground circuits (orange outlets in the USA) which are ideally on a UPS (Uninterruptible Power Supply).

On the DMX Surface, plug in the Surface power supply using the included IEC AC cord. Plug the IEC AC cord into the Mix Engine. Plug in the other end into an isolated-ground AC outlet. It takes about 90 seconds for the DMX to boot up and be ready for use.

When power is first applied to the Mix Engine, the rear panel green On LED (next to the Ethernet switch) lights up first. This is followed by the green LEDs blinking on active Ethernet switch ports. Note that the Surface Ethernet jack only blinks one LED since it communicates at 100Base-T. The ports with the Mix Engine and a Razor will have both green LEDs blinking since they both communicate at 1000Base-T. A networked PC will also only blink one LED if its NIC doesn't support gigabit speeds.

The front panel LINK LED (Figure 2-5) lights up solid when the Mix Engine's Ethernet jack is connected to port 5 of the Ethernet switch. On a stand-alone DMX, the other two front panel green LEDs: ROUTE MSTR (Route Master) and CLOCK MSTR (Clock Master) also light up solid—after about 90 seconds, indicating the Mix Engine is ready for use.

Figure 2-5 Mix Engine Front Panel Status LEDs

When the DMX is networked with existing WNIP consoles and/or Blades, the DMX Mix Engine will most likely not be designated as either master, thus its ROUTE MSTR and CLOCK MSTR LEDs may both be off. This simply indicates there are other Blades in the WNIP network which are set as the Route Master and Clock Master. With multiple Mix Engines and/or I/O Blades in a WNIP system, there will be one Blade designated as the Route Master with another Blade designated as the Clock Master.

If the red ERROR LED should ever light, it indicates an error condition has occurred within the Mix Engine. To reset this error, first try rebooting the Mix Engine using Navigator. If the red LED remains lit after the Mix Engine reboots, power cycle the Mix Engine (unplug the AC cord from the Mix Engine, wait five seconds then plug the AC cord back in).

If the red Error LED lights up again after the Mix Engine restarts it indicates a serious fault which requires service. Contact Wheatstone technical support for assistance. See Chapter 5 (page 59) for information on obtaining service and support for your DMX console.

USING COMMON DMX FEATURES

Once the Mix Engine and Surface are powered up and connected to Ports 4 and 5 on the built-in Ethernet switch, on a stand-alone DMX the Surface and Mix Engine should appear as shown in Figures 1-1 and 1-2 on page 5.

Note: Figure 4-2, on page 46, identifies each fader channel control and Figure 4-4, on page 50, identifies each monitor control.

On Fader Channel 3, with the channel turned off, rotate the Channel Encoder. The channel's normal display switches to show a list of source names (Figure 2-6). The middle name is highlighted with the name of the device that it's on listed above the white line. Blade01 is the default name assigned at the factory to each DMX Mix Engine.

Figure 2-6 Selecting a Channel Source

Rotate the encoder clockwise to move the highlighting down the alphanumeric list of names. Rotate the encoder counterclockwise (CCW) to move the highlighting up through the names. The names wrap to present a continuous list. *NoSource*, which appears at the top of the list, is a good way to identify when the names have wrapped around.

Every physical input on the Mix Engine has a signal name of up to eight characters. The default names all start with BL, for Blade, followed by that Blade's ID number (1 is the ID number assigned to a new Mix Engine). The default names end with a signal number, like S02, to identify the physical "Source" or jack on the Mix Engine, I/O Blade, or Razor. Likewise, the Mix Engine outputs are identified using unique names to identify each Destination: CB01 CR (Mix Engine 1, Control Room) CB01PGMA (the analog and AES outputs for PGM 1) and so on.

All DMX inputs are set as stereo—except for **Analog Input 1**, which is set as dual mono and is assigned the names BL01MIC1 and BL01MIC2. The three remaining **Analog inputs** are BL001S02, BL001S03, and BL001S04. The four **AES inputs** have the default names of BL001S05 thru BL001S08. These match the jack numbers (1 – 8) on the Mix Engine's rear panel.

Analog input 1 is meant to connect to **MICS OUT** so that the two mic preamps can be used without making any configuration changes. If the built-in mic preamps are not used, Analog Input 1 can be changed to a stereo input using the Navigator application.

As the encoder is turned CW past the Mix Engine's physical input signal names, the internal signals within the DMX are listed. Many, if not most, of these signals are typically not set as "visible" on fader channels, but on a new console every source signal is set visible to ease installing and configuring the DMX. Assigning which signals are visible is done using the DMX Surface Setup application.

Rotate the channel 3 encoder so that **BL001S02** is highlighted, as shown in Figure 2-6. Press once on the channel encoder knob to select that signal. This is called "clicking" the encoder–just like clicking a PC mouse button. That connects the highlighted signal to the channel. **BL001S02** is then shown, in green, in the middle of the channel 3 display (Figure 2-7).

The OLED displays are divided by a white line in the normal display. The space above the line indicates when cue, equalization, and dynamics are active on that channel (Figure 2-7). Icons for the High Pass and/or Low Pass Filters are also shown when they're active. EQ (for Equalization) and DYN (for Dynamics) are shown when they're active. From the factory, mic processing is active on channels 1 and 2, and are turned on, hence the **EQ** and **DYN** indications on those two fader channels.

Figure 2-7 Displays for Channels 2, 3, and 4

Below the source name, in orange, is the channel's mode setting. The default setting for every channel is STEREO (mono signals like mics are fed to both left and right channels by default).

To either side of the mode name are icons to indicate source status. On the right side is a lock icon to indicate that source's LIO status (logic control status). On the left side—if the channel source is identified as a mic, there's a mic icon which turns red while that channel is on.

Advanced Channel Features

Each channel has three Advanced Channel Features: **Mode**, **Pan**, and **EQ & Dynamics,** which are accessed by "double-clicking" or quickly pressing twice on the channel encoder knob. These advanced features are all unlocked when the DMX ships from the factory.

"Double-click" the channel 3 encoder (tap it twice quickly). **Mode** now appears above the white line. Rotating the encoder, while Mode is shown in the display, steps through the four audio modes (Right, Left, Mono, and Stereo) which can be set on a channel. Make sure STEREO is shown, then double-click the encoder.

Pan is then shown above the white line. Now, rotating the encoder, pans or balances the signal toward the left or right channel. Readjust the encoder so CENTER is shown then double-click the encoder again.

The channel 1 - 8 OLED displays switch to show **EQ & Dynamics** control screens. The initial display screens show whether EQ & Dynamics are turned on or off on that channel. Additional screens show the EQ & Dynamics settings. On fader channels 1 and 2, EQ and Dynamics are assigned by default for "light mic processing" since these channels are typically assigned to a Host and a Guest mic. The remaining fader channels do not have any EQ or Dynamics settings applied. For now, double-click the channel encoder again. The eight channels will switch back to their normal displays showing their current source names.

Note: The Mode, Pan, and EQ & Dynamics controls should be selectively locked out from board operator access prior to releasing the console for daily use. This is done using the *Talent Access page tab* in the DMX Surface Setup app (see page 23).

Connecting Audio

To understand the audio connections and DMX Surface controls, connect a stereo analog audio signal, like a +4 dBu test tone from a tone generator or test CD to the **ANALOG 2 IN** jack on the Mix Engine (Figure 2-9 on the next page shows the Mix Engine rear panel jacks). This audio input is named **BL001S02** and is the Channel 3 source that was set in the previous section.

Assign channel 3 to **PGM 1**, **PGM 2**, **PGM 3**, and **PGM 4** so all four assignment buttons are lit. Set the channel fader so the middle line on the fader knob aligns with the two arrows at the -12 dB position. This sets that channel for unity gain. Turn Channel 3 on by pressing and releasing the **On** button directly below that channel's display. The button turns red indicating that **BL001S02** is now feeding the assigned Program buses.

If the input signal is a +4 dBu test tone, then all green meter segments will be lit along with the first yellow segment, which indicates -20 dBFS (decibels Below Full Scale). This equals 0 VU on an analog VU meter. The **Program 1** and **Program 2 meters** should both show this level. The **Switched Meter** should also show this level when the **Meter PGM 3** or **Meter PGM 4** button is lit in the monitor section.

Connect an analog test set to each of the four PGM ANALOG outputs to confirm each has +4 dBu out. Connect a digital test set to each of the four PGM DIGITAL outputs, which have the same signals as the analog jacks, to confirm that each has a -20 dBFS output. The default sources connected to these four PGM outputs are the PGM 1, PGM 2, PGM 3, and PGM 4 busses.

To check the monitor outputs (the four OUT jacks), connect a test set or a powered monitor speaker with a balanced input to the **CR Out** jack. In the left column of the monitor section, select PGM 1 as the monitor source for the Control Room. DMX PGM1 will be shown in the left monitor display (Figure 2-8). Rotate the **CR encoder** to adjust the volume of CR Out. Its relative output level is indicated by the bottom bar graph, with the speaker icon, in the left monitor OLED display.

Figure 2-8 Monitor Displays

Move the test output to the **HDPN Out** jack. Its output level, and the DMX Surface's headphone jack level, is adjusted using the **HDPN encoder**. Its level is indicated by the bottom bar graph in the right display with the headphone icon. It has the same monitor source as **CR Out**.

Move the test output to the **STU Out** jack. Its current monitor source is shown in the right-hand display. If *NoSource* appears in the right-hand display, an on-the-fly monitor source can be set for the Studio by press/holding, for about three seconds, the **STUDIO encoder**. A list of source names appears in the right-hand display. Rotate the Studio encoder to highlight DMX PGM2 then click the Studio encoder to assign that source as the Studio monitor source, as shown in Figure 2-8.

With a source selected, use the Studio encoder to adjust the STU Out volume. The studio monitor level is indicated by the upper bar graph in the right display with the speaker icon.

Move the test output to the **CUE Out** jack. Press **Cue** on channel 3 (it lights, and **CUE** is shown in the channel display). Use the **CUE/SOURCE encoder** to adjust the cue volume. The upper bar graph in the left display, with a C, indicates the cue output level. Note that even though this is a "stereo" or two-channel output, the DMX cue output is mono.

Note: If you followed all the steps in the *Using Common DMX Features* section you should have a good understanding of how to connect audio into and out of the Mix Engine and how to use the DMX Surface controls. The rest of this chapter covers specific details on the audio and logic jacks found on the Mix Engine and the Razor Interface.

Chapter 3 (starting on page 20) covers how to install and use the supplied apps (DMX Surface Setup and Navigator) to name the DMX signals, add logic control, and configure the DMX console for your specific application.

MIX ENGINE SIGNAL NOTES

This section covers each Mix Engine connection and how the DMX apps are used to configure that signal or connection. Figure 2-9 shows the audio and logic jacks on the back of the Mix Engine.

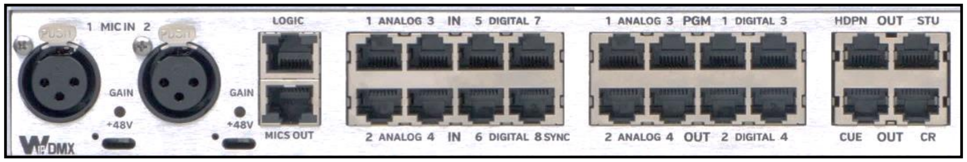


Figure 2-9 Mix Engine, rear panel, Audio & Logic Jacks

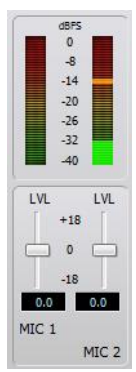
Mic Preamps – The two female XLR jacks connect to two on-board low-noise mic preamps. Any type of dynamic or condenser mic can be plugged in to boost two mics up to line level. The line level outputs for the two mic preamps are on the MICS OUT jack. It's typically connected to ANALOG IN 1 using a short CAT5 cable.

The two mics are typically assigned to fader channels 1 and 2 which is why the factory-set EQ and Dynamics Processing is applied to those two faders. If additional mic preamps are needed, the **Razor 16A Interface** has two mic preamps and the **M4IP-USB** Blade has four Super Quiet mic preamps, along with Vorsis Embedded mic processing. It also has four USB ports and local analog and AES outputs for headphone and/or studio monitors and recording outputs.

Each DMX preamp has a gain control and a recessed phantom power switch on the back panel. For dynamic mics set the phantom power switch to **off** (+48V LED is off). To power a condenser mic set the phantom switch to **on** (+48V LED is lit yellow).

Note: Leave the phantom power switch set to off until the condenser mic and its mic cable are plugged into the XLR jack, then turn on the phantom power. When a condenser mic must be unplugged, turn off the phantom power prior to unplugging the mic, or else unplug the mic cable at the Mix Engine and then unplug the microphone.

MICS OUT – This jack has the two line level outputs from the mic preamps: Mic Preamp 1 is the left signal and Mic Preamp 2 is the right signal. Connect a short straight-thru CAT5 cable from **MICS OUT** to **ANALOG 1 IN** since that input comes preset for two mono mic signals rather than for one stereo signal like all other Mix Engine inputs.



The rear panel Gain controls are adjusted, once the mics are wired and the Mics Out jack is connected to the Analog 1 In jack, by observing the Navigator app's Blade > Sources tab, shown at right, which has a Level Meter for each input on the Mix Engine. With the talent speaking at their normal "radio voice level" adjust the mic Gain pot so the top of the green bar graph signal is around -20 dBFS and the peak LED is bouncing around -8 dBFS with the Navigator gain set to 0.0 dB.

LOGIC - The Mix Engine and Razor Interface each have one RJ45 Logic jack for connecting logic to non-IP external devices like a warning light interface, mic arm LEDs, and/or a CD player. Each Logic jack has six LIO logic ports (on pins 2–7) plus ground (on pin 1) and +5 volts (on pin 8). Each logic port is independently assigned, using Navigator, to function as a logic output like an On Tally, Studio In Use tally, Start Pulse, etc., or as a logic input like a Channel On, Channel Off, Ready, Cue, etc.

Each Mix Engine also has 128 Soft LIO (SLIO) logic signals which can be used by WNIP-compatible devices, like talent stations and media servers, for bi-directional command and control of the DMX console over Ethernet. SLIO signals are also assigned using Navigator. Each can be set as either an input or an output. See *Assigning Logic* (starting on page 42) for details on assigning both LIO and SLIO logic.

ANALOG IN - The **ANALOG IN 1 - 4** inputs are designed for balanced +4 dBu signals. Unbalanced -10 dBv signals can also be directly connected without needing a match box since each input has input gain and balance controls in Navigator (see *Level & Balance Controls* on page 41).

The analog inputs have the default names: BL01MIC1 (input 1 left), BL01MIC2 (input 1 right), BL001S02 (input 2), BL001S03 (input 3), and BL001S04 (input 4) in the DMX source signal list. Any input can be separated into two mono inputs (see *Changing the Signal Type* on page 40) to allow two mono signals to connect to one RJ45 jack.

DIGITAL IN – The DIGITAL IN 5 – 8 inputs are designed for stereo differential AES/EBU (AES3) signals which, in most cases, can alternately have an unbalanced S/PDIF digital signal connected. Their default names are BL001S05 – BL001S08 in the DMX source signal list. They have the same gain and balance controls in Navigator as analog signals and any AES/EBU input can also be split into two mono signals to connect codecs and phone systems. Digital In 8 can alternately be used to connect a Sync Reference to synchronize the DMX to an external sample clock.

PGM OUTPUTS – There are eight PGM Outputs (four stereo balanced analog and four AES/EBU) which, by default, are cross connected to the PGM 1 – PGM 4 busses. The four destinations are named CB01PGMA – CB01PGMD in DMX source lists but any source can be connected to these destinations. Note that they cannot be split for dual mono out operation and the same signal is applied to both analog and AES/EBU output.

Note: If you change the source connected to any PGM or Monitor output, uncheck the DMX Surface Setup option *Use Default Signal Mapping* (see *Input/Output* on page 23) since that option, when checked, reconnects the four PGM buses to the four PGM outputs, and the default Monitor sources to the four Monitor outputs, whenever the Mix Engine or DMX Surface is rebooted or powered.

Note: The four AES/EBU outputs can only connect to balanced AES/EBU inputs. Connecting an AES/EBU output to an unbalanced S/PDIF input will not work. A signal translation device is required to connect an AES/EBU output to an S/PDIF input.

MONITOR OUTPUTS – The four stereo analog output jacks: **CUE**, **CR**, **STU**, and **HDPN** have the destination names of CB01CUE, CB01CR, CB01STU, and CB01HDPN. Normally, the sources DMX CUE, DMX CR, DMX Stu, and DMX Hdpn are connected to these four outputs allowing the four monitor level controls (CUE, CR, STUDIO, and HDPN) to control the monitor output levels. Alternate sources can be connected to any of these destinations. When an alternate source is connected to any of these outputs, the associated Surface level control no longer affects that output. See the **Note** about unchecking *Use Default Signal Mapping* on the last page when connecting alternate sources to the Monitor outputs.

CR OUT is typically connected to a pair of powered monitor speakers, while **CUE OUT** is typically connected to one or two powered cue speakers, so they are not often connected to an alternate source. If the console doesn't have an associated talk studio, the **STU OUT** jack could instead connect to a hybrid and a call recorder, an air skimmer, an internet streamer, or other device. Likewise, the **HDPN OUT** jack, if it doesn't need to connect to an outboard headphone amp, could connect to an alternate source for another stereo analog line output from the Mix Engine.

Note: The built-in ¼" headphone jack, on the right side of Surface has its own destination (**DMX Hdpn**) so it's not affected by changing the source connected to destination CB01HDPN.

ADDITIONAL DMX FEATURES

This section covers various special DMX operating features like using four-wire devices (callers and remotes) with the DMX, how to setup the PGM 4 bus assign buttons to function as Off Line bus assign buttons, how to set the source for the various EXT buttons in the monitor control section of the Surface, how to set up monitor muting, how to setup Events, and how to adjust the gain and balance control on Inputs and Outputs.

Callers & Remotes

A caller or a live remote typically uses a codec or other Telco-type of device to connect to the console. These devices have a *From Network* signal (the audio coming from that caller or remote) and a *To Network* signal (often called the return or mix-minus audio) which is sent back to the caller or remote so they can hear the board operator or hear the on-air signal—but always minus their own *From Network* audio. On the DMX this *To Network* audio signal is typically their channel's bus-minus audio which is connected to the input of the codec or other Telco device.

When the DMX will have phone callers and/or live remotes connected, the PGM 4 bus assign buttons are typically reconfigured to be Off Line bus assignment buttons. The Off Line bus is an internal bus used solely to create the *To Network* bus-minus signals for callers and remotes while their fader channels are off. The Off Line buttons are also typically setup for pre-switch and pre-fader operation to allow the talent to carry on a hands-free conversation with a caller/remote while they're not on-air (the caller/ remote channel is turned off).

When the caller/remote channel is turned on, their bus-minus signal—or *To Network* audio, switches to the air signal (typically PGM 1) so that the caller/remote then hears everything going to air—but always minus their own audio. This allows multiple callers and remotes to all go live-to-air without the board operator having to change any bus assignments or other settings. And, if the board operator needs to talk to a caller/remote while they're live, pressing the **TB** button on the caller/remote channel overrides the PGM 1 bus-minus audio so the board operator can give a cue to a specific caller or remote.

When a caller/remote is on a dedicated fader channel, setting up the *To Network* signal is simple: connect that fader channel's bus-minus signal (source names: DMXBM01–DMXBM16) to the hybrid or codec destination in Navigator's Crosspoint Grid. When the callers and remotes are set as visible on multiple fader channels things get a bit more complicated since the bus-minus signal connected to the hybrid or codec must change to follow the channel that caller or remote is on. Fortunately, the Navigator *Associated Connections* feature allows one to set up "trigger conditions" like taking a codec on channel 15. An associated connection is then assigned (like connecting channel 15 bus-minus to the codec) when the WNIP system detects the trigger condition. Thus, when the codec is dialed up on fader channel 15, the system automatically connects the fader 15 bus-minus signal to that codec's To Network destination.

Typically, an Associated Connection is created for each fader that each codec is set as "visible on" so that no matter which channel the codec gets taken on the correct bus-minus signal gets connected back to the codec. Setting up Associated Connections is covered in the *Creating and Using Associate Connections* section on page 43.

Using the Off Line Bus

The DMX Surface doesn't have dedicated Off Line bus assignment buttons, so the PGM 4 bus assign button can be setup to be Off Line assign button for any source signal that need to talk to the callers/remotes when their channel is off. This is typically setup for the board operator and/or the host mic, but also can be setup for the codecs and hybrids so they can talk/hear one another when their channels are off). This functionality is setup using the VDips page tab in the DMX Surface Setup app. See the *PGM 4 / Off Line Options* section on page 26.

Once you've assigned the PGM 4/Off Line settings the board operator's mic channel is assigned to the Off Line bus by lighting the PGM 4 button. While the caller/remote channel is off, and assigned to cue, the board operator/host can hear the caller/remote in the cue speakers (or in their headphones). The caller/remote automatically hears the board op in their bus-minus outputs allowing for a hands-free conversation. If a Host or a call screener also needs to talk to the caller/remote, then their mic channels are also setup for PGM 4/Off Line and then are assigned to PGM 4. They would then talk to the caller using their mics and would hear the caller/remote through the cue speaker.

Codecs, Hybrids, & Other 4-Wire Device Connections

Each caller/remote device's *From Network* signal typically connects to a Mix Engine or Razor analog or digital input with two devices sharing a common RJ45 jack input (use Navigator to set any input up for two mono signals). A Razor mono analog or digital output would typically be used to connect to the *To Network* input on the caller/remote device. The caller/remote Destination is connected, in Navigator's Crosspoint grid, to the bus-minus Source for the channel that caller/remote is on.

Of course, any stereo output on a Mix Engine could be connected to a caller/remote but since the Mix Engine outputs cannot be split into two mono signals, you'll end up losing one mono output by having to do it this way.

Signal Gain & Balance Adjustments

Navigator has level controls for every physical input and output on the Mix Engine and Razor (see *Level & Balance Controls* on page 41). Up to 18 dB of gain or 18 dB of trim from unity gain (0 dB) can be applied in real time to the audio. Viewing and adjusting the level controls for the inputs is done in the **Blade > Sources** tab for each Mix Engine, Razor, or Blade. The outputs are viewed and adjusted in the **Blade > Destinations** tab for each Mix Engine, Razor, or Blade.

Monitor Muting & Hot Mic Logic

Although any signal can be set to mute either monitor output, typically only mics are set to mute an output. For Control Room mics you'll want to mute the CR and CUE outputs, unless Cue only feeds the studio, in which case the Studio mics would be set to mute the STU and CUE outputs. These settings are made using the DMX Surface Setup app's *VDips page tab* (details on using the VDips Page Tab is on page 25).

Setting Monitor & Meter Button Sources

The DMX Surface's monitor section, at the right end of the Surface, has two columns of controls (see Figure 4-4, page 50). The left column (CR) has the meter and Control Room monitor controls; the right column (Studio) has the Events, Timer, and Studio monitor controls.

There are four user-set source selectors (Meter EXT, CR EXT 1 and EXT 2, and Studio EXT) and a square white SOFT button. The Soft button can be assigned to a simple function, like being set as a talk-to-producer button or an automation bypass/console active button, to very complex functions like taking a salvo to setup multiple channels for a specific function like voice tracking while the console remains live-to-air.

To assign a source to the **Meter EXT** or to the **CR Monitor EXT 1** or **EXT 2** buttons, press/hold that button for about three seconds, or until the button begins blinking and the left monitor OLED display switches to show the sources available to assign to that button. Using the Cue/Source encoder, highlight the desired name then click the encoder to assign the highlighted source to the blinking button, which then stops blinking.

Two sources can be assigned to the **Studio EXT** button: one is set while the button is lit, the other is set while it's unlit. Press/hold the unlit EXT button until it blinks (about three seconds) to set the unlit button source. Likewise, press/hold the lit EXT button (for about three seconds) until it blinks to set the lit button source. In each case the right monitor display switches to show available source. Use the Studio encoder to highlight a source name, then click the Studio encoder to assign the source.

A studio monitor "on-the-fly" source can also be set by press/holding the studio encoder for about three seconds, until the list of visible source names appears in the right-hand monitor display. Highlight the desired source, then click the Studio encoder. The selected source name appears in red lettering in the right monitor display (see Figure 2-8 on page 15). Editing the list of visible source names assigned to the EXT buttons is set using the DMX Surface Setup app's *Visibilities page tab* (page 29).

Saving and Taking Events

The four Event buttons (Event 1 – Event 4) are used to first save and to then take console events. A "Save Event" saves the current settings for all the fader channels, which includes the channel source, button assignments, and Advanced Channel Feature settings along with the monitor panel sources and the user-set button settings. Typically, engineering saves the four events, then locks out the Save function from board operator access.

To save an Event, once the console is setup for a specific show, application, or daypart: press/hold the **Event button** you want to use. After about three seconds the button will begin to blink. Continue holding the button for another couple seconds until the button is lit solid. This indicates that the status of each Surface control was saved as that Event. Note the Event number and the show, console function, or daypart saved, and relay that information to the board operators.

Once the desired Events are saved, use the DMX Surface Setup app's *Talent Access page tab* (page 23) to lock-out the board operators from accidentally overwriting saved events by unchecking **Allow Save Events**.

To take an Event, press/hold an Event button until it begins to blink then release the button. This arms the Event. Tapping the blinking button takes that event which quickly updates the DMX Surface settings. The Event button is lit solid to indicate which Event is active on the Surface.

Programming the Soft Button

The white Soft button in the CR monitor control column can be assigned to take a salvo (to make multiple system connections simultaneously), make a momentary connection (useful for talking to an external destination or for toggling a destination between two sources), as an indicator for the hotline phone ringer or indicate other status, or to control a logic output for use as a Dump button for a profanity delay unit with the button LED functioning as a ready or safe indicator.

How the Soft button operates: as a tally-only indicator, as a toggle switch, or as a momentary switch, is assigned in the *Buttons page tab* in the DMX Surface Setup app (page 29). What the button controls, and how the button's LED is controlled, is set in the Navigator app (see the *Assigning Logic* section starts on page 42).

RAZOR INTERFACE NOTES

Razors have eighteen RJ45 jacks: eight for audio inputs; eight for audio outputs; one for Logic LIO; and one Ethernet jack to connect that Razor to a Mix Engine Ethernet switch port or to a facility AoIP switch in a larger WNIP network.

On a **Razor 16A**, sixteen of the RJ45 connectors are used for analog inputs and outputs with each carrying two analog signals (stereo or dual mono). The Razor 16A also has two built-in mic preamps, just like the Mix Engine, with two female XLR **Mic Inputs** and an additional **Mics Out** RJ45 jack which typically connects to the Analog 1 Input using a short CAT5 patch cord. Note that Input 1 is factory-set for two mono inputs on this model for that purpose. See *Changing Signal Type* on page 40 for details on changing the Razor input signal type.

On a **Razor 16D**, the sixteen RJ45 inputs and outputs each carry one stereo AES/EBU signal. Any of these inputs or outputs can be split into two mono signals using Navigator (page 40, *Changing Signal Type*).

On a **Razor 16AD** jacks 1 – 4 are analog inputs or outputs while jacks 5 – 8 are AES/EBU inputs or outputs.

Note: Even though every Razor input and output could be set for dual mono signals, there are two caveats to doing this:

1. The eight input jacks can all be set for 16 mono sources. Each mono source can connect anywhere within the system with no connection limitations.

2. Although the eight output jacks can also be setup for 16 mono destinations, because the Razor only receives eight simultaneous audio streams this can restrict the total number of simultaneous mono signals that Razor can receive, depending upon their sources. If each mono source comes from a different Blade, which is not likely, you could only receive eight mono signals. But, if the mono sources come from less than eight Blades, there won't be an issue in feeding 16 mono signals to the 16 mono destinations.

If one does exceed mono sources from eight Blades, a Navigator error message "No stream receivers available" will appear when attempting to make the Crosspoint.

LOGIC – This jack is used to connect hard-wired logic to external devices like warning light interfaces, remote control panels, and tallies. Each Logic connector is wired per Figure 2-4 on page 12. The logic connections are configured using Navigator to be either a logic input or a logic output with an assigned logic function (see *Assigning Logic* on page 42).

FRONT PANEL DISPLAY & ENCODER - Each Razor has an OLED display and a rotary encoder to select between the four displays available on the Razor.

Figure 2-10 shows the four screens. The first two screens show the Razor's technical details. The first screen shows its system name, device ID#, IP address, sample rate, Ethernet connectivity (icon in the lower left corner), and how many days and hours it's been in operation. A second technical display shows the Razor's ID and IP address along with the type of Razor, its MAC address, software revision, and hardware/firmware build.

The third display (Inputs) shows the average levels of the sixteen inputs, which can be any combination of up to eight stereo or up to 16 mono inputs. The last display (Outputs) shows the average output level for each output (any combination of up to eight stereo or up to 16 mono). Each input and output signal can have its gain boosted by up to 18 dB or trimmed by 18 dB using Navigator's input or output gain controls.

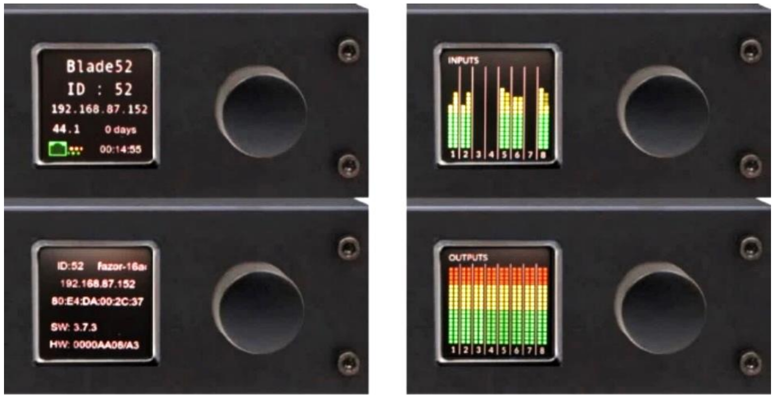


Figure 2-10 Razor front panel OLED displays

The DMX console's factory-default configuration settings allow it to be powered up and used straight out of the box. To configure the DMX for your specific application these default settings are edited using **Navigator** and the **DMX Surface Setup** app. The app installers are on the USB flash drive which ships with the DMX, but they can alternately be downloaded. To request the download links for those apps, email techsupport@wheatstone.com.

Install the two apps on an **admin PC** (any current Windows PC can be used). Before installing the apps connect the admin PC to the Mix Engine switch using **Port 1**, **2**, or **3** (Figure 2-3 on page 11). We recommend setting the admin PC's Network Interface Card (NIC) to **192.168.87.11** with a **255.255.255.0** subnet mask when the Mix Engine is on the factory-default **192.168.87.0** subnet. The Mix Engine's default IP address is **192.168.87.101**, and its name is **Blade01**. The DMX Surface has a default IP address of **192.168.87.201**, and its name is **Surf01**.

To ensure both apps get installed properly, right-click on the installer icon or its file name and select *Run as administrator*. Shortcut icons are added to the desktop. On a Win10 PC, the apps are added to a *Wheatstone* folder and to the Recently Added app list in the Start Menu.

Appendix A has the steps to take to change the default DMX Mix Engine and Surface settings, which you'll need to do before networking multiple DMX consoles together or when your WNIP network is using a subnet other than the factory-default **192.168.87.0** subnet.

DMX SURFACE SETUP APP

Double-click the desktop icon (shown at right) or use the *Start menu* to select **DMX Surface** from the *Wheatstone* folder. The DMX Surface Setup app opens showing the *Device Properties* tab. Figure 3-1 identifies the main controls and the Devices and System Info panes which can be "torn off" and viewed in stand-alone windows.

Click on the *Locator* tab to view it (Figure 3-2). This tab lists each WheatNet-IP (WNIP) device connected to the Mix Engine's Ethernet switch. For a new DMX console only two devices will be listed: the Mix Engine and the DMX Surface. Figure 3-2 shows that a TS-22 talent station and a Razor Interface are also connected to the built-in gigabit switch.

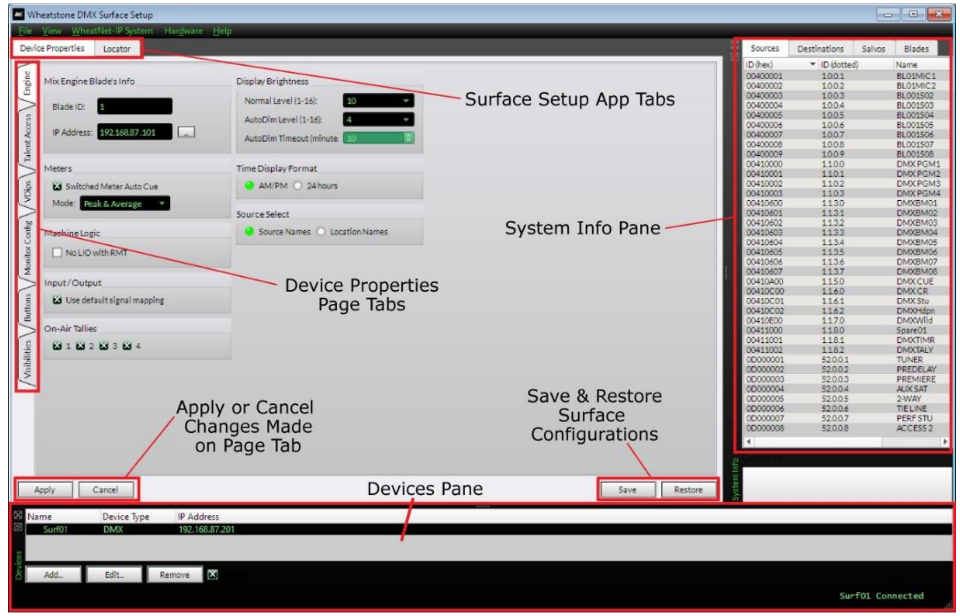


Figure 3-1 DMX Surface Setup, Device Properties Tab View

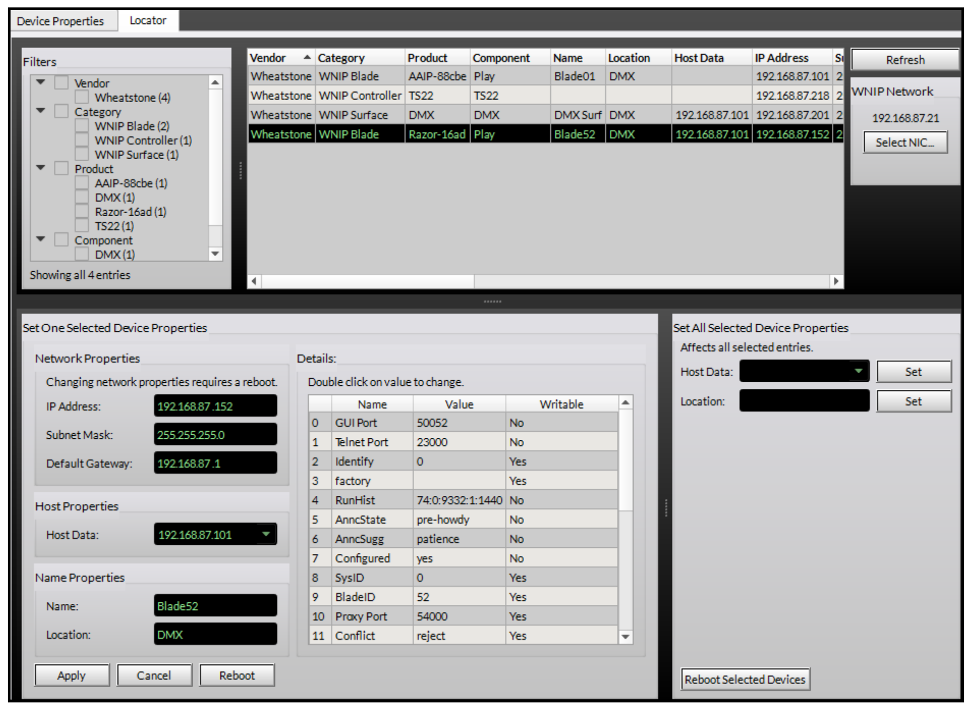


Figure 3-2 DMX Surface Setup, Locator Tab View

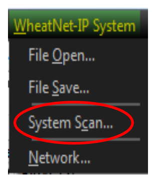
If no devices are listed, verify the correct IP address is shown above the *Select NIC...* button. If the wrong NIC is listed, click *Select NIC...* to open the **Network Setup** window to select the correct NIC. If the NIC is correct, confirm that the admin PC, Mix Engine, and DMX Surface are all connected to the Mix Engine's Ethernet switch, then click *Refresh* to re-query the network for WNIP devices.

If clicking *Select NIC...* shows no NIC with an IP address in the default subnet (192.168.87.0), close the DMX Surface Setup app and use the Windows Control Panels to assign a fixed IP address, like 192.168.87.11 to the NIC connected to the DMX, then restart the DMX Surface Setup app.

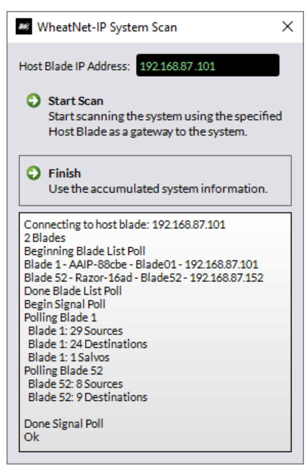
Once your devices are listed, click on a device row in the *Locator* pane (a Razor Interface is shown selected in Figure 3-2 on the last page) to show details for the selected device in the *Set One Selected Device Properties* panel. The default settings for devices can be manually edited to change the assigned IP address, Device ID, Host Device, Name, and Location when networking multiple DMX consoles and Razors together.

Note: When multiple DMX consoles are being networked, because every DMX ships set to the same Device ID (1) and IP address (192.168.87.101), you'll need to use Navigator's Locator tab to edit the default entries in the *Device Properties* pane or else do a factory reset on that Mix Engine to change its IP and Device ID. Appendix A covers how to change the default settings to avoid IP address and Device ID conflicts.

Setting the Surface Host



Before the DMX Surface can be configured, the WNIP system must be scanned to identify all networked devices to update the DMX setup app. Select the menu item *WheatNet-IP System* then *System Scan*. The **System Scan** window opens (shown below right). The *Host Blade IP Address* (192.168.87.101, the Mix Engine's default IP address) should be shown. If no address is listed manually enter that IP address into the entry box, then click *Start Scan*. The app queries that Host Blade (the Mix Engine) showing its activity in the logging window.



Once *Finish* appears in lieu of *Cancel*, click it to close the System Scan window. A "save system info?" dialog pops up. Click *Yes* to save the system info. A Save File dialog box opens showing the default save location. Click *Save* then *Yes* to save the **sysinfo.wnsi3** data file, which saves your system information to the admin PC.

The *System Info* pane, at the right side of the Setup app's window, will now be populated with *Source* names, as was shown in Figure 3-1 on the last page. Clicking on the other tabs: *Destinations, Salvos,* and *Blades* will show the scanned Destination names, that no *Salvos* have been created yet, and that the Mix Engine is **Blade ID 1** with the default name of **Blade01**.

Click on the *Device Properties* tab to return to show that tab. There are six **Page Tabs** (Engine, Talent Access, VDips, Monitor Config, Buttons, and Visibilities) along the left side of this tab (Figure 3-1 on the last page). The page tabs allow various DMX parameters to be viewed and edited to configure the DMX for your specific application. Click on each page tab to see what each has to offer then reselect the *Engine* page tab.

Along the bottom of the default Surface Setup app's window is the *Devices pane*.

Note: If this pane is not displayed, select the menu item *View*, then check *Devices* to display the *Devices* pane, as shown at the bottom of Figure 3-3.

Click the *Add* button to open the Add Device dialog box. In the Add Device window click *Locate...* to open a Locate window. If **Surf01** is not listed, click *Rescan*. Once **Surf01** is shown, click on its name to highlight it then click *OK*. The new DMX Surface name (**Surf01**) and its IP address (**192.168.87.201**) are automatically entered into the *Add Device* dialog box and the *Device Type* is set to **DMX**. Click *OK* to transfer these settings to the Devices pane.

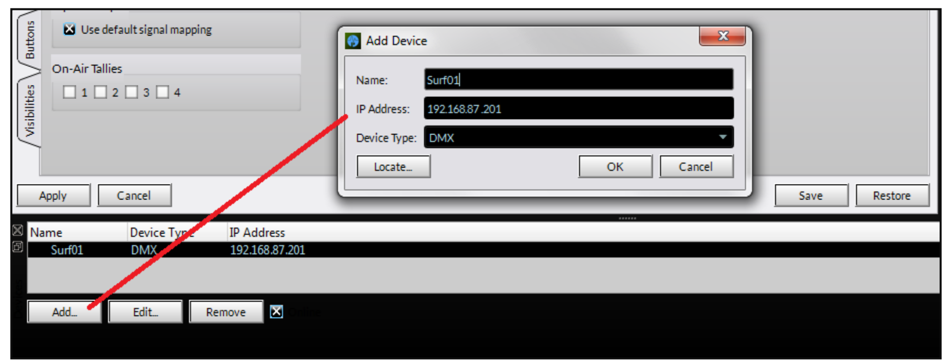


Figure 3-3 Selecting the DMX Surface in the DMX Setup App

You can now connect to that specific DMX Surface by clicking on its row to highlight it, then click the small square next to the *Remove* button to "X" that box. That action instructs the app to connect to the selected Surface. It should only take a few seconds before **Surf01 Connected** appears in the lower right corner of the Devices pane, which indicates that DMX Surface is now connected to the DMX Surface Setup app and its current configuration settings were uploaded to populate the various page tabs on the Device Properties tab.

View the *Engine* page tab on the Device Properties tab. At the upper left corner is the *Mix Engine Blade's Info* section. Verify that **Blade ID: 1** and **IP Address: 192.168.87.101** are displayed (Figure 3-4). If those fields are empty, you'll need to manually assign the Mix Engine as this Surface's "host device." Click the *Picker* icon ( ) to open the **Blade Picker** which lists all blades detected by the System Scan. Double-click the **Blade01** row to assign that Mix Engine to be the host device for this currently connected DMX Surface.

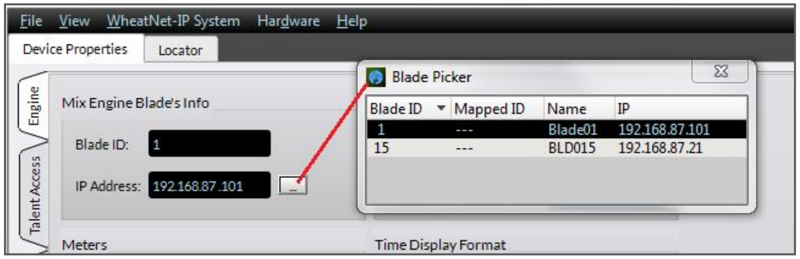


Figure 3-4 Setting the Mix Engine as the Surface Host

If the two Mix Engine entries were empty, and you used the Picker window to assign the Mix Engine, click the *Apply* button at the bottom left corner of the Device Properties tab to download the new settings to the DMX Surface.

Note: Several common settings, like editing the console's default signal names, editing their signal formats, and configuring logic, are done using **Navigator**. Information on using that app begins on page 31.

Device Properties Page Tabs

The DMX Surface Setup page tabs are divided into sections to group similar configuration settings. Changes made on a page tab are all "pending" until the *Apply* button, just below the page tabs, is clicked to download the changes to the DMX Surface.

When you make changes on a page tab but don't click *Apply* before clicking on another page tab, you'll get a pop-up warning (Figure 3-5) about applying changes to the previous page tab. Clicking *Yes* applies the changes. Clicking *No* discards all changes made on the previous tab since the last time *Apply* was clicked.

Note: We recommend **unchecking** *Do not show this warning again during this session* prior to clicking Yes or No so that you'll keep getting that popup. If it's left checked, the popup will no longer appear but since the default selection is **No**, when you switch to a new page tab any unapplied changes on the previous page tab are discarded.

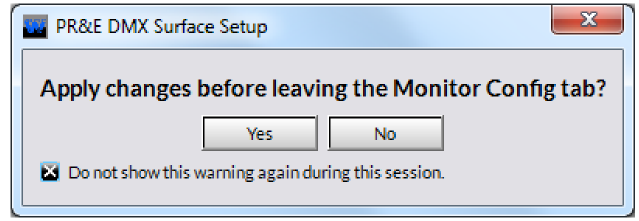


Figure 3-5 Apply Changes Warning Pop-Up

Engine Page Tab

The *Engine* page tab has eight sections which impact overall Surface operation: Mix Engine Blade's Info (covered in *Setting the Surface Host* section), Meters, Machine Logic, Input/Output, On-Air Tallies, Display Brightness, Time Display Format, and Source Select.

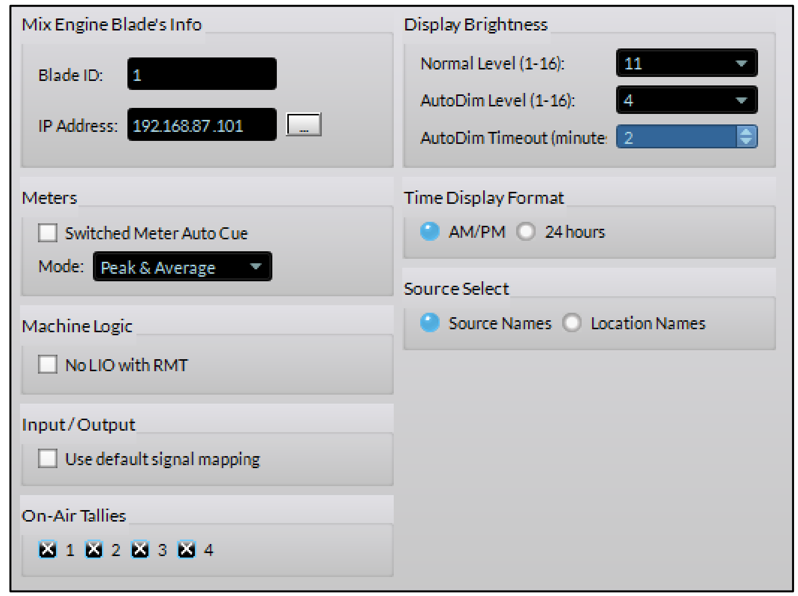


Figure 3-6 Engine Page Tab

Meters

When *Switched Meter Auto Cue* is checked (click the box to add an X) the cue levels are shown in the switched meter when cue is active, which overrides the selected source (PGM 3, PGM 4, or EXT). When unchecked, the default setting, the switched meters do not show the cue levels.

The *Mode:* setting affects all three meters. *Peak and Average* is the default setting, but the meters can alternately show *Average Only* or *Peak only* for special applications. The Average signal level is shown using a bar graph display while the Peak level is shown using one LED per channel, to indicate the highest instantaneous audio level, typically 6 – 12 dB above the average level bar graph. Note that the Peak decay time is fixed.

Machine Logic

This sets whether there's a logic output (either a tally or a start/stop pulse) when a remote channel on/off command is received. When *No LIO with RMT* is unchecked, the default setting, tally or remote start/stop commands are sent out with remote channel on/off commands. When this option is checked, no tally or remote start/stop command is sent when the channel is remotely turned on or off.

Input/Output

When *Use Default Signal Mapping* is checked, the default signal connections to the Mix Engine's four monitor and four program outputs are remade and locked when either Mix Engine or Surface is rebooted.

Note: If you change any monitor or program output connections from their default settings you must uncheck this option or else your new connections will be overwritten with the default connections whenever the Mix Engine or Surface gets rebooted.

On-Air Tallies

The four on-air tallies (1 – 4) are usually all checked even though, typically, only Tally 1 (Control Room warning) and Tally 2 (Studio warning) are regularly used. Tally 3 and Tally 4 are available to trigger hot mic warning lights in external locations like an announce booth and a newsroom or sports bullpen. The tallies are triggered when the logic function: *Studio 1 In-Use* up to *Studio 4 In-Use* is assigned as the logic output command for an LIO or SLIO.

Display Brightness

These three parameters control the OLED displays' brightness levels. The *Normal Level* setting sets the brightness of the OLED displays from 16, the maximum OLED brightness, to 1, which is virtually off. Because OLED displays will burn-in when displaying the same graphic or name over a long period of time, and will also dim over time, we don't recommend setting your new console to use a *Normal Level* of 16. The default level is 10, but you'll want to use the lowest setting which still yields an easy-to-read display. In a production room with low lighting this might mean a setting of 6 matches the ambient lighting in a production studio. To see the new setting reflected in the OLED display brightness, click *Apply* to transmit the updated setting to the Surface.

The *AutoDim Level* setting is used with the *AutoDim Timeout* entry. When the *AutoDim Timeout* is reached, the OLED displays dim to the assigned *AutoDim Level* setting. Setting the *AutoDim Level* to 1 effectively turns the displays off. The default entry is 4.

The *AutoDim Timeout* entry sets how many minutes, from one to 59 must pass after a Surface control was last touched, before the OLED displays switch from the *Normal Level* to the *AutoDim Level*. The default setting is 10, which means if no Surface control has been touched for ten minutes, the OLED displays automatically dim down to the *AutoDim Level*. When an operator touches any control, the dimmed OLED brightness immediately returns to the *Normal Level*.

Note: Setting the timeout to 0 defeats auto dimming and the *Normal Level* setting is always used. While this may be desirable during console configuration, we recommend always setting the *AutoDim Timeout* between 2 and 10 minutes with the *AutoDim Level* set between 2 and 4 so a dim image still appears in the OLED displays.

Time Display Format

The right monitor display shows the time-of-day in standard USA 12-hour format (*AM/PM*) or military/European format (*24-hour*). The left monitor display always shows the date as Month Day Year. The current date and time must be updated periodically using Navigator's *Info tab* (see page 34 *Info Tab* for details).

Note: The date and time must be updated using Navigator whenever the DMX Surface is power cycled.

Source Select

The default selection is *Source Names*. When any encoder is used to select a new source, all visible source names are listed alphanumerically with their host Blade name appearing above the white line in the display.

When *Location Names* is selected, using the encoder to select a new source first shows a list of Blade names. Clicking a Blade name then shows all the visible sources on that Blade. To view signals on another Blade, click << *locations* to step back to the Blade list so that another device can be selected. Most users leave *Source Names* selected.

Talent Access Page Tab

This page tab (Figure 3-7 on the next page) sets board operator access to various Surface functions. Three sections: *Events, Bus Assignments*; and *Channel Features* control access to the Event buttons, bus assignment buttons, and the three Advanced Channel Features. When unchecked, a board operator cannot change or select that feature.

The *Timeout* section sets whether the Mode, Panning, and EQ & Dynamics displays time-out (option checked) or stay active until they are manually closed (unchecked).

The *EQ* section allows the current EQ and Dynamics settings set on any channel to be pulled from and saved to an "EQ config file" or a saved EQ config file can be pushed to be applied to another DMX fader channel.

While configuring the console all selections are checked except for the *Timeout* entry. When it's unchecked, the Advanced Channel Features will not auto-timeout, which makes setting up channel EQ and Dynamics a lot easier.

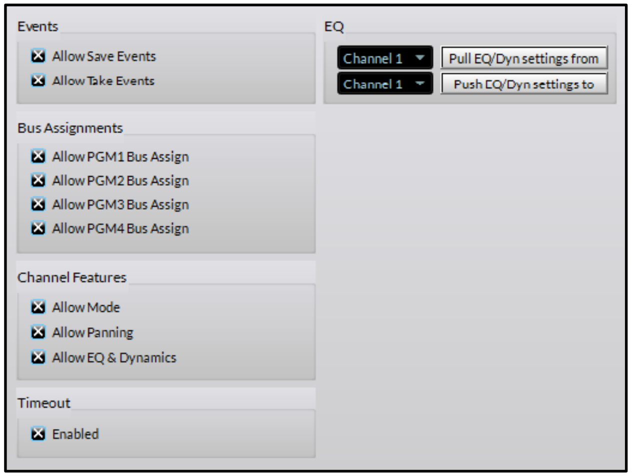


Figure 3-7 Talent Access Page Tab

It's important to selectively limit access to Advanced Channel Features before the DMX is released for daily use. Typically, in an on-air studio, *Allow EQ & Dynamics* is unchecked once these are set during configuration since, in the wrong hands, access to these controls could result in one or more channels suddenly having very "strange sounding" audio!

Allow Save Events is also typically unchecked once the four console events have been saved during console configuration.

Timeout Enabled is normally checked so that any enabled Advanced Channel Features (like *Mode* and *Panning*) will timeout (return to a normal display) after about twenty seconds.

Events

These two selections set board operator access to the four **Event buttons** in the monitor section of the DMX Surface. The Event buttons are used to first save, for later recall, up to four console setups. A console setup includes the current monitor settings, current channel sources and assignment button settings, and the Advanced Channel Feature settings. By saving the four most-commonly-used board setups, each can later be recalled by the board operator using the four Event buttons.

For example: **Event 1** could be used to save the console settings for a morning show with multiple mics and a remote traffic service; **Event 2** could setup the console for a single board operator; **Event 3** could set up the console for a regularly scheduled live remote; and **Event 4** could set up the Surface for overnight or unattended/automated operation.

When *Allow Save Events* is checked, anyone can save an Event—which means a previously-saved Event could be overwritten. When unchecked, the Event Save function is not active. This selection is typically unchecked once the four Events are saved during console configuration.

When *Allow Take Events* is checked, the board operator can take Events. When unchecked the four Event buttons have no function. If *Allow Take Events* is unchecked, then *Allow Save Events* should also be unchecked.

Bus Assignments

These four selections set whether the board operator can change the **PGM assignment button** settings across all console channels. For most applications all four buttons are checked to allow the board operators to freely set any bus assignment button on/off, as required.

Unchecking one or more *Allow PGM Bus Assign* selections (and then clicking Apply) means the PGM button settings, as currently set on the console, cannot be changed. This might be done for consoles that serve dual purposes, like simultaneous on-air and network origination. The PGM buttons would be pre-assigned for the correct operational mode and then unchecked to prevent board operators from changing those PGM bus assignments.

Channel Features

This section sets which Advanced Channel Features are accessible by the board operators.

Allow Mode, when checked, allows a fader channel's audio mode to be changed from Stereo (the default setting) to Left only, Right only, or Mono sum. When unchecked, this feature is locked out from board operator use.

Allow Panning, when checked, allows left-right panning control of the audio signal on that fader channel. When unchecked, this feature is locked out from board operator use.

When the DMX Surface Setup app is on an air studio PC, assign a user password (File > Set Password) to prevent app access, if only to keep Allow EQ & Dynamics from being checked—assuming it's been unchecked prior to releasing the console for daily usage. In the wrong hands, changing the EQ & Dynamics settings could result in a channel sounding "very strange" or even becoming severely distorted.

Allow EQ & Dynamics, when checked, allows the four-band parametric equalization, the high and low pass filters and shelving, and the audio dynamics (compression and expansion) to be adjusted and set for any channel. This feature should only be accessed by engineering while configuring the console. Once the EQ and Dynamics are set, *Allow EQ & Dynamics* should be **unchecked** to prevent board operators from accessing this powerful feature.

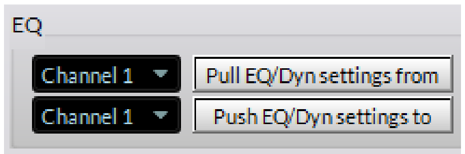
Details on using the Advanced Channel Features are presented in Chapter 4 (starting on page 52).

Timeout

Unchecking *Enabled* allows any Advanced Channel Feature to remain displayed until manually changed by the user—a very useful function for adjusting EQ & Dynamics during console configuration. Once console configuration is finished, check *Enabled* so that any Advanced Channel Feature will automatically time out and revert the channel display (or displays in the case of EQ & Dynamics) to its normal display when no surface control has been touched for about 20 seconds.

EQ

This section's controls allow the EQ & Dynamics settings from any channel to be saved to an EQ file. The saved EQ file can then be pushed or saved to any other fader channels, so it has the same EQ & Dynamics settings.



The top row saves the EQ & Dynamics settings from the selected channel to an EQ file. Select the channel with the EQ & Dynamics settings you want to save using the drop-down channel list. Then click the *Pull EQ/Dyn settings from* button. A Save Dialog box opens so you can name the EQ file. The .dmx_eq suffix is automatically added to the name you enter. The EQ files are saved to the WheatNetIP_DMX folder in the PC's Documents folder.

The bottom row allows the settings from a saved EQ file to be loaded into another channel. Select the channel from the drop-down list then click the *Push EQ/Dyn settings to* button. A pop-up warning asks whether you want to alter the settings on that channel. Clicking *Yes* opens a Find File Dialog box so a previously-saved EQ file can be selected. Click *No* to close the warning box.

Note: The EQ & Dynamics settings are immediately applied—even on a channel that's currently turned on, which can noticeably change that channel's audio when the EQ & Dynamics settings are applied.

A Wheatstone_mic.dmx_eq file is included with the DMX Surface Setup app. It has the EQ & Dynamics settings which are applied to channels 1 and 2 at the factory.

VDips Page Tab

This page tab (Figure 3-8) is where console options and logic features are assigned to specific audio sources. Mix Engine inputs and other system signals which need to have associated logic settings, like mics and peripheral devices with remote logic, are manually added to the *Signals* list using the Signal *Add...* button. Once a signal is added, clicking its **Signals list row** highlights that signal and displays its current logic settings in the seven assignment sections on the right half of the page tab. These settings can then be edited as required. To see/hear the effect of the new selections, click *Apply* to transmit the updated settings to the DMX Surface.

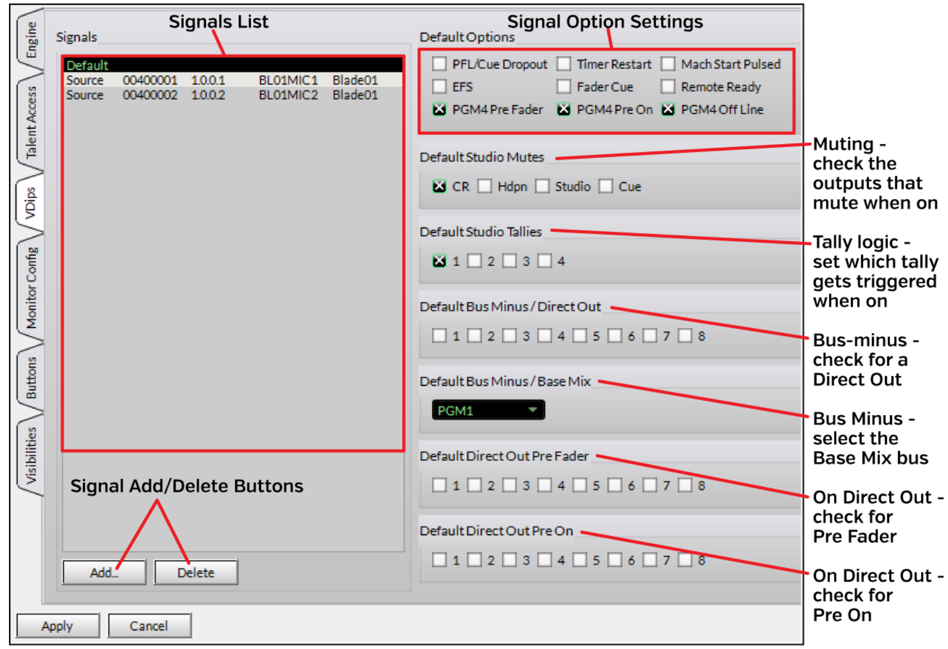


Figure 3-8 VDips Page Tab Sections

Note: Before adding your signals to the Signals list, run a System Scan (see *Setting the Surface Host* section on page 21). When signal names are edited in Navigator, they are automatically updated in the DMX displays, but the DMX Surface Setup must be manually updated by running a new System Scan to get the updated names and format changes made using Navigator.

Default Settings

Before adding any signals to the *Signals* List, read through the descriptions of the various logic settings in this section. You may find selections which should be applied to multiple sources. These common selections can be automatically checked on each newly added signal by clicking the **Default** row, at the top of the *Signals* list, and then enabling the desired "Default" selections.

Figure 3-8 shows the Default row highlighted and the *PGM4 Pre-Fader*, *PGM4 Pre-On*, *PGM4 Off Line, CR Mute*, and *Tally 1* are all checked. In this case, when Mic 1 and Mic 2 were then added, both of those signals were added with these Default settings already assigned. The Default selections can then be edited to allow different types of signals to get added with most settings pre-assigned. For example, changing CR Mute and Tally 1 to Studio Mute and Tally 2 would then allow you to add multiple Studio Mics which will each have those settings.

Be sure to edit the Default settings after you've added signals since any source that's not listed in the Signals List will have the Default selections applied to it. Needless to say if a mute or a tally was left selected, every source signal—not listed in the Signals List, will have those selections applied to it, which could falsely mute monitors or turn on a tally.

Signals List

Sources must be manually added, one at a time, to the Signals list to populate it. To add a new signal, click *Add…* to open the *Add Signal* pop-up selector (Figure 3-9). Click the *Picker* button to open the *Source Picker*. Double-click a signal row to select that signal. This closes the *Source Picker* and adds that signal number to the picker window. Click *OK* to add that source to the Signals list and close the Add Signals box.

Repeat this process (click *Add…,* click the *Picker* button, double-click on another source row then click *OK*) for each source you need to configure. Fortunately, these repetitive process only needs to be done once.

Note: When the *Apply* button is clicked all signal settings, including even for sources which are currently on-air are updated, so use caution when making signal changes on a console that's already on-air.

When any *Default Options* are checked, those options are automatically checked on each newly added signal. The new signal is automatically selected (highlighted) so additional configuration settings can be edited, as required. To see the effect this has on the source channel's functions, click *Apply* to send the updated settings to the Surface.

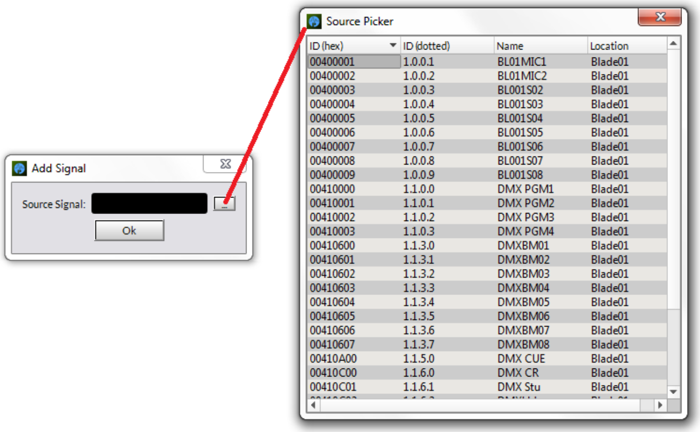
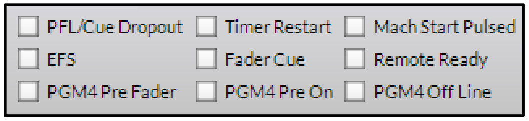


Figure 3-9 Adding a Source to the Signals List

Options



These nine settings affect fader channel operation when the selected signal is the active source for a fader channel.

PFL/Cue Dropout – Controls what happens when the fader channel is turned on. When checked cue (alternately called PFL or Pre-Fader Listen) is automatically turned off, if active, when the channel is turned on. When unchecked, and cue is active, cue remains active when the signal's channel is turned on. If *Fader Cue* is checked, this option is unchecked.

Timer Reset – When checked, a Timer Reset command is sent to the timer each time that signal's channel is turned on. If unchecked no timer reset is sent to the timer.

Note: For the Timer Reset function to reset the timer, the **Timer Auto** button must be lit on the Surface. When the **Timer Auto** button is unlit, the timer reset command, even if checked, will not reach the timer.

Machine Start Pulsed – Sets the type of logic command sent to the signal's remote logic. When checked, a Start Pulse is output. When unchecked, a sustained closure (tally) occurs for as long as the source channel is on.

EFS (*Electronic Fader Start*) – Controls whether moving the signal's channel fader affects channel on/off. When checked, moving the fader to full off turns the channel off and moving the fader away from full off turns the channel on. When unchecked, fader movement does not affect channel on/off status. This command is typically assigned the same for all signals although some users only enable this feature on microphones.

Fader Cue – This setting is like *EFS* in that fader movement affects channel status. When checked, moving the fader to full off turns cue on and moving the fader away from full off turns cue off. This action emulates a feature found on analog consoles with rotary volume controls which put the channel into cue when the pot was turned to full off. When unchecked, fader movement has no effect on cue—unless *EFS* and *PFL/Cue Dropout* are both checked, in which case cue is turned off when the fader is moved from full off.

Remote Ready – When unchecked, the standard setting, the Off-button LED lights up while the channel is off. When checked, the Off-button LED is controlled by that signal's Ready command logic which is assigned in **Navigator** (see *Assigning Logic* on page 42).

PGM 4 / Off Line Options

The PGM 4 bus differs from the PGM 1 – PGM 3 buses in that specific signals can be set to feed PGM 4 pre-On button and/or pre-fader by checking the *PGM 4 Pre-On* and *PGM 4 Pre-Fader* selections, while leaving the *PGM 4 Off Line* selection unchecked. This would allow the PGM 4 bus to feed an FX unit or an air skimmer.

When callers and/or remotes (any Telco or 4-wire signal from a hybrid, codec, two-way device, etc.) will connect to the console, the *PGM 4 Off Line* option should be checked on their signals. This option switches the PGM 4 buttons to instead be Off line bus assignment buttons. The PGM 4 buttons are then used to set which channels get fed to the bus-minus signal going back to the caller/remote while their channel is turned off. When the caller/remote channel is turned on, their bus-minus signal switches to the bus assigned in the Bus-Minus/Base Mix section (most often PGM 1, but it can be set to any PGM bus including PGM 4) so the caller/remote hears everything else going to air.

This means that when *PGM4 Off Line* is checked—**on any signal**, the PGM 4 buttons then function as Off Line bus assign buttons. Typically, when the Off Line bus is active the PGM 4 bus is not used. When *PGM4 Off Line* is unchecked **for all sources**, the PGM 4 buttons assign the channel sources to the PGM 4 bus like normal and the Off Line bus is not used. In this condition, the caller/remote bus-minus signals are always derived from the bus assigned in the Bus-Minus/Base Mix section.

Note: The *PGM4 Pre-Fader* and *PGM4 Pre-On* settings affect both PGM 4 and the Off Line audio. This means the PGM 4 output, which is still active even when *PGM4 Off Line* is checked, could have some sources active all of the time (if *PGM4 Pre-On* is checked) and those could be at a fixed level (if *PGM4 Pre-Fader* is checked), which could lead to unexpected operation. Thus, we recommend **not using** the PGM 4 bus when Off Line is active (i.e., when the *PGM4 Off Line* option is checked for any source). Navigator can be used to connect a different source to the output labeled as PGM OUT 4 on the Mix Engine.

PGM4 Pre-Fader –When checked, the signal feeds the Off Line and PGM 4 buses without fader level control (the same level as cue). When unchecked, the channel fader affects the level feeding both Off Line and PGM 4.

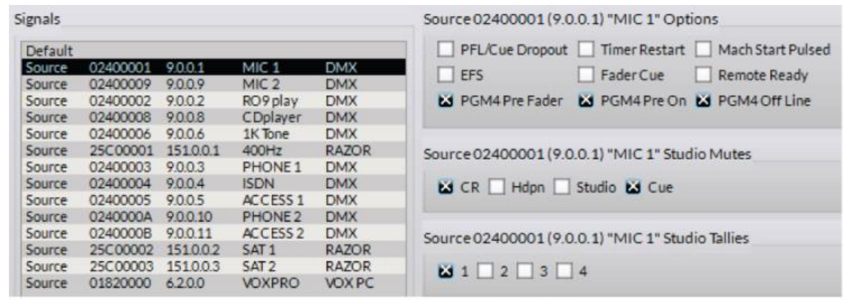
PGM4 Pre-On – When checked the signal always feeds the Off Line and PGM 4 buses, unless *PGM 4 Pre-Fader* is unchecked, in which case the fader still controls the level. When unchecked, the signal only feeds the Off Line and PGM 4 buses when the channel is turned on. Typically, this setting is checked for all sources so that no channels must be on in order to feed the Off Line bus.

PGM4 Off Line – This setting affects the bus minus signal sent to callers and remotes. It also affects how the PGM 4 assignment buttons function. When checked on any source, the PGM 4 buttons also assign the channel to the Off Line bus. When unchecked on all sources, the PGM 4 buttons assign channels to the PGM 4 bus and the Off Line bus is not active. For a caller/remote this means they always hear the bus assigned in the Bus Minus/Base Mix selection (default setting is PGM 1 for all sources).

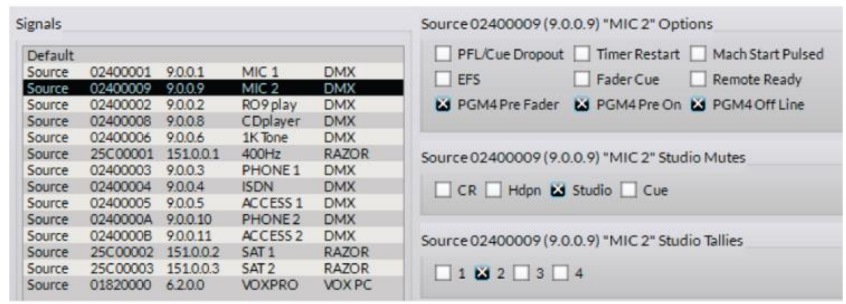
Studio Mutes

Any signal can be set to mute one or more of the monitor outputs when their channel is turned on, but typically only microphones are set to mute outputs. For mics located in the Control Room, both the *CR* and *Cue* outputs are typically checked (muted). For mics located in a Studio, typically only the *Studio* output is checked (muted), unless cue is fed to the studio, then *Studio* and *Cue* would be checked.

Usually, the HDPN output is never checked. Figure 3-10 shows the typical settings for a CR mic and a Studio mic.



Typical Control Room Mic Settings



Typical Studio Mic Settings

Figure 3-10 CR & Studio Mic Settings

Studio Tallies

Each mic is typically set to trigger one of the Tallies. Tally 1 is typically used for the Control Room while Tally 2 is typically used for an associated talk studio. Tallies 3 and 4 can be used to trigger hot mic warning lights at two additional locations like a producer position, call screener, newsroom, etc. Assigning a Tally activates two Surface displays when a mic channel is on: a red **On-Air indicator** lights up in the Meter Bridge; and a **mic icon**, shown in the channel display to the left of the Mode name, which turns red. These mic status indicators will not be shown if a mic is not assigned to a light a Tally.

Bus-Minus / Direct Out

The DMX includes an internal mono signal that's associated with each fader channel. For a DMX-8 Surface there are eight check boxes, for a DMX-16 Surface there are sixteen check boxes. These signals have the default names: DMXBM01 – DMXBM16, with the BM standing for *Bus-Minus*, which are audio signals specifically used with callers and remotes. Any of the mono "BM" signals can alternately be used as a Direct Out signal to feed an outboard FX unit, air skimmer, or system monitor.

The settings in this section set which type of signal is generated when the signal is on a channel. If no boxes are checked a bus-minus signal is created on any channel that signal is on. When one or more boxes are checked, and that signal is taken on a checked channel, a Direct Out signal is created for that channel.

Note: The Direct Out audio is affected by the settings in the *Direct Out Pre-Fader* and *Direct Out Pre-On* sections.

For a caller or a remote device, no channel boxes should be checked so that a bus-minus signal is created when that those devices are taken on a fader. This means an IFB (Interruptible Fold Back) audio signal is created that which can be returned to the caller/remote so they can either hear the board operator or hear the program audio (always minus their own audio). This also means that any fader channel can have a codec or a phone hybrid on it since every channel has its own unique IFB signal.

Bus Minus / Base Mix

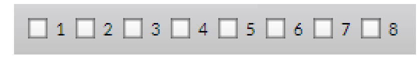
This selection sets which bus: PGM 1, PGM 2, PGM 3, or PGM 4 is used to create a bus-minus IFB (the BM signal) which would be sent back to the caller/remote while their channel is on—assuming the Off Line bus is used to feed the caller/remote while their channel is off. The default, and normal selection, is PGM 1.

Note: When *PGM4 Off Line* is not checked on any signal, this setting sets the bus that feeds the channel Bus-Minus signals regardless of whether the caller/remote channel is on or off. The board operator would then have to use the TB button on each caller or remote channel to talk to that caller or remote.

Direct Out Pre-Fader

When checked for the same channel numbers as set in the Bus-Minus/ Direct Out section, the Direct Out audio is sent at a fixed level (same level as the cue bus). When unchecked, the level feeding the Direct Out is controlled by the channel fader.

Direct Out Pre-On

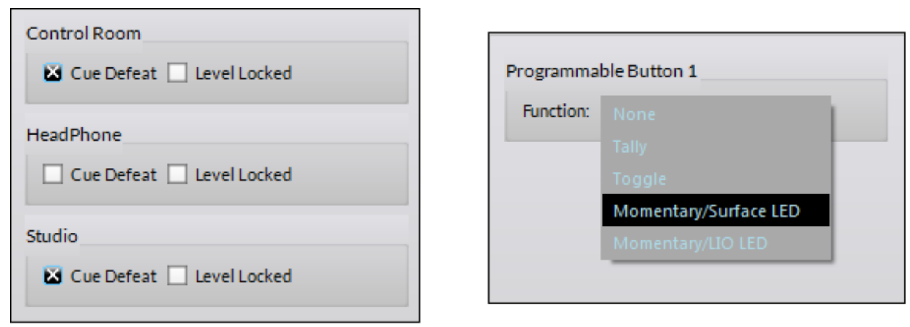


When checked for the same channel numbers as set in the Bus-Minus/ Direct Out section, the direct out audio is not affected by the channel on/off status. The audio is always feeding the Direct Out unless *Direct Out Pre-Fader* is unchecked, then the channel fader affects the level going to the Direct Out. When unchecked, the Direct Out audio follows the channel on/off status.

Monitor Config Page Tab

This page tab's settings affect the CR, STU, and HDPN monitor outputs on the Mix Engine. The Headphone settings also affect the board operator's headphone jack located in the right side-panel on the Surface.

Cue Defeat - When checked the cue system does not affect that monitor output. When unchecked, any channel with cue turned on gets switched into that monitor output replacing the monitor audio. The normal setting is *Cue Defeat* checked for the Control Room and Studio and unchecked for the Headphone output (as shown in Figure 3-11, left illustration).



Monitor Config Page Tab   Buttons Page Tab

Figure 3-11 Monitor Config & Buttons Page Tab Entries

When cue is set to auto-switch into the board operator headphones (Headphone *Cue Defeat* unchecked), the *Split Cue* button in the monitor section on the Surface sets how cue is fed into the headphones. When the Split Cue button is unlit, cue is fed in stereo, cutting off the monitor audio. When the Split Cue button is lit, cue is fed to the left ear while the CR monitor source is fed to the right ear.

Level Locked - When checked, and the *Apply* button is then clicked, the current level for that level encoder (CR, STUDIO, or HDPN) is fixed at that output. Turning that encoder will no longer affect the volume for that output. Uncheck *Level Locked* then click *Apply* to return level control to that encoder.

Checking *Level Locked* might be used when a studio-mounted volume control (like that found in a TS22 talent station) is available, checking the Studio *Level Locked* would allow the TS-22 to control the monitor volume. The Headphone *Level Locked* would be checked when mics have talent stations or when an external headphone amp and cabinet-mounted level control/headphone jack panels are used.

Buttons Page Tab

Figure 3-11, right illustration, on the previous page, shows the function choices in a drop-down list which can be assigned to the **Soft button** in the Surface's monitor section. After the function is selected, Navigator is used to configure the logic which has the functions: *Switch 1* and *Switch LED 1*. Here's what each function choice does and what they could be used for:

- **None –** The default setting. The switch and LED have no assigned functions and the button does not light when pressed.
- **Tally –** Sets the Soft button LED to function solely as an indicator with no switch functionality. Add a new logic-only destination in Navigator with the *Direction: input* and the *Function: Switch LED 1* (see *Assigning Soft Button Functionality* on page 41). The source Spare01 is then connected to the new Logic-only destination in the Crosspoint tab. When that logic input is triggered, the Soft button LED lights up. This setting could be used to create a hotline phone ring indicator.
- **Toggle –** Sets the switch to function as a sustained mechanical toggle switch. A logic-only destination is created with the *Direction: output* and *Function: Switch 1*. The Surface itself controls the button LED, indicating whether the output logic is on (LED is lit) or off (LED is off). This setting could be used to control studio access (studio door locked = lit, unlocked = unlit) or used to bypass the console by using it to trigger a Momentary Connection (see *Using the Soft Button* on page 43).
- **Momentary/Surface LED -** Sets the switch to be momentary. A logic-only destination is created with the *Direction: output* and *Function: Switch 1*. The Surface controls the button LED, lighting it to indicate the button is being pressed. This setting might be used to create a talk button to talk to a call screener or to a producer.
- **Momentary/LIO LED –** Sets the switch to trigger a logic output while pressed and sets the LED to be controlled by a logic input. It combines the logic-only settings for Tally and Toggle. This setting could be used to create a profanity delay Dump button, with the LED being the Safe or Dump Ready indicator.

Visibilities Page Tab

At the top of each DMX fader channel is a channel encoder which is used to select a new source for that channel. There are also three source selectors in the monitor section used to set the sources for the Switched Meter EXT button, the CR monitor source EXT 1 and EXT 2 buttons, and the two Studio monitor sources that can be assigned to the Studio EXT button. Each of these selectors has its own **Visibilities List** which sets which source names get displayed while using each selector.

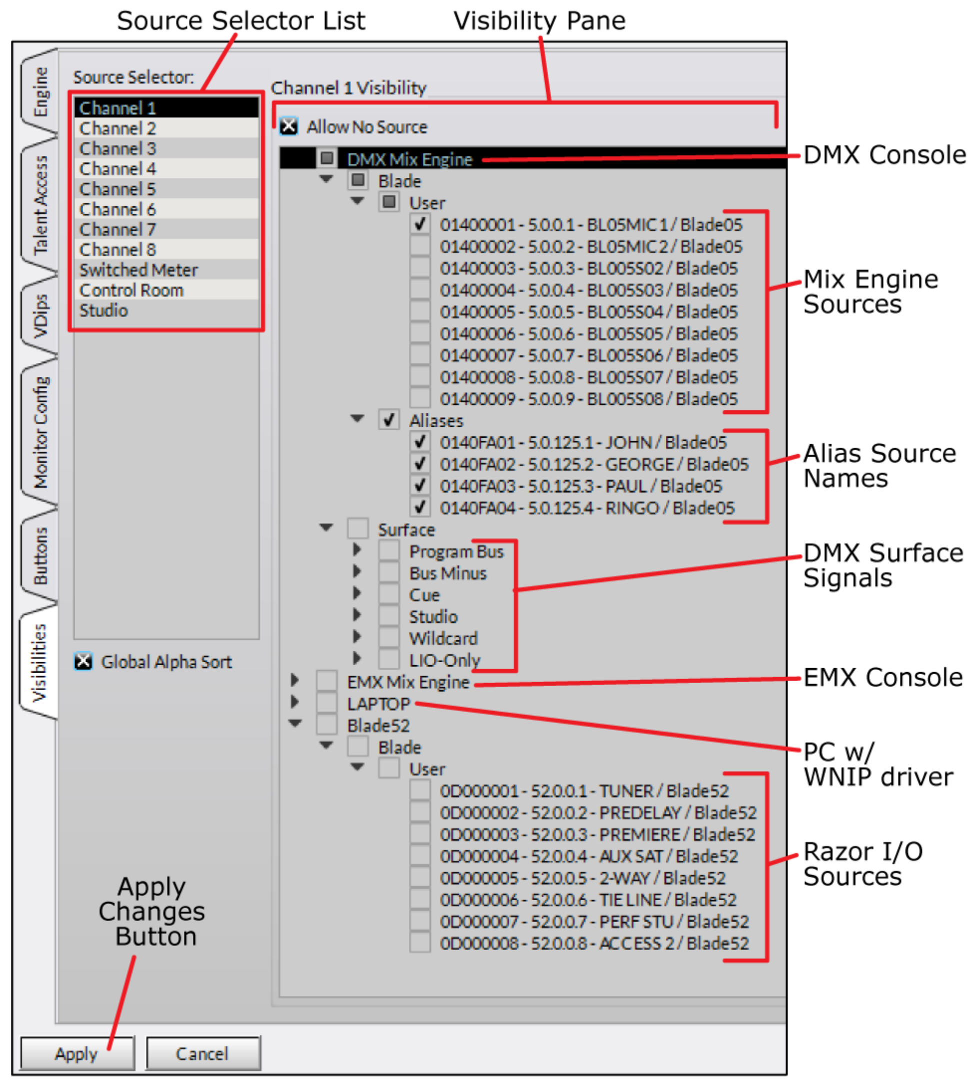


Figure 3-12 Visibilities Page Tab Settings

From the factory the Visibility list for every selector includes all sources which is desirable when installing and configuring the console, but for daily operations it’s essential the Visibilities lists—especially those for the fader channels, be edited to remove all signals which shouldn’t be listed, like the PGM buses and other internal console signals and non-air sources, like pre-delay processor outputs and off-air tuners.

Figure 3-12 on the previous page shows the Visibilities page tab. The Channel 1 Source Selector is highlighted in the Source Selector list. That channel is typically dedicated to the board operator or the host mic. In the example, there are five sources checked: the left channel of the Analog In 1 jack (BL05MIC1), plus four *Alias* names and *Allow No Source*. Thus, when the channel 1 encoder is rotated with the channel off, **NoSource**, **BL05MIC1**, **GEORGE**, **JOHN**, **PAUL**, and **RINGO** will appear in the channel 1 display. Taking any source then displays that source name. In the case of the four aliases, taking any of them connects the BL05MIC1 mic signal to the channel but the OLED display shows the alias or talent name.

The source: **NoSource** can be taken to silence a channel. If *Allow No Source* is unchecked in the Visibilities page tab, then that name won’t appear in the source list. This is typically done on the Switched Meter, Control Room and Studio selectors since they normally wouldn’t be silenced at any time. But it’s typically checked for the channel visibility lists since it’s useful not only to silence a channel, but to also identify when the list has wrapped around since it’s the first name in the visibilities list.

Since the visibility settings on a new console is for every selector to display all sources, before the console gets released for daily operations it’s essential the list of visible sources, on at least the channels be edited—if only to prevent a Program bus from being taken on a channel. If a PGM bus is set as the channel source, and it gets assigned to that same bus and the channel is turned on and the fader is raised, you’ll get “digital feedback” which will overload the bus, causing it to mute. Therefore, the signal category *Surface* is typically unchecked on all fader channels.

Visibility Lists

On fader channels with a dedicated source, like the Host mic or phone channel, only one source, or maybe several talent name Aliases are checked along with **NoSource**.

On media server playback channels, there is often a B Server or backup server, so typically only the Main and B Servers on the media server channels are checked. On fader channels with sources which get used periodically, like a CD player, checking a handful of alternate sources adds flexibility to the non-dedicated source fader channels.

Many users set up a few faders nearest to the monitor section as “remote line selectors” which typically will check every source which might go live to air or need to be recorded. Of course, these lists would still exclude the Surface signals and sources like off-air tuners and processed pre-delay monitor signals.

Caller & Remote Visibilities

In earlier Audioarts consoles, Telco devices (commonly referred to as callers & remotes) could only connect to a “Phone channel,” a dedicated channel which often could only be assigned to one bus at a time since the bus selection typically also controlled the return or mix-minus audio that was going back to the caller.

On the DMX, since every fader channel has its own bus-minus signal, which is typically used as the return signal for each caller/remote, this means there are no channel restrictions so a caller/remote can be set visible on any fader channel.

However, setting a caller/remote visible on multiple channels means that when it’s taken on a channel, that channel’s bus-minus signal must then be connected back to that caller/remote. This can be done manually using Navigator, but it can also be done automatically by creating an *Associated Connection* for each caller and remote device on each channel that they’re visible on. Thus, to minimize how many Associated Connection you need to make we recommend limiting the number of faders each caller/remote will be visible on. An overview of creating Associated Connections for callers/remotes is covered on page 35.

Setting the Visibility List

Click on a name in the *Source Selector:* list. That Channel number, Switched Meter, Control Room, or Studio is highlighted, and the current visibility settings are shown in the *Visibility Pane*. The *Visibility Pane* title shows the highlighted selector’s name (Figure 3-13).

When the Setup app is first used Channel 1 is selected and the system devices are all collapsed with a check box indicating what sources are selected on each device. An empty box means no signals are selected; a filled box means at least one signal is selected; and a checked box means every signal on that device is selected. On a new console every device will have a checked box making every signal visible on every selector.

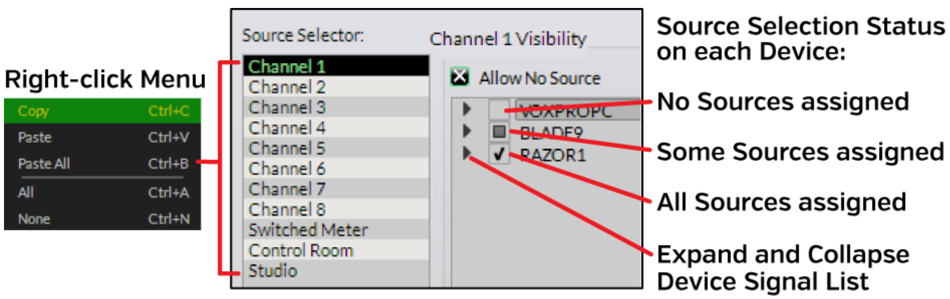


Figure 3-13 Visibilities Controls

To expand a device, click its right-facing arrow. The arrow then points down and sub-categories (Blade, Surface, etc.) are shown, as in Figure 3-12 on page 29. Sub-categories are expanded in like manner until you reach the User and Alias sub-categories. A device, or one of the sub-categories, are collapsed by clicking their down-facing arrow, which then points right again. The current view is maintained until the Setup app is closed. Reopening the app will again show all devices as collapsed.

Visibility Pane

Figure 3-12 (page 29) shows the expanded sources for a DMX console (Mix Engine and Surface) and a Razor Interface. Two other devices, a console, and a PC with AoIP driver, are shown collapsed. The Program busses and other internal bus signals appear under the Surface sub-category below the Mix Engine. These signals are all normally **unchecked** on channel selectors, but most may be left visible on the monitor and meter selectors.

When the User and Alias sub-categories are expanded individual signals can be checked to set that signal as visible on that selector. Unchecked signals will not be listed as an alternate source on that selector.

The *Allow No Source* selection is checked by default so that **NoSource** is available to silence a channel. Typically *Allow No Source* is unchecked on the monitor and meter selectors since one typically doesn't want to silence the monitors or the switched meter.

The *Source Selector:* channel, switched meter, and monitor selectors each have a right-click context menu (Figure 3-13 on page 30) that allows the visibility setting from one selector to be copied and pasted into another selector. After setting a selector's visibility list, right-click on its highlighted selector name and select *Copy* then right-click on another selector and select *Paste* to update that selector's settings.

Note: There's a *Paste All* selection in the context menu. Selecting this pastes the copied list of sources to every channel, meter, and monitor selector.

Once visibilities are set, whether for one channel or for all channels, click *Apply* to transmit the changes to the Surface. The source selectors will immediately update to use the new visibility lists.

Global Alpha Sort

This option is typically checked so visible source names are listed in alphanumeric order, regardless of which device the source is physically on. As each source name is highlighted, the name of the device it's on is shown above the white line in the display. This allows sources with the same name, like PGM 1, but which are on different devices to be identified without having to assign every signal a unique name. Identical source names, even if on the same device, will still appear as separate entries in the list, but we recommend trying to use unique names—if only to make it easier for board operators to select the correct source.

When Global Alpha Sort is unchecked, sources are sorted and listed alphanumerically by each device. The devices are listed in Blade ID number order. Thus Blade 1's sources will be listed in alphanumeric-order first, then Blade 2's source names would be listed, then Blade 3's names, and so on.

NAVIGATOR

Navigator is initially used to edit the names and configure the input and output signals on the Mix Engines, Razors, and Blades networked together to form a WheatNet-IP system. This can be done without obtaining a Navigator license, but to access advanced features, like updating Mix Engine, Razor, and Blade software, you must have a Navigator license.

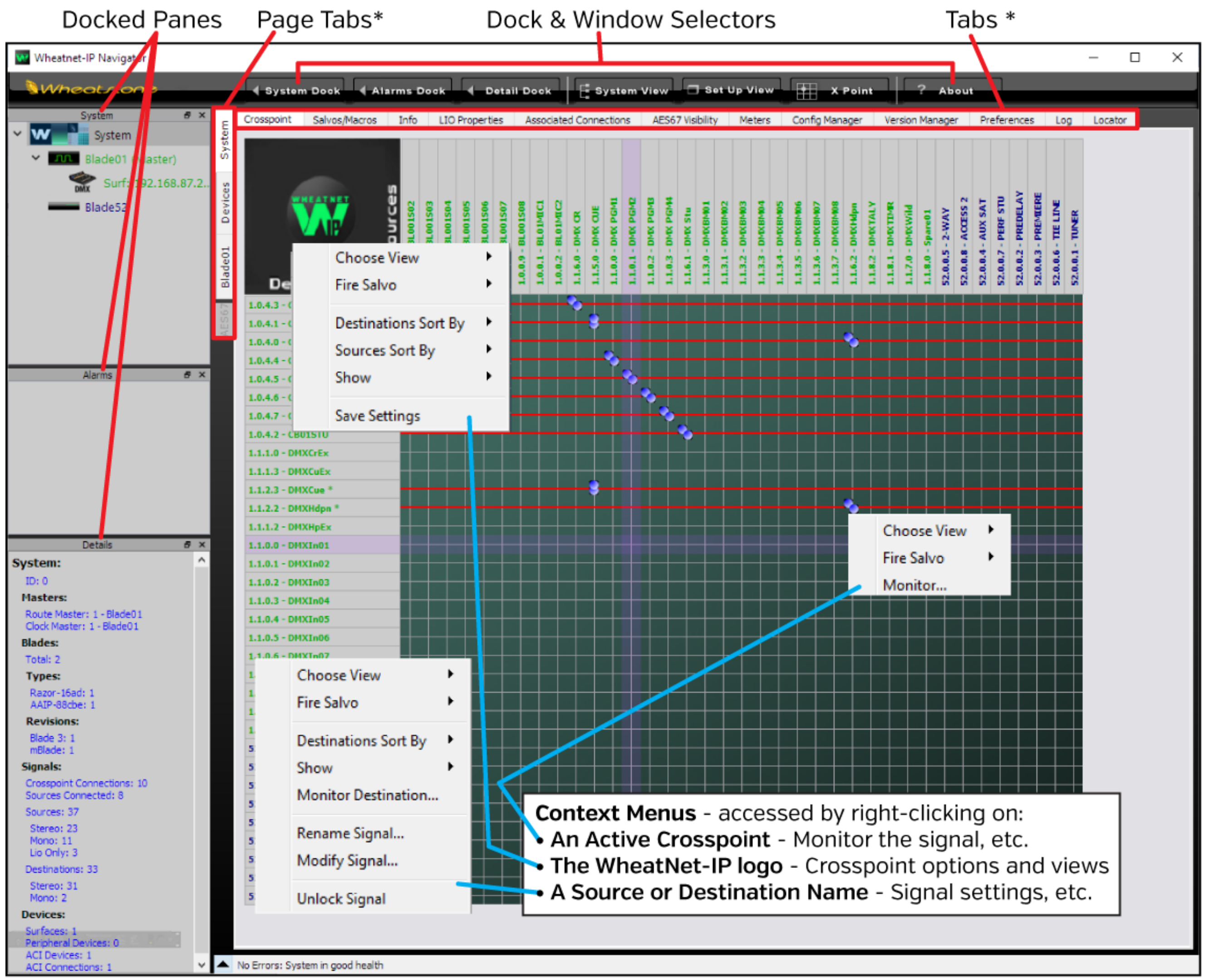
Note: A 30-day Navigator demo license is included with your DMX. Email techsupport@wheatstone.com the Seed ID, obtained from Navigator, to receive your demo license. Contact your Audioarts dealer or Wheatstone sales about purchasing a Navigator license.

Once your audio and logic signals are configured, Navigator will mainly be used to connect sources to destinations using the Crosspoint and Salvos/Macros tab. Here are some of Navigator's other main functions and features:

- Setting the mode for the physical inputs and outputs (mono, stereo, and 5.1 surround are supported)
- Editing the names of system sources and destinations
- Viewing and updating the Mix Engine, Razor, and Blade software
- Manually configuring system devices
- Setting up weighting so specific Blades will be set as the Route Master or Clock Master in a multiple-Blade system
- Assigning system-wide settings like the sample rate and the date/time that's shown on the DMX Surface monitor displays
- Assigning functions to the LIO (the RJ45 LOGIC jacks) and to SLIO (Software Logic I/O) available on most WNIP devices
- Creating Associated Connections to automatically make signal connections in response to different trigger connections
- Viewing the devices connected in the system
- Monitoring (signal levels and audible) of just about any crosspoint in the system
- Logging system activity

To start Navigator, double-click its desktop icon or, from the Start Menu, select WheatNet IP Navigator from the Wheatstone folder.

WheatNet IP Navigator, commonly called Navigator, opens on its **Crosspoint Tab** (Figure 3-14).



* The active System Dock selection [System or a WNIP device] controls the **Tabs** and **Page Tabs**.

Figure 3-14 Navigator Controls Overview

A Login popup box is also shown. When first using Navigator click *OK* to close the Login popup since there is no password set by default. We do recommend setting a Navigator password (click *Set Password...* on the popup window)—when Navigator is installed on a studio PC since unexpected connection changes and other undesired operation could occur when the app is used by untrained personnel.

System Tabs

The **System > Crosspoint tab** is the most often used tab which is why it's shown when the app starts. Clicking on a different tab; on a different Page Tab; or on a Blade, a Surface, or other device icon listed in the System Dock, switches what's shown in the main window.

To redisplay the Crosspoint Grid, click the System icon at the top of the System Dock, or the *System page tab*, then click the *Crosspoint tab*. A floating Crosspoint Grid window can also be opened by clicking the *X Point* window selector.

The Crosspoint grid shows the system sources on the horizontal axis and system destinations on the vertical axis. The colors of the signal names match the colors of the device icons in the System Dock. The color assigned to each device can be edited using the *Blade Preferences* section of the **System > Preferences tab**.

The icons shown in the XY grid indicate the connected signals. A small green square on a crosspoint indicates a logic-only or LIO connection. Audio connections are shown by small round "LEDs." The LED colors indicate whether audio is present with dark blue indicating no or very-low audio; green indicates normal audio levels, while red indicates a very hot level. These default colors can be edited in the Preferences tab.

A single LED on a crosspoint indicates a mono source is connected to a mono destination. Two LEDs at a 45° angle indicate a stereo source connected to a stereo destination. Two LEDs stacked vertically indicate a mono source connected to a stereo destination and two horizontal LEDs indicate a stereo source is being summed to feed a mono destination. A large blue multicolor LED indicates a 5.1 connection, rarely used in radio but often used in TV. The LED can show various colors to indicate the levels on the six channels in a 5.1 surround signal.

If clicking on a crosspoint yields a large gold LED, that means the destination could not subscribe to the source which typically indicates a system fault. On a DMX this can also occur when the built-in Ethernet switch is networked with other devices, but the built-in switch is not configured for IGMP messages. On a stand-alone DMX it could indicate that Mix Engine's switch is not configured correctly. In either case, Appendix A (page 61) covers how to use the Netgear switch configuration utility to edit the Mix Engine's switch configuration to either turn on or turn off IGMP support.

Connecting Signals

As the mouse pointer moves over the grid, purple XY crosshairs (shown in Figure 3-14 on the previous page) point to a source and a destination to identify which will be connected if you click on that grid crosspoint. Clicking on a crosspoint adds an icon to the grid to indicate that connection is now active. To disconnect a signal, click on an active crosspoint to silence the destination on an audio connection, or to turn off the logic on a logic crosspoint.

Note: To prevent "accidental crosspoint" changes use the *General* section of the **System > Preferences tab** to check the option *Use Ctrl/click to make/break a crosspoint connection*. When checked, the **Ctrl key** must be pressed to then click to make or unmake a crosspoint.

A destination with a horizontal red line extending across the grid indicates a destination that's locked. Locked signals may be manually set, system-assigned (like DMX PGM1 being connected to the PGM 1 output), or auto-set to appear while a fader channel is On. Regardless, any locked signal can be unlocked by right-clicking on the Destination name and selecting *Unlock Signal* from the context menu (shown in Figure 3-14 on the previous page).

Note: Use caution when unlocking DMX channel destinations DMXIn01 – DMXIn16 since those red lines indicate a fader channel is currently On. This would also be true for dedicated outputs feeding air or going to an internet streamer since changing those destinations' sources would affect your air or stream feed.

Remote Monitoring of Audio Signals

Right-clicking on an active crosspoint connection, on a source name, or on a destination name brings up different context menus, as listed in Figure 3-14 on the previous page. Then selecting *Monitor…*, *Monitor Source…*, or *Monitor Destination…* opens a Monitor window (Figure 3-15) showing the source levels and, if that signal is connected to a Destination, the Destination levels.

The bottom section of the popup window has the controls to listen to the audio. A drop-down list sets which PC audio destination to listen to the signal (typically the PC's built-in speakers). Click *Listen to Source…* to hear the audio on your PC. Use the *Volume* control to adjust your listening level.

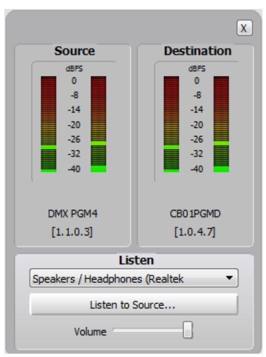


Figure 3-15 Signal Monitor Pop-Up

Note: This monitoring feature is very handy when you remote into the system from home or the road since the audio should then play back on the remote PC to confirm the correct audio is present. For remote access to your admin PC, we recommend TeamViewer.

Salvos/Macros Tab

Clicking the **System > Salvos/Macros tab** shows a display that looks like an unpopulated Crosspoint grid. The difference is that this grid is not "live." In the Salvo grid (Figure 3-16) the signal connections needed for a future show, a new daypart, to setup connections on a Blade, or to setup for a special event, are saved as a Salvo. That Salvo can later be "fired" to make the various crosspoints saved in that Salvo.

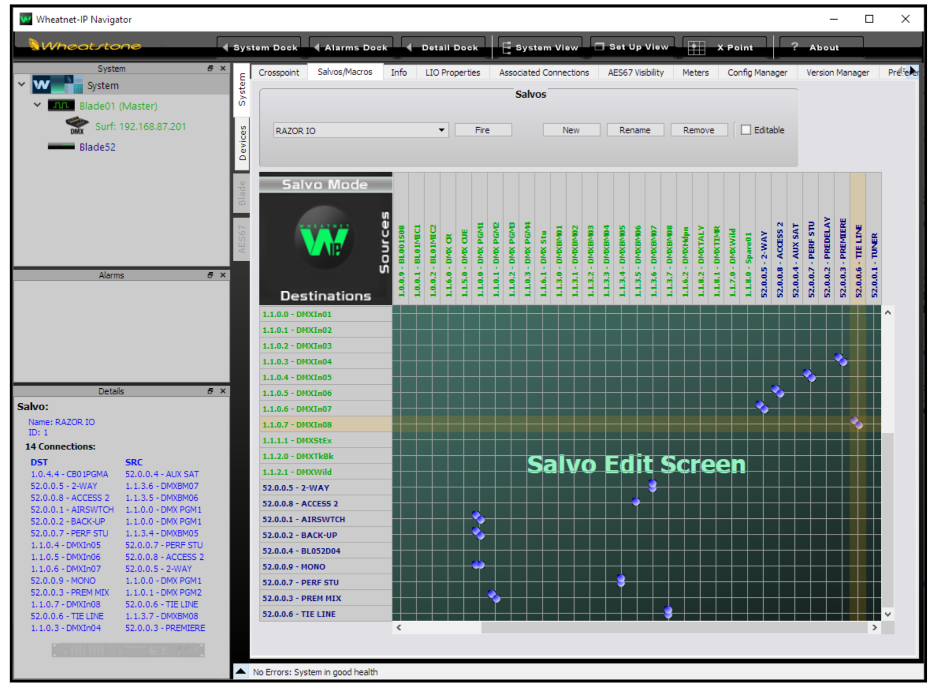


Figure 3-16 Salvo/Macro Tab

Dozens of audio and logic crosspoints can be set and saved in dozens of Salvos. Every Salvo begins life as a number (Salvo 1, Salvo 2, and so on), which doesn't mean much, so you'll want to rename any Salvos you create. In Figure 3-16, Salvo 1 was renamed as RAZOR IO by selecting that Salvo in the drop-down list then clicking *Rename*. Be sure to use a name that easily identifies the Salvo. Even though you can enter a Salvo name using an almost unlimited number of characters, only about 25 characters are shown in the Salvo drop-down list. However, when assigning a Salvo to a programmable button in the LIO tab, only the first eight characters will be shown, so it's best to limit your names to eight characters, like: RAZOR IO, DanceRMT, SuperPre, or GrdnShow, or at least make sure the first eight characters will allow one to identify the Salvo which can offer more details in the rest of a longer name.

Click *New* to create a new Salvo with the name *Salvo X (empty),* with X being the next unused Salvo number. Any existing connections from a previous Salvo are removed so you have a clean crosspoint grid. As soon as you make the first connection "(empty)" is removed from the name indicating that Salvo is in-use. There's no Save Salvo button since the active Salvo gets updated with each edit.

When you select a Salvo using the drop-down list, its connections are shown in the Salvo Edit Screen in read-only mode. The **Details Dock** also lists that Salvo's connections by their destination and source signal numbers and names. If you want to make changes, click the *Editable* button, then you can add or remove crosspoints.

To take a Salvo, select it from the drop-down list then click *Fire*. The crosspoints assigned in the Salvo are all immediately connected—except for any connection going to a locked destination, like channels which are currently turned on. To make those connections, turn off the channel(s) or unlock the locked destinations, then fire the Salvo again. You can also assign the Soft button, in the monitor section of the DMX Surface, a button on a programmable button panel, or a logic input from a hardware button to fire a salvo.

Info Tab

This tab has three sections, highlighted in Figure 3-17, which may need to be checked and/or set. In the **Set Date and Time** section check *Use this PC's time* then click *Apply* to update the date and time on the monitor OLED displays to that PC's date and time. If the PC running Navigator is also connected to the Internet (using a separate NIC), click the *NTP* radio button then select a remote time server to keep the system time synchronized.

Note: If the Mix Engine is power cycled, or if it or the Surface is rebooted, the date and time must be manually updated.

In the **Clock Master Info** section, check that the *Clock Master Frequency* is properly set. The default setting is 44.1, but this can be changed to 48 kHz (needed by some audio playback systems) by clicking the 48 kHz radio button.

If there are any Blades or Razor Interfaces in the system, the **Blade Display Settings** section is used to control the brightness of the front panel OLED displays. There is no *Screen Saver* on a Razor so that setting is not applicable to a Razor, but the *Brightness*, *Dim Brightness*, and *Screen Dim* are set like setting up the OLED displays on the DMX Surface Setup app.

The remaining sections on this tab are not applicable to a stand-alone DMX. Refer to the WheatNet-IP System User Manual about the other settings on this tab.

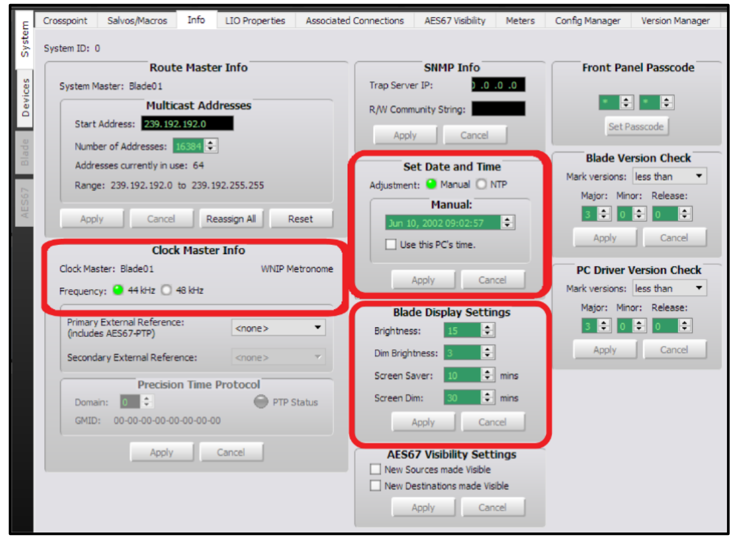


Figure 3-17 Info Tab

LIO Properties Tab

This tab (Figure 3-18) can be used to set the unconnected state condition for the User-defined logic functions (User 1–User 500). Each User-defined logic function can be set so its unconnected state is High, Low, or leave it in its Last State when disconnected. The default setting is *Low*. This tab also shows the unconnected state settings for the pre-assigned logic functions, but none of those settings can be edited.

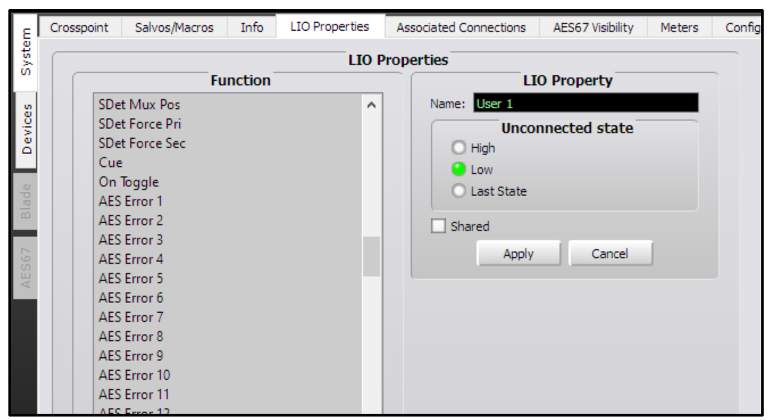


Figure 3-18 LIO Properties Tab

Associated Connections Tab

This tab is used to create, view, and edit Associated Connections. An Associated Connection can be used to automate common system connections by monitoring for a **Trigger Connection** to occur. When this occurs one or more **Associated Connections** are then made.

A typical use for an Associated Connection (AC) is to automate connecting bus-minus signals to callers/remotes by creating an AC for each channel that the caller or remote is visible on. The Trigger Connection occurs when the caller/remote is taken on a “trigger channel.” The Associated Connection then connects that channel’s bus-minus signal to that caller/remote.

This type of AC is shown in Figure 3-19, but one can automate many other system connections in response to a predefined trigger connection. One can also setup a trigger on a Disconnect like removing a crosspoint or taking NoSource on a fader channel. Additional details on Associated Connection applications can be found on page 43.

Setting up an AC is a four-step process, as shown in Figure 3-19. In the tab’s **Triggered Connections** section, click the *Add…* button (1) to open the *Add Triggered Connection* dialog box (2) where a source and destination are selected. In the example, the triggered connection is the PHONE being taken on channel 8 (DMXIn08). Clicking *OK* adds that entry to the Triggered Connections list and closes the dialog box.

Click on the Triggered Connection to highlight its row then click the *Add…* button in the **Associated Connections** section (3) to open the *Add Associated Connection* dialog box (4) to select the source and destination you want connected when the trigger connection is detected. In most cases you’ll want to checkmark the *Lock Override* option. When it’s checked, and the trigger connection is detected, the selected source is connected to the destination regardless of whether that destination is locked. When *Lock Override* is not checked, the Associated Connection will not be made on a locked destination.

As shown in Figure 3-19, multiple Associated Connections can be assigned to the same trigger. In the example, when the PHONE is taken on channel 8, the channel 8 bus-minus signal is connected to the PHONE plus the caller and host mic audio are connected to a VoxPro editor (the host mic, BL01MIC1, connects to the left channel and the incoming PHONE audio connects to the right channel of the VoxPro output).

Once you setup the Associated Connections, click the all-important *Apply* button, at the bottom of the tab, to tell the system to start monitoring for that Trigger Connection.

If the PHONE is set visible on multiple channels, you’ll need to repeat this process for all of the other channels the PHONE is visible on to ensure the correct bus-minus audio is returned to that caller, regardless of which channel the PHONE is taken on.

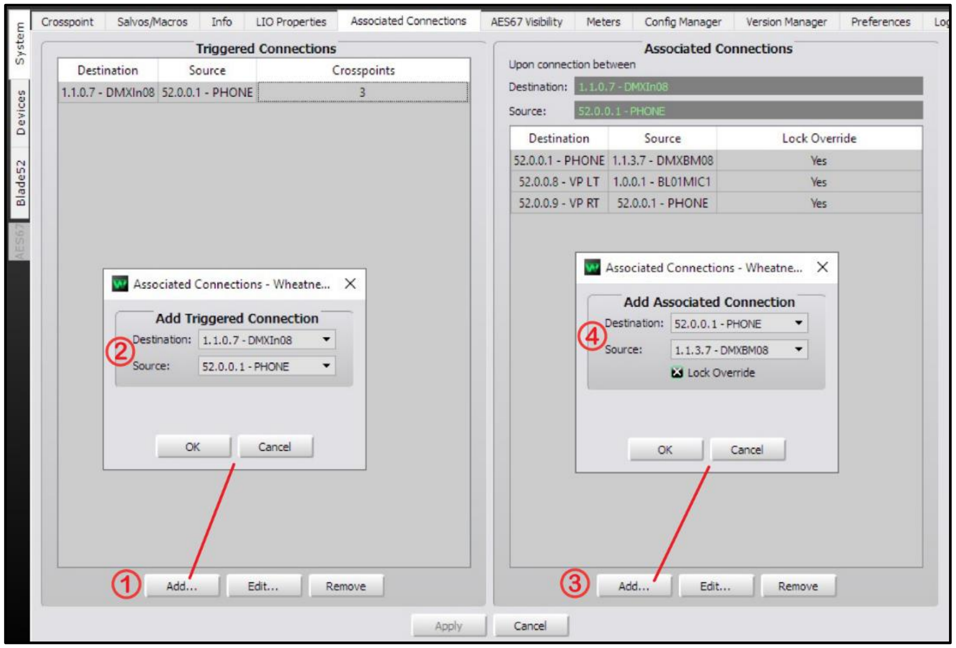


Figure 3-19 Associated Connections Tab

AES67 Visibility

This tab (Figure 3-20 on the next page) sets which WNIP system signals will be transmitted to AES67 devices and which WNIP destinations can receive an AES67 device stream. This tab is not used unless Navigator has an AES67 License; there is at least one Blade-3 Blade in the system that can be designated as the “AES67 Translator;” and there are AES67-compatible devices networked with your WNIP network.

Sources and Destinations are selected like the Visibilities tab in the DMX Surface Setup app (see pages 29 and 30 for signal selection details). Setting a source as visible, and clicking *Apply*, means the selected audio is connected to the “AES67 translation Blade” in preparation for being streamed out to an AES67 device. This means that checking a lot of signals could cause audio streaming traffic issues and even when the network is properly configured, one could run out of bandwidth. The bottom line: select only those signals which need to stream audio to an AES67 device.

The same holds true for Destinations that receive streamed audio from the AES67 translation Blade. For more details about networking AES67 devices with a WNIP system see the Blade-3 User Manual or the Commissioning AES67 White Paper (both available for downloading from the main Wheatstone web page under Support & Downloads.

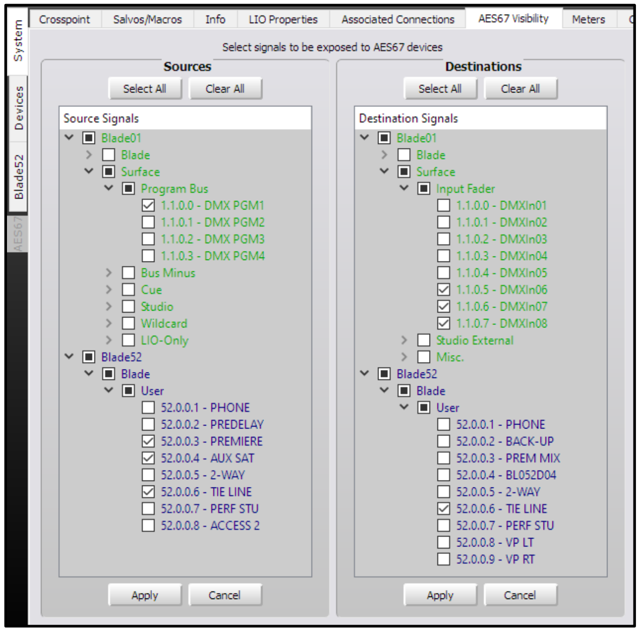


Figure 3-20 AES67 Visibility Tab

Meters Tab

This tab (Figure 3-21) can have up to 64 source and destination meters, selected to show critical signal levels across a system, in one view. The only signals which can't be metered are Mix Engine destinations like console fader channels and external monitor inputs, and AES67 signals. The meters are shown in rows of eight meters. Mono and stereo signals each take up one meter space. A 5.1 signal meter takes up three meter spaces.

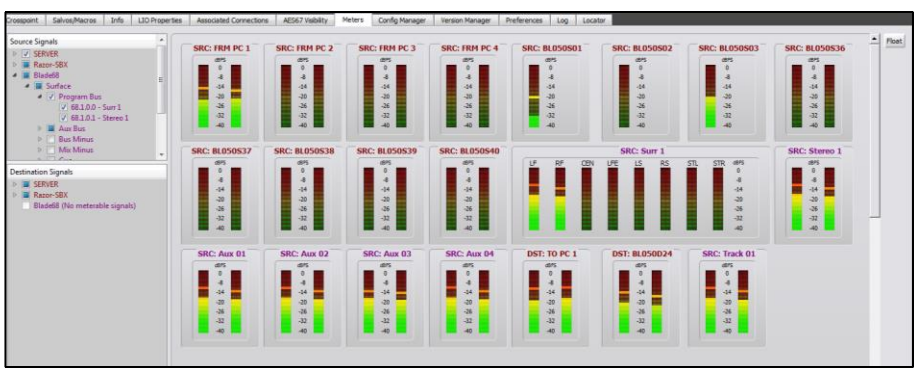


Figure 3-21 Meters Tab

A meter is created when a signal is checked in the Meters Sources and Destinations lists, so it's best to map out how to group the meters prior to selecting them. To remove a meter, uncheck the signal in the Meter Source and Destination Signals lists. Since the remaining meters are not rearranged, you'd need to checkmark another signal to fill in any gaps opened by deleting meters.

This tab is especially useful when accessing the system remotely since one can see signal levels from multiple devices immediately, as opposed to selecting a Blade and then viewing the Sources or Destination tab to view that Blade's signal levels. The Meters tab can also be set as a Floating Window by clicking the Float button in the upper right corner. To re-dock the tab, click the Unfloat button.

Config Manager Tab

This tab (Figure 3-22) is used to backup system configuration settings as well as save or restore the crosspoints. Even though every Blade keeps the configuration settings for all other Blades, it's important to periodically save a system backup—especially after you finish setting up a new system so that your configuration settings are saved. This can allow a system device to be manually restored if you have to factory reset a device or replace a device and the automatic replacement process fails.

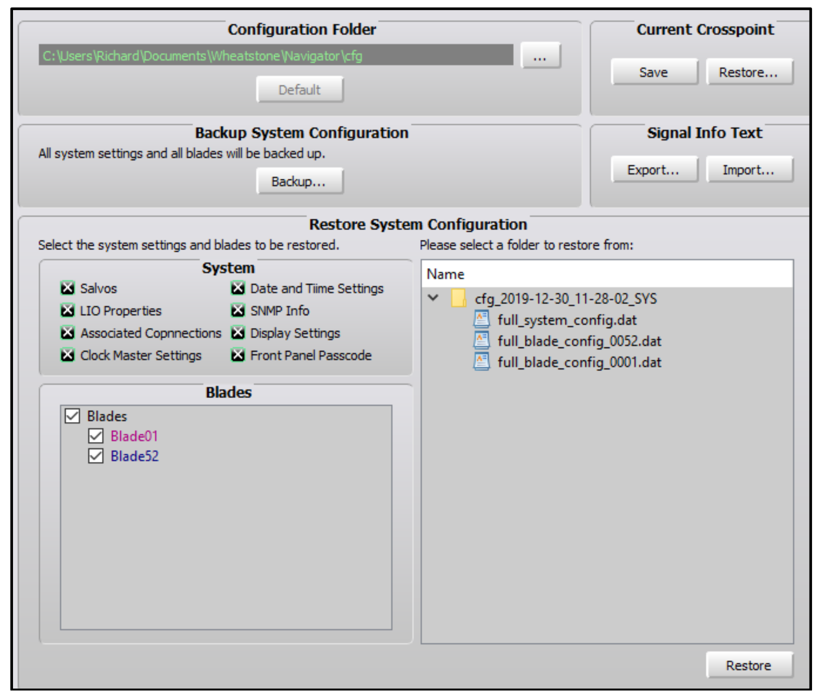


Figure 3-22 Config Manager Tab

The **Current Crosspoint** section of this tab allows the current crosspoints to be saved to a file which can later restore those crosspoints. Note that the *Save* button is not shown unless the option *Enable Advanced Controls* is checked in the System > Preferences tab.

Clicking *Save* saves the current crosspoints to a file, named by the saved date and time, in a Documents\Wheatstone\Navigator\Connections folder. Right after saving a crosspoint file if you click *Restore…* to open a dialog box you can edit the default name to identify the show, daypart, or other function saved by the crosspoints, so you'll be able to easily select the correct file when using the Restore function.

There's one important caveat to using *Restore Crosspoint* versus using a Salvo to take multiple crosspoints: Restore overwrites locked signals—including ones locked because a channel is currently on-air, whereas taking a Salvo will not change the source on a locked destination.

The **Configuration Folder** section sets the folder path for saving your device configuration files. Clicking the … button to open a save dialog box so you can change the path as required. Clicking the *Default* button restores the Configuration folder back to using the default path.

The **Backup System Configuration** section allows all system settings to be backed up by clicking *Backup…*. A pop-up warning box appears listing the filename, which includes the date and time. Clicking *Yes* saves a separate Config file for each blade within a System Backup folder. The saved configuration folders are listed in the **Restore System Configuration** section of the tab. Each system folder can be expanded to view the individual system device files, which are named by device number.

The **Restore System Configuration** section is used to restore a saved configuration. The check boxes, in the **Restore** section, allow you to select which system devices and settings will be restored when the *Restore* button is clicked. Click on the name of a saved configuration folder and then click *Restore* to restore the configuration settings held in that saved configuration folder. Once the settings are restored, any Blade that's being restored must be rebooted to use the restored settings.

Version Manager Tab

This tab (Figure 3-23) is used to update the operating system code on Mix Engines, Razors, and Blades, either individually or all together. When Navigator was installed, the current release of the WNIP operating system software was copied to the PC, but an updated software archive file may also be available separately from Navigator.

Note: Navigator must be licensed to update the operating system software on your Blades.

In the **Blades available for update** section, checkmark which Blades to update. Blades can be updated one at a time or all at once. Click *Select All* to checkmark all Blades. Click *Clear All* to uncheck all Blades. Once the Blades you want to update are checked, click *Update…* which opens a dialog box to select the binary code file holding the updated code files. In most cases there will only be one file in the directory which has the version of that code in its name. Select the file and click *Open* to load that file. A warning dialog box pops up asking the age-old question: *Are you sure?* Click *Yes* to load the new code or *No* to cancel the update.

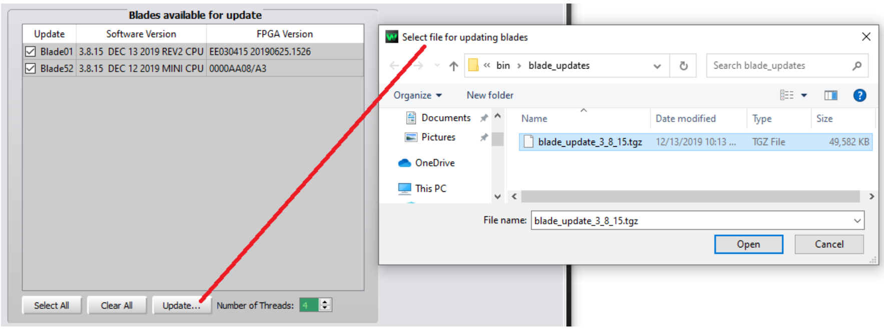


Figure 3-23 Version Manager Tab

Note: While the software is being uploaded there is no impact on system operation. Since you must reboot the Blade to then run the updated software, that will cause momentary audio interruptions on each Blade as it completes rebooting and remakes all of its existing connections.

Preferences Tab

This tab (Figure 3-24 on the next page) sets Navigator's appearance and some of its functionality. The **General** section **Default View** sets whether the default crosspoint view (*No View*) or a custom view is shown when Navigator opens. **Default Grid Labels** sets how the source and destination names are listed as name only, names and location, name and signal format, or signal ID and name. The **Default Sort Order** section sets the order in which the Source and Destination names are displayed and how the Blades they are on are listed in the grid.

The five check boxes at the bottom of the **General** section set how Source and Destination signal ID numbers are shown (**Use dotted Signal IDs rather than hexadecimal**), the overall look of the Navigator app (**Use Style Sheet**), whether 12-hour or 24-hour is used in Navigator (**Use 24 Hour Time**), how a crosspoint connection is made or broken (**Use CTRL+click to make/break a crosspoint connection**), and whether to show advanced controls (**Enable Advanced Controls**).

The **Blade Preferences** section allows each Blade's color to be edited by double-clicking on a Blade Name. A *Select Color* window opens so a new color can be assigned to that Blade. The color setting not only affects the color of the Blade names in the dock it also affects signal name colors in the crosspoint and salvo tabs, and the color of log entries from that

Blade. Since System Announcer log messages are black, we don't recommend assigning that color to a Blade.

The **Crosspoint Preferences** section allows one to customize how the Crosspoint grid looks. You can change the color of the normal pointer crosshairs (**Grid Hilite**), change the color of the crosshairs when over a crosspoint with Associated Connections (**Assoc Hilite**), change the color of the audio connection LEDs from their default purple, green, and red, change the color of the logic connections from the default green, and change the grid background and/or add your own logo behind the grid.

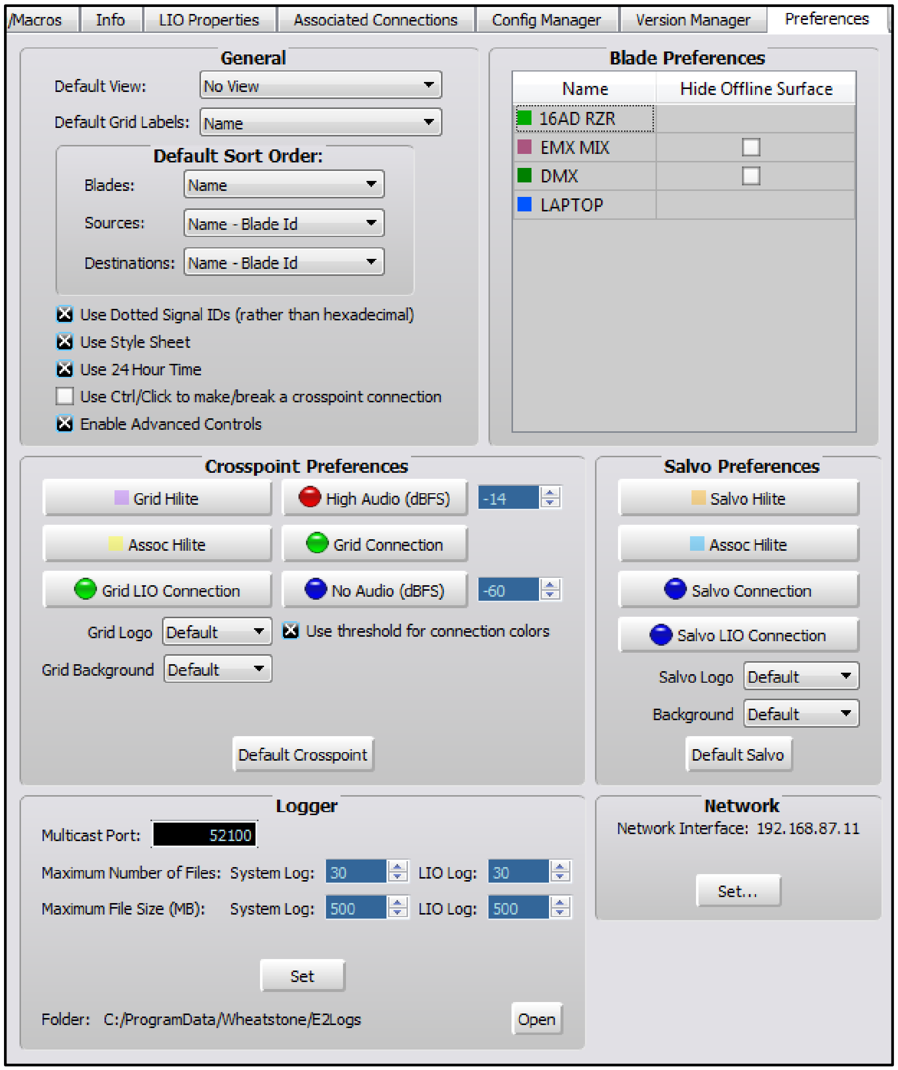


Figure 3-24 Preferences Tab with Advanced Controls Enabled

The *Default Crosspoint* button allows you to return the settings to their defaults if you find that "customizing" the crosspoint tab didn't work out the way you thought it would. The Salvo crosspoint display can likewise be customized as well, knowing the *Default Salvo* button can be pressed to return it back to its default settings.

The **Logger** section is only shown when *Enable Advanced Controls* is checked in the **General** section. These settings affect the System Log operations and the **Log** tab. We recommend using the default settings, but if a change is made, clicking *Set* applies the updated settings. Click *Open* to open the E2Logs folder which is where the various types of log files (text files) can be viewed.

The **Network** section shows which NIC Navigator expects is connected to the WNIP network. If no devices appear in the System Dock, look at the NIC address shown here. If it's not correct, click *Set…* to open a dialog box listing that PC's NICs so the correct NIC, which must have an address in your WNIP subnet (default subnet is 192.168.87.0) can be selected.

Log Tab

The Log tab (Figure 3-25) shows system messages, sent by the WNIP devices, by date and time. Its main purpose is as a troubleshooting tool, although you can get a good sense of network communications by observing the log as the system is actively being used.

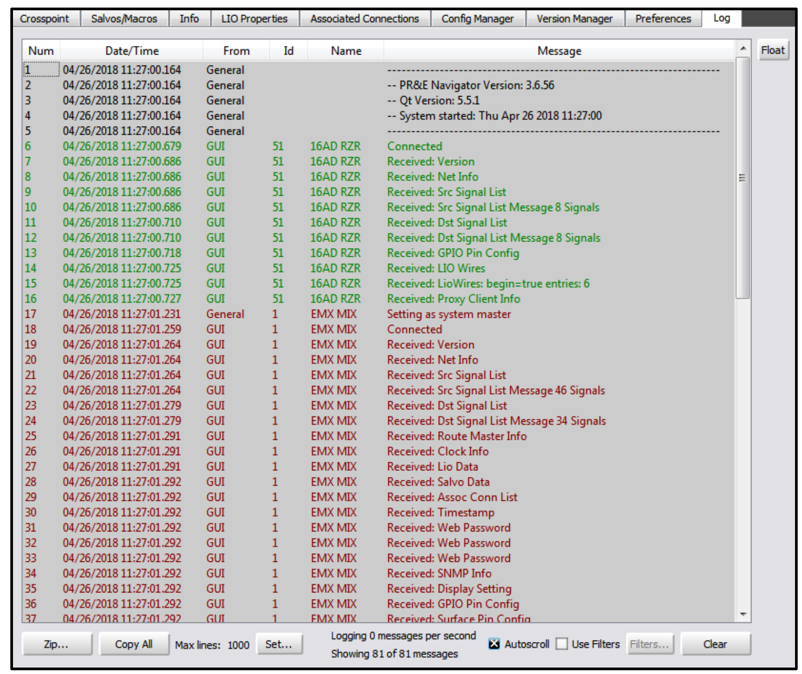


Figure 3-25 Log Tab

The log information shown in Navigator is generated by the WheatNet IP Logger service which runs on any PC that has Navigator installed. Although Navigator doesn't have to be running to save the log, a PC with Navigator installed on it would need to be connected to the WNIP network, and would need to be operating, for system logs to be generated and saved to C:\ProgramData\Wheatstone\E2Logs.

Locator Tab

This tab (Figure 3-26) shows all Blades, Surfaces, and other devices connected to your WNIP network. Click *Refresh* to rescan the system and refresh the view. Note that a similar view is obtained by clicking on *System View* but in that view the Surfaces are shown under a separate tab from the Blades.

Just as with the DMX Surface Setup Locator screen, clicking on a device lists its properties in the **Set Selected Device Properties** section where the settings can be viewed and manually edited, if required. Typically editing these settings will require that device to be rebooted before the new settings are used.

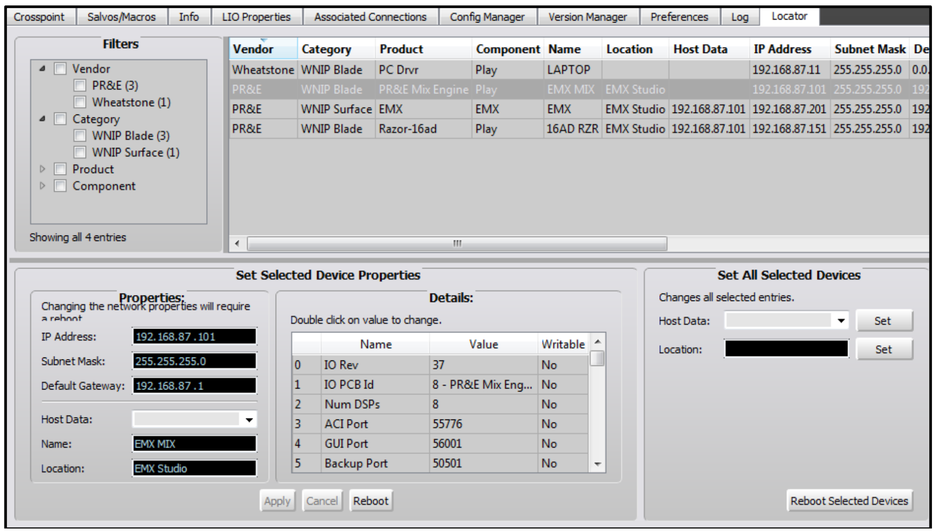


Figure 3-26 Locator Tab

Blade Tabs

Selecting a Mix Engine, Razor I/O Interface, Blade, or PC Blade (a PC/server running the WNIP driver) by clicking on its icon in the System Dock, changes the list of tabs from those shown when System is selected. The six **Blade tabs** for a Razor are shown in Figure 3-27.

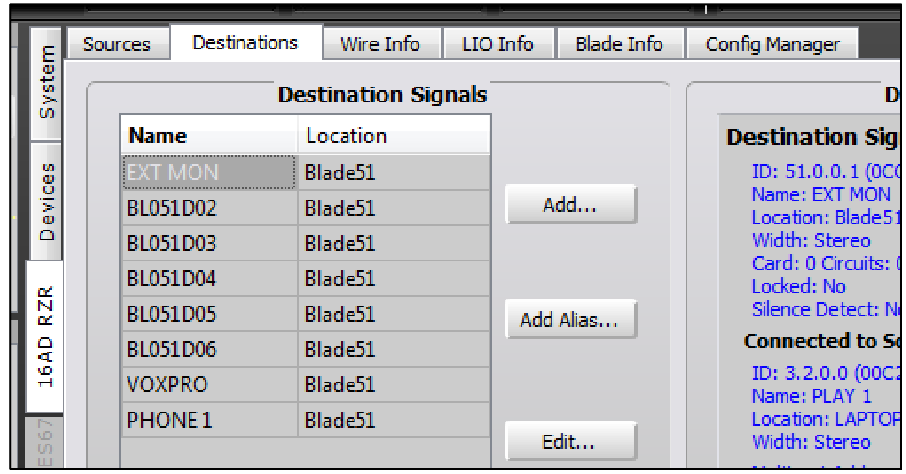


Figure 3-27 Blade Tabs

The Blade tabs can be used to update the settings for that Blade's input signals on the **Sources tab**, its output signals on the **Destinations tab**, to save and/or print out a list of input and output connections on the **Wire Info tab**, view and test the device's logic connections on the **LIO Info tab**, view and edit the Blade network settings on the **Blade Info tab**, and save or restore that Blade's settings on the **Config Manager tab**. The M4IP-USB and other WNIP Blades will have additional tabs like **Silence Sense** and **Utility Mixer**.

Sources & Destinations Tabs

These two tabs have identical sections. These tabs are used to edit the Source or Destination names, delete and add new signals, edit a signal's format, add logic commands to associate them with a particular audio signal, add and edit logic–only signals, create signal Aliases, and edit the Blade's location name.

The complete Sources tab is shown in Figure 3-28. The lower half of the Sources and Destinations tabs is the **Meters** section which shows signal levels for the physical inputs (on the Sources tab) or outputs (on the Destinations tab) on that Blade. Just below the meters are level controls and, for stereo signals, a balance control. The level controls allow each signal level to be adjusted by up to +/-18 dB from nominal. Details on using the level and balance control can be found on page 41 in the *Level & Balance Controls* section.

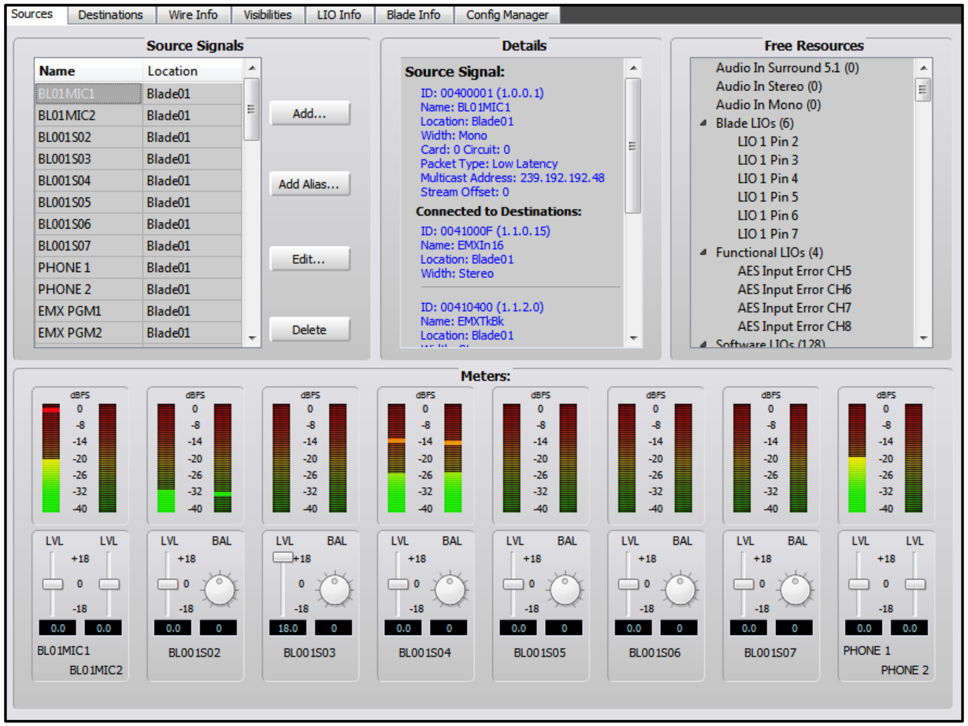


Figure 3-28 Sources Tab

Naming Signal Inputs and Outputs

Every Blade input and output signal has a default name that can be edited, using the **Blade > Sources** or **Blade > Destinations** tab, to identify the peripheral device or the signal connected to each input and output. The names for the internal signals, like the PGM buses, console fader channels, the bus-minus signals, etc. are typically left using their default names.

As an example of how to edit names, the Mix Engine's Analog In 1 signals have the default names of BL01MIC1 and BL01MIC2 since those two inputs are set as mono signals. Analog In 1 is designed to jumper to MICS OUT. These two signal names might be edited to MIC 1 and MIC 2 or to HOST and GUEST—any name using up to eight characters. Input signals are edited using the **Sources tab** (Figure 3-28). To see this tab, click on the Mix Engine icon (Blade01) in the **System Dock** then click the **Sources tab**.

To edit the name, click on the name in the **Source Signals** section of the tab then type in a new name of up to eight characters. You can alternately click once on the name to highlight it then click *Edit…* to open the **Signal Edit** window (shown in Figure 3-29) and edit the name in that window, as shown by the red outlines.

Changing the Signal Type

If you aren't planning on using the two built-in mic preamps on a Mix Engine, or on an Analog-only Razor, two other mono sources, like a pair of phone hybrids, or a phone hybrid and a codec, could alternately connect to Analog In 1. If no mono signals need to be connected the two mono Analog In 1 signals can be changed into a single stereo input by clicking on BL01MIC2 in the *Source Signals* list to highlight it, then clicking **Delete** to remove that signal—after clicking *Yes* on the "Are You Sure?" pop-up box.

Click on the other mono signal (BL01MIC1) to highlight it and click **Edit...** to open the edit window (Figure 3-29). Change the **Signal Type** selection from Mono to Stereo. You can then edit the name back to its generic name: BL001SO1 or, if you know what stereo device will connect to that input, enter the device's name in the *Name* entry box. Click *Finish* to close the Edit Window.

Note: You cannot edit a source or destination signal while it's connected in the Crosspoint grid. When the **Signal Edit** window opens (Figure 3-29) look at the bottom of the window to verify it shows "Signal is not connected." If it shows "Signal is connected" you won't be able to make any changes to that signal other than editing its name.

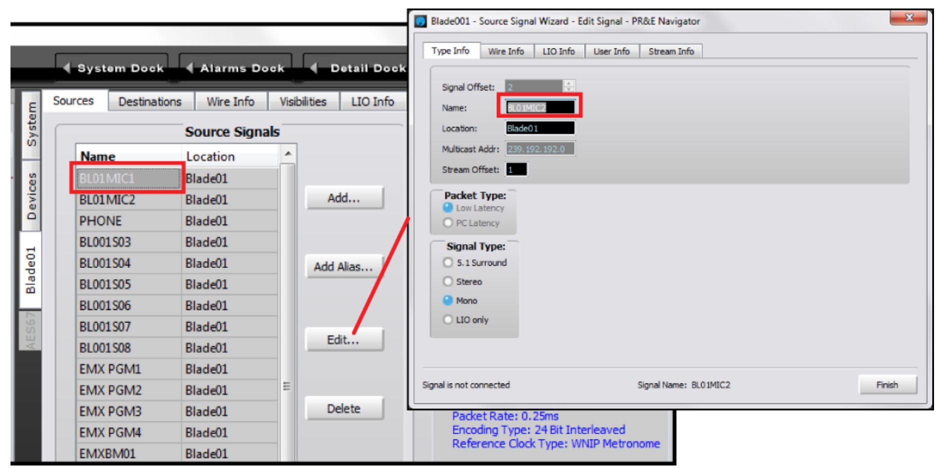


Figure 3-29 Editing a Signal

Creating Aliases

A source or destination may need to have a different name displayed, depending upon where or how the signal is being used, which might differ from the one assigned in Navigator. To accomplish this, an *Alias* can be created for any signal. An Alias is often used with mics to allow them to be named as: MIC 1, MIC 2, MIC 3, and so on in Navigator, but each mic can have multiple Aliases created, named by the "Talent names" so the channel displays show the talent names rather than the generic names, like MIC 1, MIC 2, and so on, which you set in Navigator.

To create an Alias, click on a Source name (you can also set up an Alias on a Destination or on a logic signal, but most often an Alias is created for a Source) then click *Add Alias…*. As an example, to create an Alias for BL01MIC1, click on that name (highlighted in Figure 3-29 on the last page) then click *Add Alias…* to open the *Add Alias* window. Type in the "Talent's name" in the *Name:* entry box. Click *Finish* to close the *Add Alias* window. The Alias name is now added as a completely new "source" that can then be added to channel source visibility lists.

When adding alias names and editing source and destination names, use the DMX Surface Setup app to run a new **System Scan** (shown on page 21 in the Setting the Surface Host section). This action updates the Surface Setup app with the changes you make in Navigator, updating the signal names and adding the new Alias names to the Source and Destination lists so you can then add them to the Visibility lists.

Even though an Alias is created using a specific source or destination, each Alias is treated as a brand-new signal. This means that a mic signal Aliases will need the same VDips tab settings (see page 25) assigned to it as its source, like CR, Studio, and Cue mutes, and studio tally.

Level & Balance Controls

The **Meters** section of both the **Sources** and the **Destinations tabs** (Figure 3-30) shows the levels for the stereo and mono inputs, or for the stereo and mono outputs, on the selected Blade (note that the DMX Mix Engine only has stereo outputs).

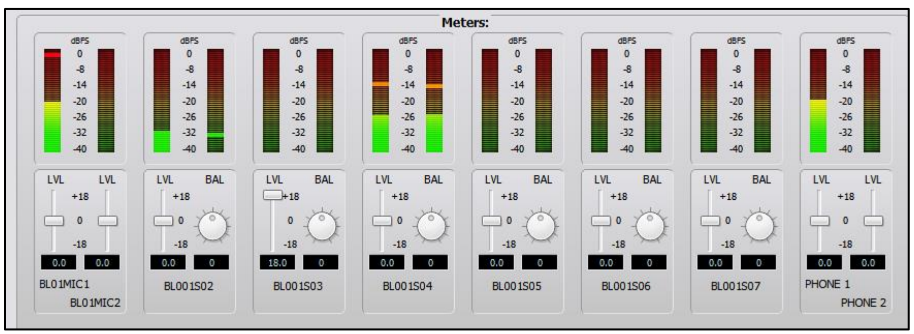


Figure 3-30 Meters: Section of the Sources and Destination tabs

Below each meter are level (**LVL**) controls for mono inputs and outputs. On stereo inputs and outputs there's a level and a round balance (**BAL**) control. These controls affect the signal prior to the meter display so changing the level or balance will be reflected in the meter display when audio is present on that input or output. The level controls affect the audio in real time so use caution on live systems when adjusting the levels.

Levels are adjusted using any of these methods:

- Click/drag the level "knob" or BAL control to adjust level/balance
- Click above/below the level control adds/subtracts .5 dB per click
- Click on the LVL or BAL control then adjust your mouse wheel to fine tune the level or adjust the pan/balance
- Directly enter the amount of gain/trim or balance. Highlight the level or balance entry number, then type in a new number. For levels: -18.0 up to +18.0 are valid. For balance: -100 (full left) to +100 (full right) are valid. Press *Enter* to assign the new entry

Note: To allow for headroom the top of the average bar graph should be near -20 (the peak LED is then near -8). The meters for the Razor I/O Interfaces only show average level bar graphs.

The meters on the DMX Surface have finer resolution than the Navigator meters so use them for the most accurate level adjustment. Connect a test tone, at +4 dBu, from a connected device. If needed, adjust the input control so the top of the bar graph is at -20 using the Navigator meters. Assign that signal to a DMX channel. Align the channel fader with the unity gain arrows (-12 dB). Turn the channel on and assign it to PGM 1 with no other channels assigned to PGM 1. The Program 1 meter's bar graph should light up all the green LEDs and one yellow LED, which indicates -20 dBFS. If that's not the case, adjust the input level or balance control.

Balanced vs. Unbalanced Inputs & Outputs

One common use for the input level controls is to boost the gain of signals coming from unbalanced devices. Many "Prosumer" devices have ¼" jacks on them but most are unbalanced TS (Tip-Sleeve) rather than balanced TRS (Tip-Ring-Sleeve) jacks. Their nominal output level is often the same as that found on consumer unbalanced "RCA" jacks: 300 mV, not the 1.2 V nominal level found on true balanced output devices.

The unbalanced device's hot signal is connected to the + input and, on a TS connector, the – input will end up being connected to ground. This will cause the input signal to be 6 dB lower, since it's now unbalanced, but it also typically has a lower output level to begin with so adjust the input level control to add gain (typically +12 dB to +18 dB) to compensate for the lower level from unbalanced devices. Doing this eliminates the need for a level match box for most unbalanced output devices.

Note: A Blade's balanced output cannot directly connect to an unbalanced device—even though you can lower the output gain to match the unbalanced device input. Blade outputs are active balanced signals, so do not connect the – (minus) signal to ground to "unbalance" a Blade output since this will lead to overheating the balanced line driver IC. You must use either a matchbox or a balun (a line level balanced-to-unbalanced transformer) to connect any balanced analog output to an unbalanced device.

Assigning Logic

Blades have six LIO (Logic Inputs or Outputs) ports on each RJ45 Logic jack (there's one jack on the Mix Engine and Razor Interfaces and two on WheatNet-IP Blades). The LIO ports are used to connect legacy devices using hardware GPIO (General Purpose Input/Output) logic like a Henry Superelay, CD players, hot mic arm LEDs, mic cough panels, etc.

In addition to the six GPIO ports or pins on each RJ45 Logic jack, Blades also have 128 "Software logic" or SLIO signals which are used to carry logic between the DMX and other networked WNIP-compatible devices like Wheatstone talent panels, Eventide BD600W+ delay units, and automation software from RCS, RPB, BE, Wide Orbit, ENCO, and other vendors.

LIO and SLIO logic is set using the same **Edit Signal** window as is used to set the audio format (stereo/mono) and to name the audio signals.

When the logic is associated with an audio input signal, like BL01MIC1, select that audio input name (it highlights) then click *Edit* to open the Edit Signal window (the process shown in Figure 3-29 on page 40). Click the **LIO Info** tab (Figure 3-31) to then view, edit, or delete logic settings associated with that audio signal.

Figure 3-31 also shows the typical settings used with a WNIP Talent Station (TS) in green text. It uses five logic signals to carry Remote On, Remote Off, and Cough commands from the TS to the DMX channel and sends two logic return signals (the On and Off Tally logic) back to the TS. Since TS panels are networked devices, they also use SLIO signals. Clicking *Add* or *Edit* opens the **Assign LIO** window, also shown in Figure 3-31.

Note: WNIP 6-channel GPIO Relay Interfaces are available from NotaBotYet. Each connects to an RJ45 Logic jack using a standard straight-thru CAT5 cable. These interfaces have screw terminals for connecting the wiring for the peripheral devices.

Logic-Only Signals

A logic-only signal can be created as either a source (select the **Blade > Sources tab**) or as a destination (select the **Blade > Destinations tab**). As an example of creating a logic-only signal, here's how to setup a LIO Only destination for a Control Room Warning or Hot Mic Light:

On the **Blade > Destinations tab** click *Add...* to open a new *Add Signal* window. In the **Signal Type:** section, select *LIO only*. A default name is assigned to the new signal, like BL01D09. Edit that name to something useful, like CR WARN, by double-clicking on the name in the **Name:** entry box and entering a new name of up to eight characters. Click the *LIO Info* tab to open the **Logic Assignments** window, shown in Figure 3-31. On a new logic-only signal click *Add...* to open the **Assign an LIO** window, also shown in Figure 3-31. The first open port is on pin 4, which is highlighted in Figure 3-31, which allows that pin to be configured. To set that pin for a Control Room warning command, click Output as the *Direction* then select the *Function* from the drop-down menu. A CR WARN command typically uses the Function: *Studio 1 In-Use*. Clicking *Apply* sets that logic command. Click *Close* then *Finish* to close the signal edit window.

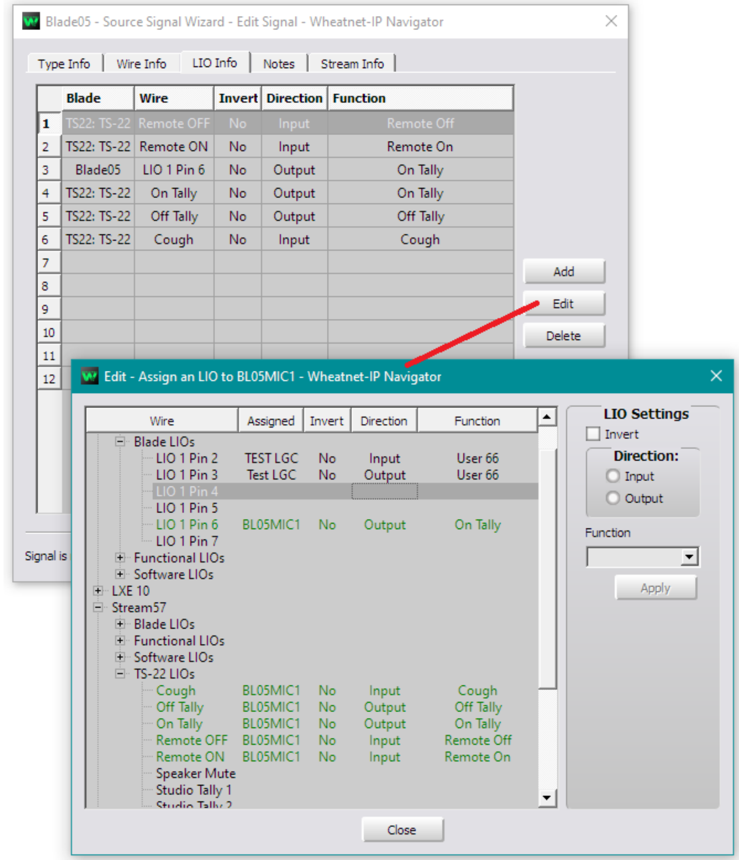


Figure 3-31 Logic Assignment tab

Switch to the Crosspoint map (**System > Crosspoint tab**) and locate the DMX console logic source: DMXTaly. That signal carries all the internal logic commands from the console so connect it to your new CR WARN LIO-only destination by clicking (or CTLR + clicking) on that crosspoint. A green square indicates it's a logic-only connection.

Now, when a channel with a CR Mic is turned on, there will be a contact closure between pin 4 (CR WARN) and pin 1 (ground). If a talk studio is associated with the console, a second LIO-only signal could use another pin, and be labeled STU WARN, with its Function set for *Studio 2 In-Use* so that when a Studio mic channel is turned on, there's a contact closure between the STU WARN pin and pin 1. These Logic jack pins are physically connected directly to Hot Mic or Warning LED lights but could also go to a warning light interface, to mic arm LEDs, to a mic skimmer, etc.

Using the Soft Button

The square white Soft button can be setup to manually trigger external logic and/or to light up in response from an external logic input by adding a new LIO-only Destination, as outlined in the previous section. Set the Output Function to **Switch 1** and/or the Input Function to **Switch LED 1**. The logic destination is then connected to the button source: **Spare01**, using the crosspoint grid, so that it triggers the outgoing logic and/or lights up following the incoming logic.

The Soft button can be setup to make a momentary connection and/or fire a Salvo using the **Blade > LIO Info** tab (Figure 3-32). The Soft button is named Spare Btn1, listed under Surface 1 Spares.

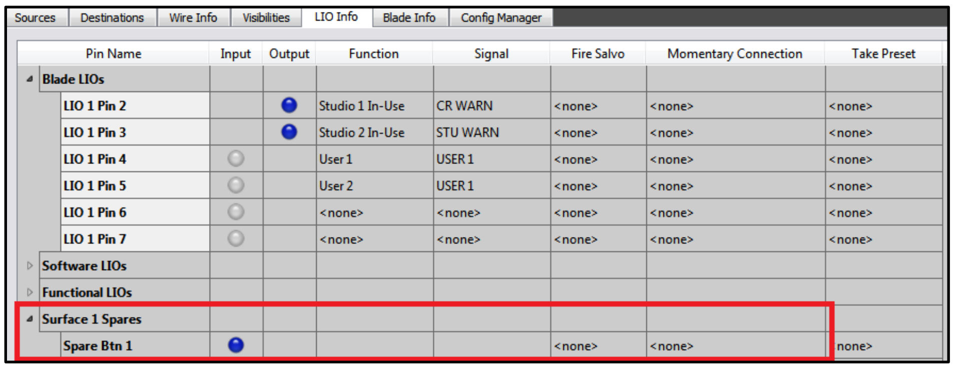


Figure 3-32 Blade > LIO Info tab

To use the Soft button as a momentary talk button, use the DMX Surface Setup **Buttons page tab** to first setup the Soft Button for *Momentary/Surface LED* so the button lights up when pressed.

In the **Blade > LIO Info** tab double-click on <none> in the **Momentary Connection** column for *Spare Btn 1*. This opens the dialog box shown in the left side of Figure 3-33. Click the checkbox to activate the Source and Destination signal selectors. In the example, the source is MIC1 (the board operator mic) and the destination is a call screener's monitor output (CALLSCNR). Clicking *OK* closes that dialog box and adds that entry to the Momentary Connection column for *Spare Btn 1*.

When the Soft button is pressed, the board operator mic (MIC 1) is switched into the call screener's headphones or speaker. When the Soft button is released the call screener's audio is reconnected to their monitor.

To assign the Soft button to fire a Salvo, double-click on <none> in the **Fire Salvo** column for Spare Btn 1. The Fire Salvo window opens (right illustration in Figure 3-33). Click the Fire Salvo checkbox to activate the *Salvo:* drop down list. Choose which Salvo to fire then click *OK* to close the window. The Salvo name then appears in the LIO Info tab.

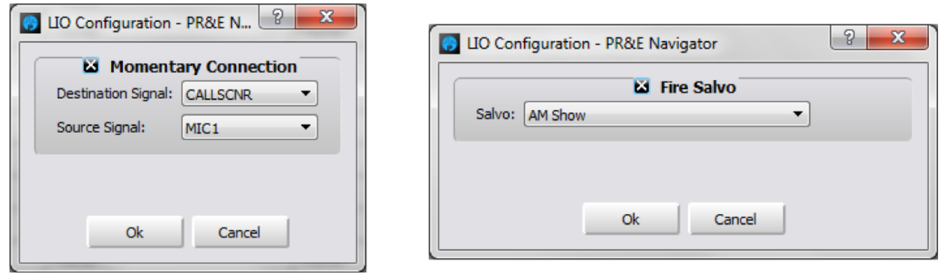


Figure 3-33 LIO Dialog Boxes

Note: The Soft button can be set to perform any combination of firing a Salvo, making a Momentary Connection, taking a mic processor preset, and sending out an LIO and/or SLIO command. This allows the Soft button to accomplish very complex actions at the press of the button.

Creating and Using Associated Connections

To create a new Associated Connection, click **System > Associated Connections tab**. If any Associated Connections have already been entered, they'll be listed in the tab, as shown in Figure 3-34.

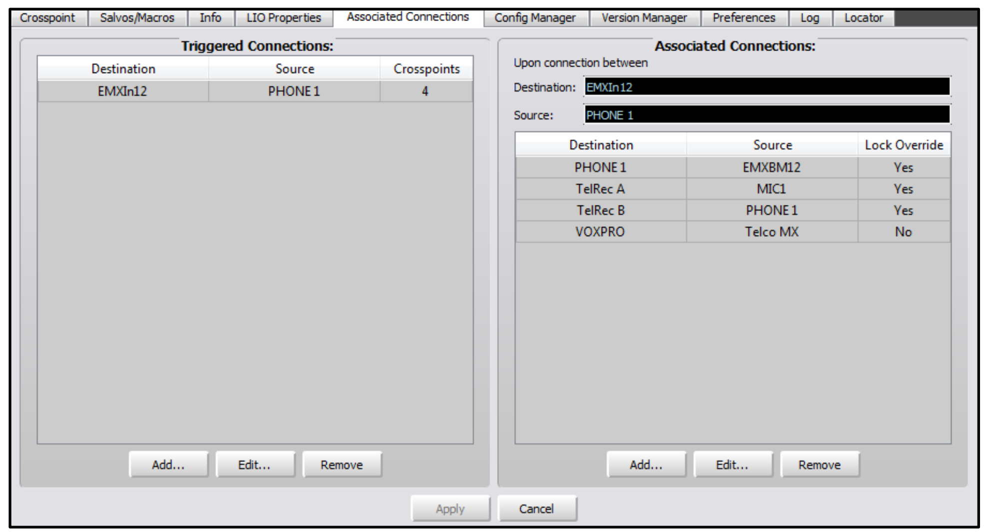


Figure 3-34 Associated Connections tab

To add a new Associated Connection, the **Triggered Connection** must first be defined in the left half of the tab. Click the *Add...* button to open the **Add Triggered Connection** window where the source-to-destination trigger connection is assigned. In Figure 3-35 on the next page, left illustration, shows the source set for PHONE 1 and the destination is fader channel 8 (DMXIn08). Clicking *OK* in the dialog box adds that connection to the Triggered Connections list (like that shown in Figure 3-34).

All the Associated Connections that should be made in response to that trigger connection are then defined on the right half of the tab. Highlight your new Triggered Connection then click the *Add...* button in the Associated Connections side of the window. This opens the **Add Associated Connection** window (Figure 3-35, right illustration) where a source-to-destination connection, made when the triggered connection is detected, is assigned. Note that this window has an additional selection: *Lock Override*. When checked this causes the source-to-destination connection to be made even if the destination is locked. When *Lock Override* is unchecked, the source-to-destination connection will not be made if the destination is locked.

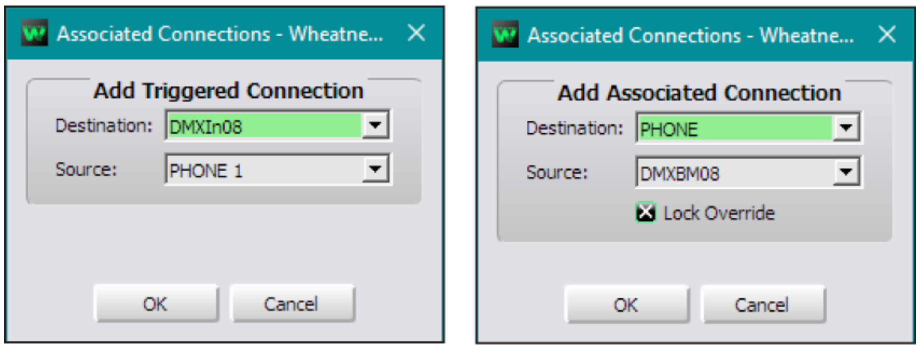


Figure 3-35 Associated Connection Dialog Boxes

Associated Connections are most often used to control the bus-minus connections for hybrids and codecs which are visible on multiple fader channels. For example, if Codec A is set as visible on channels 13 – 16, four Triggered Connections would be created to monitor for Codec A being taken on any of those four channels. The Associated Connections would then connect that channel's bus-minus to the Codec A output.

An example of a more complex Associated Connection is shown in Figure 3-36 which shows how four Talk Studio mics could be switched between a two consoles. When the Host mic is taken on channel 4 on the "EMX console" the Co-Host mic is automatically taken on channel 5, and the two Guest mics are taken on channels 6 and 7. The Studio warning logic is then connected to the "EMX Tally."

When the same Host mic is taken on channel 1 of the "DMX console," the Co-Host and Guest mics are connected to channels 2 – 4 on the DMX console and the "DMX Tally" is connected to the STU WARN logic output, which could be on the DMX, on the other console, on a Razor, or a Blade.

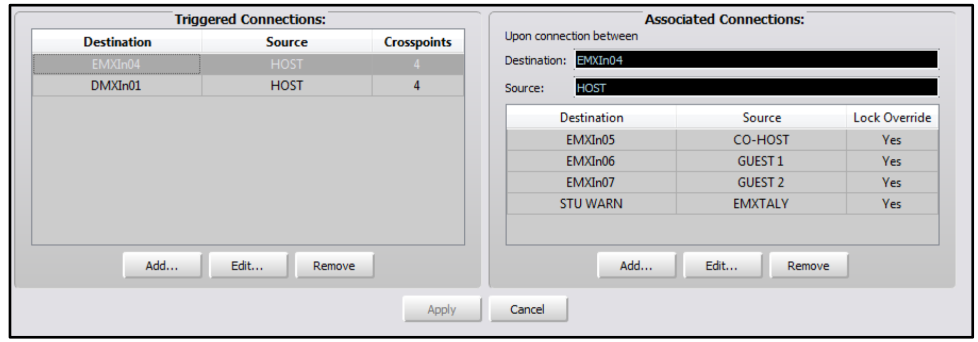


Figure 3-36 Creating Associated Connections for Sharing Talk Studio Mics

SOFTWARE UPDATES

Software revisions for the DMX Surface, Mix Engine, Razor I/O Interfaces, Blades, and other WNIP devices may be released. The DMX Surface software is updated using the DMX Surface Setup app's Hardware menu (Figure 3-37). The Mix Engine and Razor software is updated using Navigator's Version Manager tab (Figure 3-38). Note that to update Blade software requires a Navigator-3 license.

Surface Software Update

To see the current version of software on a DMX Surface, use the DMX Surface Setup app's *Hardware* > *Version…* menu item. A pop-up window lists the software version number and firmware running on that DMX Surface.

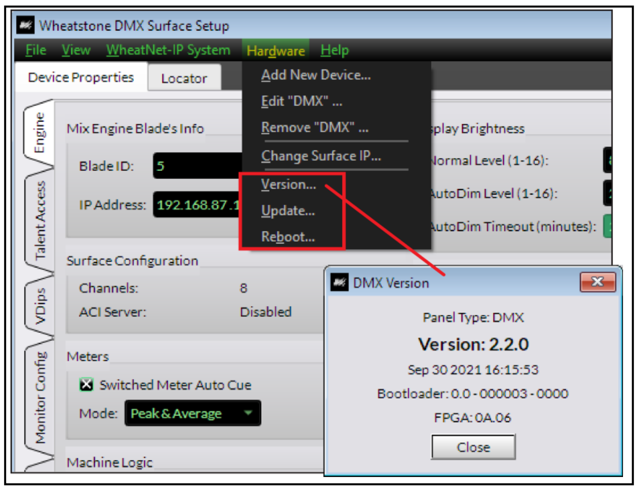


Figure 3-37 Surface Software Options

DMX Surface software is saved to that PC as part of the DMX Surface Setup app installation so if you get an updated version of the Surface Setup app it may also have updated Surface software. Updated versions of the Surface software may also be made available as a separate file download. To update the DMX Surface software, click *Hardware* > *Update…* to open a Windows search window pointing to the folder where the software file gets saved when the Surface Setup app is installed. If a software file was downloaded separately, navigate to that folder (e.g., Downloads). The software update file is: **DMX_surface_X_X_X.rbn** with the Xs being the version number of that software release.

Click *Open* to upload the new software to the Surface. This action can be done while the console is actively being used since the software only gets uploaded to the Surface's static memory. A popup message advises that the Surface must be rebooted to use the new software. Although the DMX will continue to pass audio while the Surface reboots, the Surface controls will not be available during the minute or so it takes the Surface to complete rebooting and again be ready for use.

Note: The PGM bus levels and the monitor outputs may momentarily duck or change levels as the Surface regains control after being rebooted.

Mix Engine & Razor Software Update

The Mix Engine and Razor software is saved to the PC when the Navigator app is installed. If Navigator is updated, then there may also be an updated version of the Blade software. Updated operating system software for the Blades may also be made available as a separate download file.

The software version running on each Blade can be viewed, and the software updated, using the *Version Manager* tab in Navigator (Figure 3-38). The Software Version column lists the software build currently running on each Blade, which includes the Mix Engine and each Razor Interface. Blades can be individually updated by checking just one or two Blades, or they can all be updated simultaneously by clicking *Select All*.

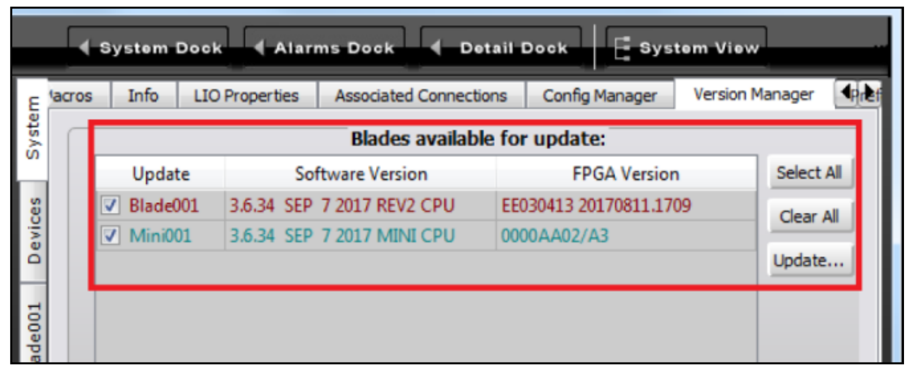


Figure 3-38 Mix Engine and Razor Software Version and Update options

To update the checked Blades, click *Update...* which opens a Windows search window on the folder Navigator used to save the software update file that it installed. If the software file was downloaded separately, then the software file may be in your "Downloads" folder so use the search window to locate the new software file. The Mix Engine and Razor software is in an archive file: **Blade_update_ X_X_XX.tgz**, with the Xs being the software version number of that software release.

Once the software gets uploaded to the Blade(s), each Blade must be rebooted to run the updated software. As the updated software is subsequently run there will be momentary muting or audio glitches on signals connected thru those Blades so, if possible, the software update process should be done while the system is bypassed and not carrying on-air audio.

This chapter covers DMX operations for board operators. It includes an overview of the DMX Surface controls and their functions; how to use the DMX to accomplish specific tasks; how to use the Advanced Channel Features—when not locked-out from board operator access; and how to use the Razor I/O Interface front panel controls and display.

CONSOLE OPERATION

DMX operation is identical, regardless of which size DMX Surface is being used—only the number of fader channels (either eight or sixteen) differs between the DMX-8 and DMX-16. Figure 4-1 is an overview of a DMX-8 Surface listing its main component parts.

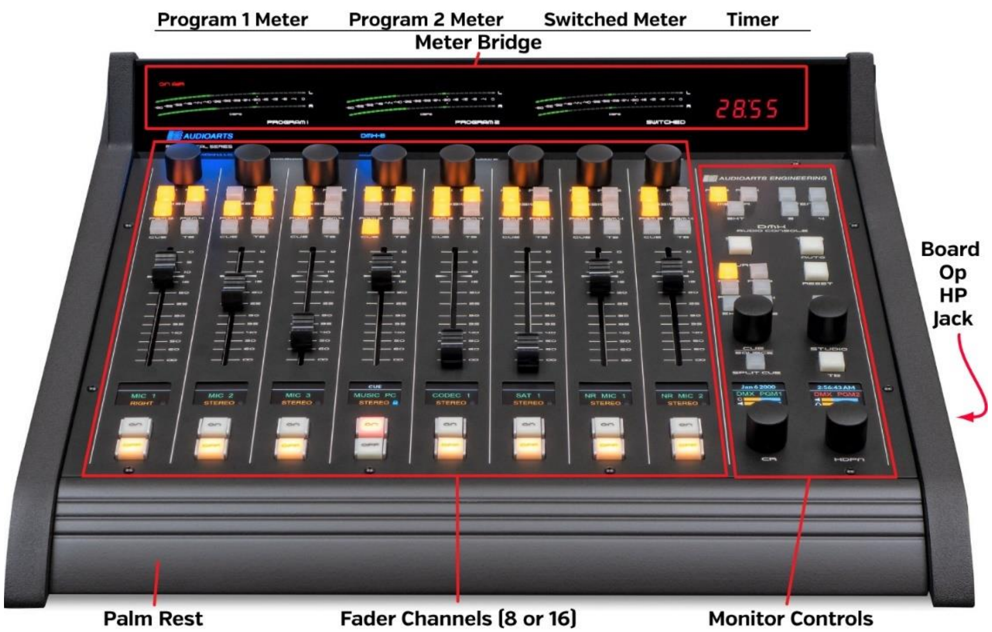


Figure 4-1 DMX Surface Overview

Fader channel controls are detailed in Figure 4-2. The monitor controls are detailed in Figure 4-4 on page 50. The first mention of each control in the text is in **Bold** text.

Meter Bridge Features

There are three VU/Peak Program meters in the Meter Bridge. The left meter shows the Program 1 (PGM 1) bus level while the middle meter shows the Program 2 (PGM 2) bus level. The right meter is switched to show the Program 3 (**PGM 3**) or Program 4 (**PGM 4)** bus levels, or an external source (**EXT**) as assigned using the **Switched Meter Source Select buttons** at the top of the left-hand column of monitor controls. Note that the switched meter may also show cue levels (a DMX configuration setting) while cue is active on any channel.

Assigning the source for the Switched Meter EXT button and Timer operation is covered in the *Monitor Control Operation* section (page 49).

Fader Channel Operation

Each fader channel normally has an audio source assigned to it. The source name is shown in green in the middle of the channel display. If a source has not been set for a channel, then **NoSource** is shown in the channel display.

Most fader channels have alternate sources, assigned when engineering configured the console to allow different sources to be "dialed up and taken." The alternate source names are listed in alphanumeric order when the **Channel Encoder** (the large knob at the top of each fader channel) is rotated while the channel is turned off. If the channel encoder is rotated while the channel is on, a **FADER ON** message appears in the display indicating you cannot change the source while the channel is on.

With the channel turned off, rotate the channel encoder. The channel display changes showing three source names with the middle name highlighted. Rotating the encoder Clockwise (CW) moves the highlighting down through the list of names. Turning it counterclockwise (CCW) moves the highlighting back up through the list of names. The source list wraps around for a continuous display. If *NoSource* is set as a visible channel source, seeing its name indicates the source list has wrapped around and you're at the top of the list again.

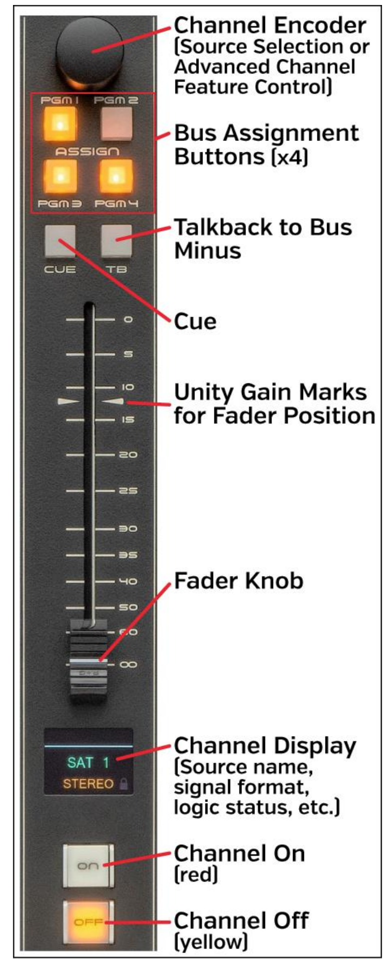


Figure 4-2 Fader Channel Controls

To select a new source, highlight its name then press down once or "click" the encoder knob. This "takes" the new source. Its name then appears in green text in the middle of the display.

Note: Channel functions may change when a new source is selected since channel settings may be associated with the audio source rather than with the channel. One example is monitor muting when the channel is turned on. If a Control Room mic is the source, the CR monitors will mute, but if a CD player is the source on that same channel the monitors won't mute.

Some channels may have a dedicated source like the board operator mic, the phone, or the media server playback channels, so there may only be one or a couple sources shown when the channel encoder is rotated. On channels with sources that are only used sporadically, like a CD player or an auxiliary studio input, there may be a dozen or more sources available even though that one source is typically "dialed up" on that channel.

Typically, a few channels near the monitor controls will be setup to list every system source that might need to feed air, be recorded, be fed to a network, or be used to feed an Internet streamer.

Bus Assignment Buttons

Each fader channel can be assigned to any combination of the program buses using the **PGM 1**, **PGM 2**, **PGM 3**, and **PGM 4 Assign buttons**. A lit button indicates that channel is assigned to that bus. An unlit button indicates the channel is not assigned to that bus.

PGM 1 – **PGM 3** have identical operations in that all three buses are fed post-fader and post-Channel On switch. In other words, to hear the channel's audio on these three program buses the channel must be assigned to that bus (the **PGM 1**, **PGM 2**, and/or **PGM 3** buttons lit) and the channel must be turned on and the **Channel Fader** must be "potted up."

PGM 4 may function differently from the other program buses because it can be setup for pre-Channel On switch operation (the audio is always connected to that bus) and/or pre-fader operation (the fader does not affect the channel source's level to the PGM 4 bus). These settings allow the PGM 4 bus to function as a Send bus, to feed an FX unit, or to function as an Off Line bus, to feed to phone callers and/or remotes.

When the PGM 4 bus buttons are setup as Off Line bus assignment buttons, the PGM 4 bus assign buttons then control which channels' audio is returned to the caller's or remote's bus-minus while their channel is off. Using the special features of the PGM 4/Off Line bus is covered in more detail in the next section (*Caller/Remote Operation*).

If no callers, remotes, or FX units are connected to the console, the PGM 4 bus will function the same as the other three PGM buses: the channels feed it post-fader and post-Channel On switch.

For an on-air studio, PGM 1 is typically used to create the on-air signal. This means that all channels assigned to PGM 1, turned on, and potted up will feed air. PGM 2 is often used to create a network or Internet stream feed which requires alternate program elements from what's being broadcast over the air.

The PGM 2, PGM 3, or PGM 4 buses may also be used to simultaneously feed selected channels to recorders to time-shift-record programming and/or to do voice-tracking while the console simultaneously feeds completely different source material to air.

In a production room, PGM 1 is typically connected to a PC running Adobe Audition or other on-screen editor. If the console is in a newsroom or a talk studio, the PGM 1 output might be included as a source on a fader channel on the air console to allow the board operator to mix in a news announcer going live from the newsroom while allowing the news announcer to control their own mix of playback actualities and other sounders along with their mic audio.

Caller/Remote Operation

When phone callers and/or live remotes are connected to the DMX, the PGM 4 buttons are typically configured as *Off Line bus assignment buttons*. The Off Line bus creates the caller or remote bus-minus signal while that caller or remote fader channel is turned off. This allows the board operator to carry on a hands-free conversation with the caller or remote using their mic. The caller and/or remote channels would be assigned to cue so the board operator can hear them in either the cue speaker and/or in their headphones.

The board operator's mic channel is typically assigned to both PGM 1 and PGM 4 when the PGM 4 button is configured by engineering to feed the Off Line bus. Since the Off Line bus is typically setup to be pre-switch and pre-fader the mic channel remains off when talking to a caller or remote. Typically, no other channels have the PGM 4 button lit so that the caller or remote only hears the board op's mic. However, if there's a host or a producer that also needs to talk to the remote or to a caller then their mic may also be assigned to PGM 4 so they can also talk to the caller or remote.

When the caller or remote channel is turned on to go live-to-air, their return audio (typically their channel's bus-minus audio) switches from the Off Line bus to the PGM 1 bus (assuming that's the bus feeding air). This allows the caller and/or remote site to hear all the show's on-air elements which are feeding air—minus their own voice, so they can interact with the host and guests in the studio, or the live remote host can interact with a caller.

When the caller or remote channel is turned off, their bus-minus audio switches back to the Off Line bus so the board operator can give cues for a live remote or conclude a call with a phone caller.

The audio that's sent to the caller or remote is often called an IFB or Interruptible Fold Back signal since, while the caller/remote listens to the fold back audio (e.g., their bus-minus audio), the board operator can interrupt that audio at any time by pressing the **TB** (Talkback) button on the caller or remote channel to give them cues.

Even though every channel has a TB button, it has no function—even though it will light when pressed, with channel sources like CD players since that channel's bus-minus signal is not connected to any destination.

When there's a lot of noise in the control room, typical of a morning zoo format, "hand's free" call answering, where the board operator or Host mic channel is assigned to the Off Line bus, may not work—or you may not want a caller to hear everything that's going on in the studio! In this case, the board op's mic channel would NOT be assigned to PGM 4, so to answer the phone or talk to a remote the board operator would need to press/hold the TB button on the caller/remote channel to talk with them.

When no channels are assigned to Off Line (no PGM 4 buttons are lit) then the caller/remote will not hear the control room unless the board operator presses and holds the caller/remote channel's TB (Talkback) button. Although this setup defeats hands-free operation, it ensures control room conversations are not heard by the caller or remote.

Note: Turning the cue speaker up too loud during a hands-free call can allow the caller's voice to be picked up by the host mic, which can cause echoes as their voice gets returned to them slightly delayed through the hybrid or codec. We recommend sending cue to the headphones to talk with callers.

There are two configuration settings which affect how a source is fed to the PGM 4/Off Line bus: each source can be independently set for Pre-Fader and/or Pre-On switch operation. Since there's no indication how these are assigned on the Surface, engineering will need to let board operators know how the PGM 4/Off Line bus is being used since these settings affect both regular PGM 4 and Off Line bus operation.

Note: When the DMX is setup to use the PGM 4 button for Off Line, any mics assigned to the Off Line bus, and setup for pre-fader and pre-On switch operation, will always be heard by the callers or remote even when the mic channels are turned off and the faders "potted down" if their PGM 4 buttons are lit.

If the PGM 4 Pre-Fader option is not assigned, then the mic channel faders must be "potted up" for the caller/remote to hear the channels assigned to the PGM 4/Off Line bus. If PGM 4 Pre-On is not selected, an unusual setting for this application, the channel must also be turned on to feed the Off Line bus, which means it must be unassigned to PGM 1, otherwise the mic audio will also feed air.

Cue

When the **Cue** button is lit that channel's pre-fader and pre-on switch audio is sent to the cue system to verify the correct audio is present on a satellite feed or a remote as well as for hands-free answering of phone calls. When the cue button is not lit, that channel's audio source does not feed the cue system.

The default connection for the cue bus is to feed a **Cue monitor** output. The output level is controlled by the **Cue level control** in the left-hand column of monitor controls. But cue may also be auto-switched into the board operator headphones. Cue can be fed in stereo (the **Split Cue button** is unlit) or in split mode (Split Cue button is lit). This puts cue in the left ear and sums the stereo monitor audio into the right ear.

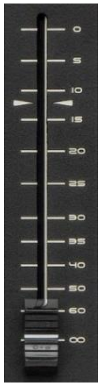
There are several other cue functions which are assigned during console configuration. Cue can be set to automatically turn off when a channel is turned on. It can also be automatically turned on when the fader is moved to full off. The cue audio levels may be shown in the switched meter while cue is active. The Cue audio can also be setup to feed the control room and/or the studio monitor outputs.

TB (Talkback)

Press/hold the channel TB button to momentarily talk into the Bus Minus audio for a caller or remote regardless of whether their channel is on or off. The TB button is typically used when the caller/remote is live (their channel is On) and the board operator needs to give them a cue. When the caller/remote channel is off, the board operator mic is normally assigned to PGM 4/Off Line and thus their voice will already be heard by the caller/remote while their channel is Off.

Note: While TB is pressed, the base mix audio (whether Off Line or a PGM bus) gets cut off to that caller or remote. Also, for talkback to work the console's talkback destination (DMXTkBk) must be connected to a source in **Navigator's** *Crosspoint Grid*. Typically, the board operator's mic input is the talkback source.

Fader Control



The channel faders control the audio level going to the assigned Program busses. The level range goes from full off (the fader is set to the ∞ symbol) to adding 12 dB of gain to the signal (when the fader is at the 0 indication).

In an ideal world, each fader knob on each channel would be set with their center line aligned with the two unity gain marks (the two arrows at the -12 position). When a channel is assigned to Cue and to PGM 1, and the fader is set to the unity gain marks, if the switched meter is set to display cue, the Switched meter and the PGM 1 meter levels will be identical.

Channel Control

To turn a channel on, press the **On button**. Red LEDs light up the button to show that channel is on. To turn the channel off, press the **Off button**. For most channel sources, yellow LEDs light up the Off button to indicate that channel is off. But some sources, like media server playback channels, have a logic command (Ready) which controls the Off button LEDs to indicate play status. This function was originally used with cart decks to show their status with no Off LEDs lit to indicate the cart was not ready; a solid Off LED indicates the cart is cued and ready to play; while a winking Off LED indicates the cart had already played. Some automation systems use this to indicate playback channel status.

Channel faders can also be setup to automatically control the channel on and off as they are moved. A function called EFS or Electronic Fader Start. This means the channel is turned on when the fader is moved up from full off (the ∞ symbol) and then turns off when the fader is moved down to full off.

Note: The channel on and off switches still function normally even with EFS being active.

Channel Display

Each channel has a color OLED display to show channel status. The normal display shows the source connected to that channel in green text in the middle of the display, like BLO1MIC1 shown in the left display in Figure 4-3. If no source is connected to the channel, then **NoSource** is shown in green text.

Figure 4-3 Channel Displays – Normal (left), While Selecting a New Source (right)

The display is divided by a white line with the area above the line listing whether channel equalization is active (**EQ**), whether Dynamics are in use (**DYN**), and whether Cue is active (**CUE**). When a new source is being selected by using the channel encoder the Device Name with the highlighted source appears above the line. In the right display in Figure 4-3 the highlighted source (BL001S02) is on Blade01, which is the default name for the DMX Mix Engine.

In the normal display, the channel's mode setting is shown in orange below the channel source name. The default setting for every channel is Stereo (a mono signal like a mic or a phone caller is automatically fed to both left and right channels), but alternate settings: Left, Right, or Mono may also be shown.

To either side of the Mode name are icons to indicate channel status. On the right side is a lock indicator to indicate the channel source's logic control status. A gray lock icon indicates that signal has no logic control assigned to it. A blue lock icon, as shown in Figure 4-3, indicates that channel fader has logic control over that source. When the icon is an opened red lock, it indicates that source is active on another channel fader (either on this DMX or on another console) which has the logic control and thus this fader channel doesn't have any logic control.

To the left of the Mode is a mic icon when the channel's source is a Control Room, a Studio, or an External mic. This icon turns red while the channel is on.

Rotating the Channel Encoder, while the channel is off, changes the normal display to show a list of the alternate source names available on that channel (Figure 4-3, right display). The current source is initially shown highlighted in the middle. Rotating the encoder clockwise moves the highlighting down the list and rotating it counterclockwise moves the highlighting up through the list.

Source names are typically listed in alphanumeric order, but the source names may also be ordered by location (a console configuration setting). This alternate setting means that when the encoder is first rotated, a list of networked devices like Blade01, Blade02, Blade03, etc. is shown. Highlight a device then click the encoder to select that device. The list then shows the source names on that device in alphanumeric order. Clicking the encoder selects the highlighted name and returns the display to the normal display where the selected source name is shown in green.

Monitor Control Operation

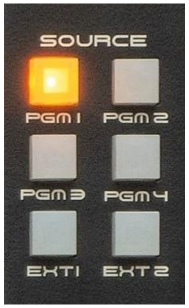
Monitor controls on the DMX Surface are identified in Figure 4-4 on the next page. The controls are arranged in two columns. The left column has the switched meter source selectors and the control room (CR) monitor controls. The right column has the studio (ST) monitor controls along with the Event and Timer controls.

Left Column Controls

At the top of this column are the **three Switched Meter Source** select buttons (PGM 3, PGM 4, and EXT). A lit button indicates which source is shown on the Switched Meter.

A source may have been assigned to the **EXT** (External) button during DMX setup, but it can be changed as required by the board operator. Press/hold the EXT button for about three seconds: until the EXT button's LED begins blinking. The left-hand monitor OLED display switches to show the list of sources that can be assigned to the EXT button. Rotate the **Cue/Source Encoder** to highlight the desired source name in the display. Tap or "click" the Cue/Source encoder to assign the highlighted source to the EXT button. The EXT button source selection is saved as part of a console Event so the EXT source can change depending upon which console Event is active.

A white **Soft button** is below the three switched meter buttons. This button is user-defined, so engineering needs to identify whether this button is active and, if so, what it's setup to do. Here are a few examples of how the Soft button could be configured: to switch the console between board operated and automated operation; as a remote dump button, with safe indication, for a profanity delay; as a talkback button to talk to a producer, call screener, or a news room; or as a hotline phone ring indicator.



Below the Soft button are six **Monitor Source Select buttons** to select which source feeds the CR monitor speakers and the operator's headphone jack. The current source button is lit and its name is shown in green in the middle of the **CR Display**.

Note: A "wild" or temporary source can feed the CR monitors by press/holding the Cue/Source encoder, for about three seconds, to show a list of source names in the CR Display. Rotate the Cue/Source encoder to highlight the desired source name. Click the Cue/Source encoder to select that source. Its name then appears in red text indicating it's a wild or temporary source. A source in red text can also indicate that Navigator was used to connect a wild source to the CR monitor. A wild monitor source can also be saved in an Event or in a Salvo.

Cue

The Cue/Source encoder normally controls the level for an outboard amplified cue speaker connected to the Cue Monitor Output on the Mix Engine. Rotating it affects the Cue bar graph, the top orange/blue bar graph in the CR Display with the C icon.

Note: The cue speaker(s) should also be set to mute whenever a control room mic channel is on.

The Cue/Source encoder is also used to select the source assigned to the Switched Meter EXT button and to the EXT 1 and EXT 2 Monitor Select buttons. To assign a source, press/hold the EXT, the EXT 1, or the EXT 2 button for about three seconds—until its LED begins blinking and a list of source names appear in the CR Display. Rotating the Cue/Source encoder steps through the visible source names. When the desired source is highlighted, click the Cue/Source encoder to assign that source to the blinking button. The blinking EXT, EXT 1, or EXT 2 button then turns off indicating the source was assigned.

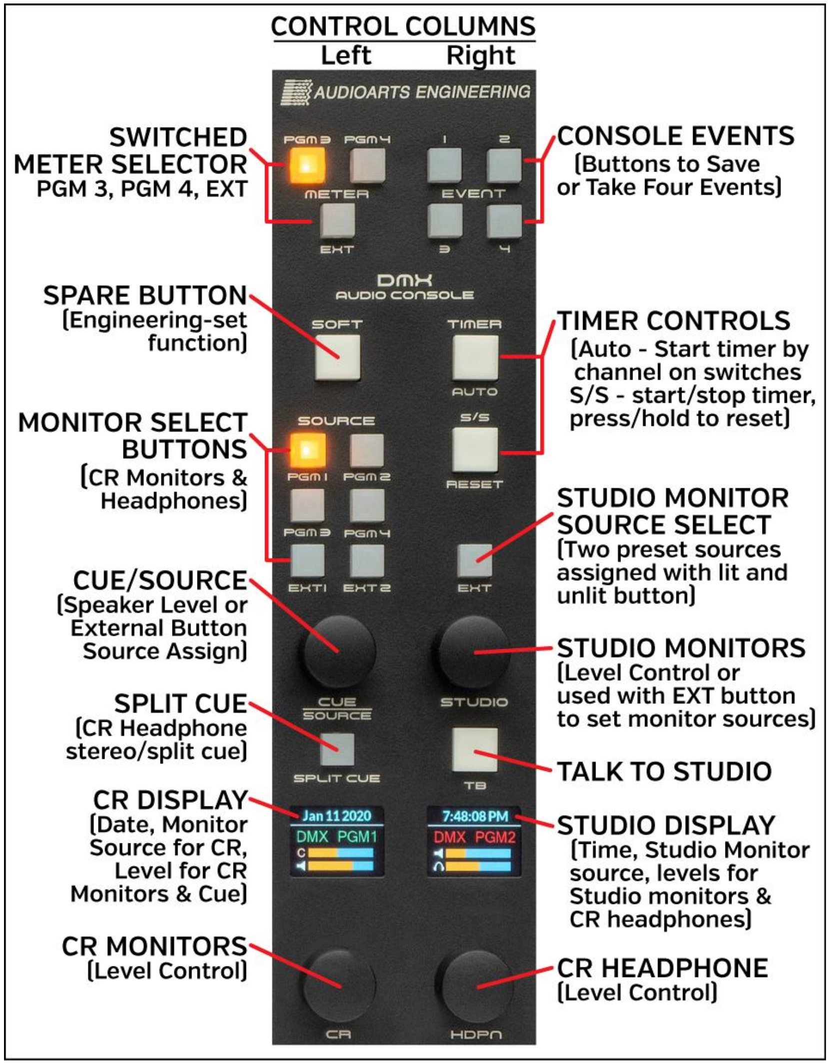


Figure 4-4 Fader Channel Controls

Split Cue

The **Split Cue button** affects how cue is fed to the operator headphones. When lit and cue is active, cue audio goes to the left ear and the CR monitor source gets summed in the right ear. When unlit and cue is active, cue is sent to both ears (stereo cue) cutting off the CR monitor source audio.

Note: Whether cue is auto-switched into the board operator headphones is a console setup option. Cue can also be setup to go to the CR or Studio monitors.

Monitor Displays

The two OLED displays in the monitor section: CR Display and Studio Display, show the current date and time above their white lines. The name of the current CR monitor source is shown in the middle of the CR Display while the current Studio monitor source is the shown in the middle of the Studio Display. A name shown in green text indicates the source is assigned to one of the select buttons. A name in red text indicates a “wild” monitor source—one not assigned to any of the source select buttons, is the active monitor source.

The four bar graphs at the bottom of these two displays show the Cue and CR monitor speaker levels in the CR Display and the Studio monitor and CR headphone levels in the Studio Display.

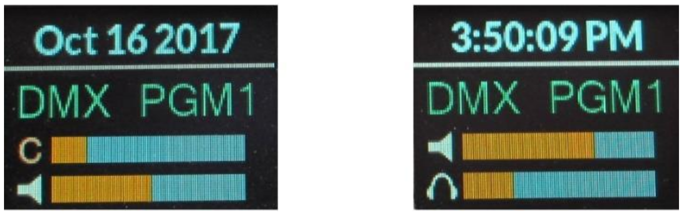


Figure 4-5 Monitor Displays

The Control Room monitor speakers and the cue speakers are typically muted, and an on-air warning light triggered, whenever a control room mic is turned on. Likewise, when a studio mic is turned on the Studio monitors mute. A red X is displayed over the bar graph icons to indicate an output is currently muted.

CR Monitor Speaker Level Control

The **CR Encoder** controls the level of the control room monitor speakers. Its relative level is shown in the lower orange/blue bar graph with the speaker icon in the CR Display.

Right Column Controls

The four console **Event buttons** are at the top of this column. These buttons allow four console setups to be saved during console configuration so that they can later be ‘taken” to change the console setup. An Event saves the current channel sources, the channel bus assignments, and any Advanced Channel Features settings, along with the selected CR and Studio monitor sources and the sources assigned to the various EXT buttons. Once four Events are saved, the Event Save function is typically locked out from operator use by engineering, so that board operators can only take Events.

Even though taking an Event affects the entire DMX Surface and could change every fader channel setting and monitor control panel setting, most Events typically only change a few channel settings and may not change anything in the monitor section since the console may not need that much reconfiguration between different shows or dayparts.

The daypart, show, or console function like production, backup air, morning show, etc. that’s assigned to each Event button must be identified by your engineering department. Here’s an example of how the four Event buttons could be configured:

EVENT 1 – Sets up the console for a multiple-mic morning show with traffic and news/weather remotes

EVENT 2 – Sets up the console for a regularly scheduled live remote show

EVENT 3 – Sets up the console for single-mic board operation with one caller channel

EVENT 4 – Sets up part of the console for voice tracking while the rest of the console continues to feed air

Taking an Event is a two-step process:

1. Arm the Event. To arm an Event, press/hold the Event button you want to take for about three seconds until it begins blinking. Release the button. The blinking indicates that Event is now "armed." It stays armed for an indefinite time so if you must switch events at a specific time, you can arm it at any time prior.

2. Take the Event. At the proper time (like during the last commercial in a stop set before the new daypart starts), press and release the blinking Event button. It lights solid to indicate that Event is now active. All Surface settings are updated within a few seconds.

Note: Fader channels which are turned on when an Event is taken are not affected by the new Event settings. They maintain their current settings. Once those fader channels are turned off, re-arm and re-take the same Event (press/hold the lit Event button until it blinks, then let go, then press it again) to update those fader channels.

To save an Event, setup the console as required then press/hold an Event button for about six seconds. The button will begin blinking after about three seconds (continue holding the button). In another three seconds it will light solid indicating the Event settings have been saved, which erases any Event that was previously saved on that button.

Note: If you press/hold an Event button and nothing happens that indicates both the Take Event and Save Event functions are locked out. If the button begins blinking but never lights up solid, that indicates the Save Event function is locked out, but the Take Event function is active. To cancel taking an Event when an Event button is blinking, tap another Event button.

Timer Controls

The meter bridge timer is controlled using two buttons: **Auto Reset** and the **S/S (Start/Stop) button**.

To manually start the timer, tap the S/S button. To stop the timer while it's running, tap the S/S button. The elapsed time is displayed. Tapping the S/S button again starts the timer from the elapsed time. To reset the timer to 00:00 press/hold the S/S button. If running, the timer resets to 00:00 and begins counting. If stopped, the timer just resets to 00:00.

When **Auto Reset** is lit, the timer will be reset to 00:00 and will begin counting when a fader channel, whose source has the *Timer Restart* option checked, is turned on. Which sources have this Timer Reset capability is set by engineering during console configuration.

Studio Controls

The **Studio Encoder** is normally used to control the level of the monitor speakers in a separate talk studio, news room, producer location, call screener, performance studio, or other external location—anywhere that a microphone, when connected to the console, would need monitor speaker muting when a studio mic is live. The Studio monitor speakers are automatically muted, and a studio warning light typically triggered, when a studio microphone is on.

The volume of the STU output on the Mix Engine is normally controlled by the Studio encoder, but if a studio-mounted level control is used, the Surface's STUDIO encoder would be set as inactive (a console configuration setting).

The studio monitor source can be toggled using the **Studio EXT** button. Note that you can select two sources: one source is selected when the EXT button is lit, and a second monitor source is selected when the EXT button is unlit.

To set the unlit EXT button source, press/hold the unlit EXT button until it begins blinking (about three seconds). The Studio display switches to show source names that can be assigned to the EXT button. Use the Studio encoder to highlight a source name then click the Studio encoder to assign it as the unlit EXT button source.

To set the lit EXT button source, press/hold the lit EXT button (for about three seconds) until it begins blinking. Use the Studio encoder to select the source as was done for the unlit EXT button.

An "on-the-fly" studio source can also be selected by press/holding the Studio encoder for about three seconds, until the list of visible source names appears in the Studio display. Highlight the desired source then click the Studio encoder to select that source. Its name is shown in red text in the Studio display (shown in Figure 4-4 on page 50) to indicate it's a temporary monitor source.

The list of the source names available to be assigned to the EXT button or as an on-the-fly monitor source is assigned by engineering during console configuration.

The board operator can talk into the studio monitor speakers using the **TB button** located just below the Studio encoder. See the *TB (Talkback)* section on page 48 about setting the source for the talkback audio.

Board Operator Headphone Level

The **HDPN encoder** controls the volume of the board operator's headphones. Its level is shown in the lower orange/blue bar graph with the headphone icon in the CR Display.

Note: A 1/4" TRS headphone jack is located on the right side of the Surface, in-line with the monitor displays, for the board operator headphones. Consumer headphones with 3.5mm TRS plugs require using a 3.5mm-to-1/4" TRS adapter. Since the top of many of these adapters, when plugged in, would be flush or even sit below the side panel, they can be difficult to unplug. TRS adapters with longer shafts are available that extend past the side panel. Otherwise, require board operators to use headphones with 1/4" TRS plugs or ones that incorporate screw-on TRS adapters so they can easily be unplugged from the 1/4" TRS jack.

USING THE ADVANCED CHANNEL FEATURES

Each fader channel has three Advanced Channel Features: Mode, Panning, and EQ & Dynamics. These channel features may not be available for board operator access since any or all can be disabled using the DMX Surface Setup app's *Talent Access page tab* (Figure 4-6 on the following page). For regular use of the console, these controls are typically disabled (unchecked) for on-air consoles but are usually enabled (checked) for production studios and other non-air console uses.

When an Advanced Channel Feature is active on a channel, but the channel encoder is not touched for about 20 seconds, that channel's display—or displays when the EQ & Dynamics feature is active, timeout and the channel display returns to the normal view.

During console setup the auto-timeout feature can be disabled which can simplify setting up EQ and Dynamics. To stop the auto-timeout feature uncheck the Enabled option in the *Timeout* section of the Talent Access page tab, as shown in Figure 4-6. Typically, this option will be checked prior to releasing the console for daily operations.

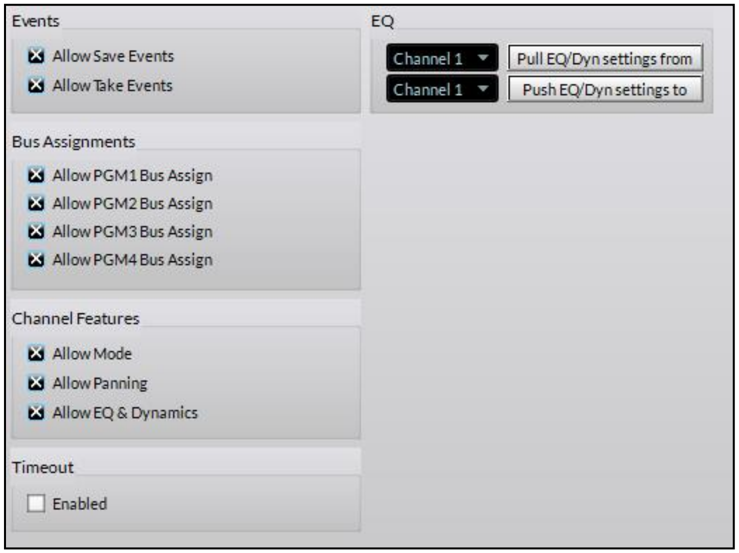


Figure 4-6 Talent Access Controls in Surface Setup App

Mode & Panning

When the Channel Feature *Allow Mode* is checked in the Talent Access page tab, double-clicking any channel encoder (tapping it twice quickly) switches the channel display to show **MODE** above the white line. Rotating the encoder steps through the four audio modes: LEFT only, RIGHT only, MONO sum, and STEREO, available for each fader channel.

Double-clicking the encoder, while **Mode** is active, switches the channel display to show **PAN** above the white line if *Allow Panning* is checked in the Talent Access page tab. Now, rotating the channel encoder pans the signal from center toward full-left or toward full-right. If *Allow Mode* is not checked, double-clicking a channel encoder brings up **PAN** rather than **MODE**.

EQ & Dynamics

If *Allow Mode*, *Allow Panning*, and *Allow EQ & Dynamics* are checked, double-clicking the encoder while Pan is active changes eight channel displays to show the various **EQ & Dynamics controls**. When *Allow EQ & Dynamics* is unchecked, double-clicking the encoder while Pan is active returns the display to the normal mode. If *Allow Mode* and *Allow Panning* are unchecked in the Talent Access page tab, but *Allow EQ & Dynamics* is checked, double-clicking a channel encoder immediately changes eight channel displays to show the EQ & Dynamics controls. To exit the EQ & Dynamics controls, double-click the channel encoder again. The channel displays then return to their normal views.

The EQ & Dynamics Advanced Channel Feature takes over all eight channel displays on a DMX-8. On a DMX-16, when EQ & Dynamics is active on channel 1 to 8, the displays on channels 1 - 8 show the EQ & Dynamics screens. When it's active on channel 9 to 16, the EQ & Dynamics screens are shown on the channel 9 – 16 displays. Six of the channel OLED displays show the EQ and Dynamics status (Figure 4-7).

Figure 4-7 EQ & Dynamics In/Out Selection Screens

The channel 7 / channel 15 display shows a graphic of how the EQ or Dynamics settings are affecting the frequency response and gain (the left OLED display in Figure 4-8). The channel 8 / channel 16 display (the right OLED display in Figure 4-8) shows which channel is being affected by the EQ & Dynamics settings above the white line. Below the line is a list of the various EQ and Dynamics screens. Rotating the channel 8 / channel 16 encoder steps through this list of control screens. The highlighted selection (like EQ Lo shown in Figure 4-8) affects what's shown in the other seven EQ & Dynamics screens.

The active channel's OLED display is outlined in white to further identify which channel you are affecting. In Figure 4-8, since channel 7 is active the OLED display on channel 7 is outlined in white.

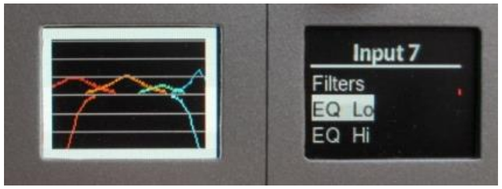


Figure 4-8 Active Channel Indications

Saving & Recalling EQ & Dynamics Settings

EQ & Dynamics are assigned on a channel basis. The settings for each channel are saved in the Surface's memory so the current settings get restored if the Surface gets rebooted or power cycled.

When a channel has any of the five types of EQ (the settings shown in Figure 4-7, screens 1 - 5) set In instead of Out, **EQ** is shown above the white line in that channel's normal display to indicate that equalization is active. When Dynamics (Compression and/or Expansion) is set In, **DYN** is shown above the white line in that channel's normal display to show that Dynamics control is active.

Note: When shipped from the factory, fader channels 1 and 2—typically used for the Host and a Guest mic, have microphone EQ and Dynamics applied to them, which is why **EQ** and **DYN** are displayed on those two channels by default.

Once the EQ & Dynamics settings are set on a channel, these settings can be "pulled" from that channel and saved to an "EQ config file." A previously-saved EQ config file can then be "pushed" or loaded into another DMX channel. Both the "Push" and "Pull" functions are done using the DMX Surface Setup app's EQ Section on the *Talent Access page tab* (Figure 4-9).

To pull the EQ and Dynamics settings from a channel, use the upper row of the EQ section controls to select the channel, using the drop-down list, then click the *Pull EQ/Dyn settings from* button. A File Save dialog box opens to name the file (like "hybrid" shown in Figure 4-9). Click Save to save the file (the .dmx_eq suffix is auto-added to identify the file as an EQ Config file.

To recall a previously saved EQ config file, select which channel you want to apply the settings to, using the lower EQ row drop-down list, then click the *Push EQ/Dyn settings to* button. A file find dialog box opens so a saved EQ Config file can be selected to apply to that channel.

Note: The EQ and Dynamics settings assigned to faders 1 and 2 at the factory are saved in the EQ file: WNIP_mic.dmx_eq.

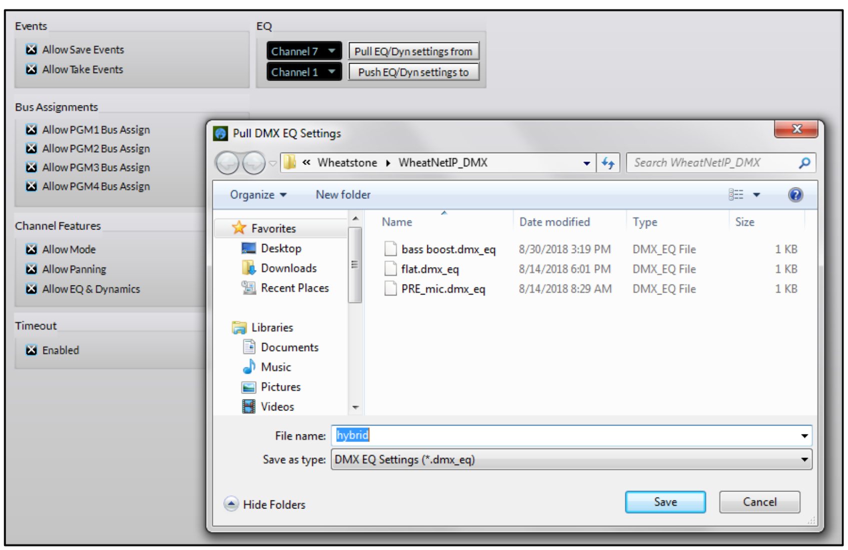


Figure 4-9 Saving & Recalling EQ Configuration Files

EQ & Dynamics Screens

While the EQ controls are active, the channel 7 / channel 15 display shows an audio frequency graph (Figure 4-10, left and middle illustrations) that shows the current settings of the various EQ controls. The EQ graph shows frequency response from 20 Hz (left side) to 20 kHz (right side) with the center being roughly 600 Hz since it's a logarithmic scale. The middle horizontal line represents unity gain (0 dB) with each white horizontal line above or below representing 5 dB of gain or cut.

The four parametric EQ graphs are color-coded. The filter controls and the graph for the HPF and Low EQ settings is red; the Low-Mid EQ is orange; the High-Mid EQ is green; and the LPF and High EQ settings are blue.

If no EQ is active (all EQ status are Out) then lines are shown across the display, as shown in the left screen in Figure 4-10. When any EQ is active (EQ status is In) then the audio passband is represented in gray, as shown in the middle screen in Figure 4-10.

When the Compression or Expansion control screens are active a Dynamics display is shown (the right screen in Figure 4-10). It's a very different type of graphic to show the ratio between the incoming audio and the dynamically processed output audio. When Dynamics is Out the response line is gray, but when Dynamics is In, the line is orange. The top half of the display reflects the Compressor settings while the bottom half reflects the Expander settings.

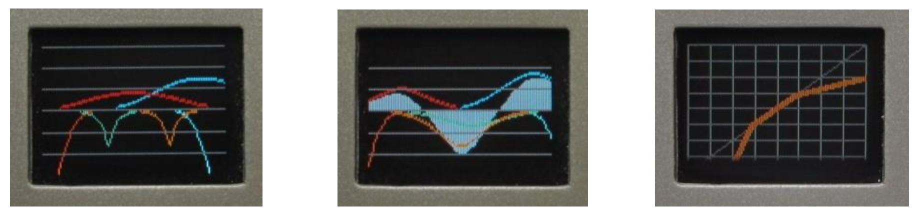


Figure 4-10 EQ Out, EQ In, and Dynamics Graphic Screens

Even though the EQ & Dynamics settings are made on a channel-basis, rather than being applied to a particular source, because these settings are saved with a console **Event,** along with the channel sources and other channel and monitor settings, with careful planning one can apply different EQ & Dynamics, Mode, and Panning to specific audio sources by using Events.

Note: Taking an Event can change the EQ & Dynamics settings, mode, or pan settings currently assigned to the channels.

High Pass & Low Pass Filters

The High Pass Filter (HPF) and Low Pass Filter (LPF) can be used independently on any channel (each can be set In or Out) to narrow the audio frequency range of that channel's audio to less than 20 Hz – 20 kHz.

The frequency pass band is shown graphically as the controls are adjusted: a red line represents the HPF or low frequency cutoff while a blue line represents the LPF or high frequency cutoff.

Adjusting the channel encoder on the channel showing the HPF screen (Figure 4-11, left OLED display) adjusts the frequency where the lower sounds begin to be rolled off or attenuated. The control range is from 16 Hz to 500 Hz. Audio below the selected frequency is rolled off at 24 dB per octave. Audio above the selected frequency is not affected. The HPF is commonly used to remove low frequency rumble and sounds emitted by air conditioners, AC line hum, nearby traffic, footsteps, cabinet noises picked up by moving microphone arms, etc.

Figure 4-11 High and Low Pass Filter Screens

Adjusting the channel encoder on the channel showing the LPF display adjusts where the higher frequency or brighter sounds begin to be attenuated. The control range is from 20 kHz down to 1 kHz. This control can selectively remove higher frequency sounds like air conditioner hiss, chair squeaks, and other high frequency noises without affecting the voice sound—until the frequency is adjusted below about 5 kHz. Audio above the selected frequency is rolled off at 24 dB per octave while audio below that frequency is not affected.

The HPF and LPF can be used together to create the classic phone filter effect. Adjust the HPF to about 300 Hz and the LPF to about 3 kHz. If this setting is saved as an EQ file, it can then be loaded into any channel using the DMX Surface Setup app’s EQ controls.

Four-Band Parametric Equalization

The four parametric EQs (EQ Lo, EQ Lo Mid, EQ Hi Mid, and EQ Hi) are all identical in operation and have the same set of parameters and control ranges.

The three EQ Lo screens are shown in Figure 4-12. Note their red color which matches the color shown in the EQ graph display. Each parametric EQ has a FREQ (frequency) control to adjust the center-point of that EQ; a BW (bandwidth) control to adjust the width of frequencies being affected by that EQ; and a LEVEL control to set the amount of boost or cut applied to that equalizer.

These controls are active when EQ Lo is In and the Lo Shelf EQ is set Out. The channel 8 / channel 16 encoder is used to select between showing EQ Lo (which shows both the EQ Lo and the EQ Lo Mid controls) and EQ Hi (which shows both the EQ Hi Mid and the EQ Hi controls).

Figure 4-12 Parametric EQ Lo and Lo Mid Screens

Each parametric EQ’s center frequency is adjustable from 16.1 Hz up to 20.2 kHz. The bandwidth is adjustable from 0.20 octave up to 3.00 octaves wide. Each equalizer can boost or cut the selected frequency band by +14 or -14 dB from unity gain in 0.1 dB steps.

Typically, the parametric equalization is adjusted prior to adding any compression or expansion since the EQ settings and the compression and expansion settings interact with one another, thus affecting overall operations.

EQ Lo Shelf & EQ Hi Shelf

These two filters are only active when EQ is set In along with either or both Shelf’s being set In. When active, the screens for the EQ Lo and/or EQ Hi are affected. While the Shelf is active it assigns a fixed bandwidth, so the BW (Bandwidth) screen is not shown. The FREQ and LEVEL controls operate the same as for the regular parametric equalizer.

The two Lo Shelf controls are used to boost or cut the low frequencies below the selected frequency by the amount set by the level control.

The two Hi Shelf controls are used to boost or cut the high frequencies above the selected frequency by the amount set by the level control.

The channel encoder with the EQ Shelf LEVEL screen sets the amount of boost (up to +14 dB) or cut (up to -14 dB) that’s applied. The channel encoder with the EQ Shelf FREQ screen sets where that shelf ends or begins. The Lo Shelf FREQ control adjusts where the low shelf ends, from 20.3 Hz up to 198 Hz. The Hi Shelf FREQ control adjusts where the high shelf starts, which can be from 4 kHz up to 20.2 kHz.

Dynamics – Compression and Expansion

When **Dynamics** is set In, the Compressor and Expander functions are activated. The Compressor has five controls with an orange background. They affect the overall dynamic range of the audio signal by controlling the loudest audio levels. The six Expander controls, which have a teal background, work at the opposite end of the dynamic range, controlling what happens with lower-level audio signals.

Compression

A Compressor controls the overall loudness of an audio signal. A small amount of compression is often applied to mics and phone callers to “even out” their overall audio levels. Applying higher levels of compression, along with setting a higher threshold, allows one to obtain consistent maximum loudness for “shock jocks” and “in-your-face” talk show commentators.

The compressor in the DMX is a “soft-knee” type which means its operation is less noticeable—for most settings. Setting the controls to their extremes: to get that “loud and in your face” sound that’s right on the edge of audio distortion, will cause audible compression artifacts.

To adjust the Compressor (Figure 4-13 shows the compressor screens), set the Compressor RATIO to 1.0:1 (no compression) and the Compressor MAKEUP level to 0.0 dB (no added gain). The Threshold (THRESH) point sets where the compressor begins to act upon the audio (which can be set from -40 dB up to +10 dB). A good place to begin is 0 dB.

Figure 4-13 Compressor Screens

As the audio level goes above the Threshold point, the compressor begins controlling the ratio of the outgoing audio when compared to the incoming audio. A RATIO of 1.0:1 (or no compression) means the output level tracks the input level. As the RATIO control is adjusted from 1.0:1 up to the 6.0:1 range, the amount of level control over the compressor’s output is increased. With a 2.0:1 setting, when the input rises 10 dB over the threshold level, the compressor’s output only increases 5 dB. With a 6.0:1 ration a 10 dB increase in the input audio (above the threshold) means the audio output only increases by 1.7 dB.

Setting a very high RATIO, like 10:1 up to 20:1, sets up the compressor to function more as a limiter. Typically, with high RATIO settings you’ll also set the THRESH level higher since you only want to only “squash” the very hottest input signals so you can maintain the compressor’s output as loud as possible without overdriving and causing distortion further on in the signal chain. If needed, the MAKEUP gain can be adjusted to compensate for high compression levels.

The ATTACK control sets how fast the compressor reacts (from 0.10 mS up to 330 mS) as the audio level goes above the THRESH level. Usually, you want to use the fastest setting possible, but it all depends upon the program audio since using faster settings can be audible. Using very slow settings will allow the audio to go way past the threshold point before the compressor kicks in which can lead to momentary distortion and audio “pumping” as the compression kicks in, noticeably lowering the output.

The RELEASE control sets how fast the compressor “un-compresses” the audio signal once the audio drops below the THRESH level. Release can be set from 50 mS up to 3.00 seconds. Again, using settings in the middle of the range allows for the most transparent compression. Using extremely fast settings can cause audio “pumping” as the compressor opens and closes in response to fast audio changes.

Again, the type of program audio that’s being compressed, whether voice, music, phone call, remote, etc., must be considered when adjusting the Compressor settings.

Expander

The six Expander screens are shown in Figure 4-14. The Expander is an automatic attenuator controlled by the settings of the RATIO and DEPTH controls. These controls allow one to slightly duck or dim the audio down to virtually muting (gating) the input audio once its level falls below the THRESH (threshold) setting. The Expander is most often used to automatically mute inactive mics in a multi-mic talk studio by gating their audio or to decrease background noise on phone calls or remotes by ducking the audio as it falls below the THRESH setting.

How quickly this ducking/gating occurs and releases—and thus how noticeable it is, is controlled by the OPEN (1.00 mS up to 100 mS), HANG (0.00 mS up to 1.00 S), and CLOSE (50 ms up to 3.00 S) settings. HANG sets how long the expander waits, as the audio level passes the THRESH setting, before it either opens or closes. CLOSE sets how fast the ducking or gating occurs and OPEN sets how fast the audio gets un-ducked or un-gated as the audio level rises above the THESH setting.

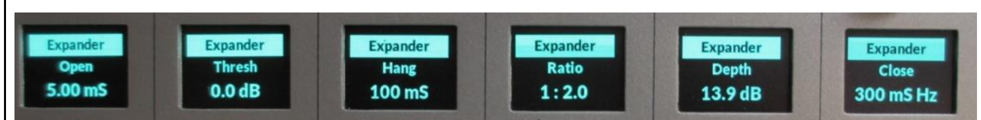


Figure 4-14 Expander Screens

The DEPTH control sets whether the audio is ducked or gated as the audio level drops below the THRESH setting. At 0.0 dB no ducking or gating of the audio occurs. Settings from 10 – 25 dB cause the signal level to duck. Higher settings, 25 - 40 dB, create a virtual audio gate to silence the audio once its level drops below the THRESH setting.

As with setting the Compressor controls, adjusting the Expander controls toward the end of their range will cause the ducking and gating to be more audible. For gating of unused mics in a talk studio, you’ll want to adjust the THRESH so that the mics don’t open when other active mics are being used in the studio but will still quickly un-gate when someone handles the mic or begins talking into it. Typically, slower HANG and CLOSE times are used to prevent the gating from occurring during a breath or a normal speech pause. A faster OPEN setting will ensure the audio isn’t “upcut” when the mic is first used after being gated.

DMX APPLICATIONS

The DMX console is most often used in an air studio to control the audio going to a transmitter and/or streaming to the Internet. In almost all cases, the PGM 1 bus is used to feed the air chain and hence has the air signal. PGM 2 can simultaneously be used to create a different console mix for a syndication feed, to feed a recorder or a mic skimmer, or to go to a line output to feed a video camera or a personal recorder.

When the console is used in an air studio, and there's no talk studio associated with the console, the Studio monitor controls can be "repurposed." One typical use is to control the monitor source feeding the guest headphones in the control room to allow the board operator to change their own monitoring source without affecting the guests' headphone feed. In this case the studio monitor output would feed the guest headphones either through Talent Stations (Figure 4-15) or a headphone amp system. The studio monitor output would be set to a fixed level (the Studio Monitor level control would thus be inactive) but the studio monitor controls would still be used to select the monitor source for the guest headphones. This allows the board operator to freely switch between control room monitor sources without affecting the audio going to the guest headphones. It also gives the board operator the ability to talk into the guest headphones using the Studio TB (studio talkback) button.

Figure 4-15 TS-4H Talent Station

The Studio monitor output could alternately connect to a newsroom, a sports bullpen, or other location which doesn't have a console, but which needs to go live-to-air. The mics in that location would be set as "studio mics" so they mute the studio monitor output while on. Their monitor source would typically be "Pre-Delay." The board operator can then talk to the talent in the newsroom, sports, or other location using the Studio TB button. If studio monitor speakers are not required, then the studio output could be used to feed a headphone amp. In this case the mics would not be set to mute the Studio output.

Network Origination

Flagship stations for sports teams often need to have a second output from their main air signal to function as a regional sports network feed. This output is typically identical to the broadcast signal except it doesn't have local commercial spots and liners. It may also require adding special signal tones or other logic.

Creating this secondary network is easily done by using PGM 1 as your main broadcast output and PGM 2 as your network feed. If the media server channels are setup to consolidate local spots and station liners on specific playback channels then those faders can be unassigned from PGM 2, whereas the national spots playback fader(s), the game liners, and the play-by-play remote inputs would be assigned to both PGM 1 busses.

Production Usage

In a production studio, PGM 1 is typically the main record signal for an in-room PC running Pro Tools®, Audition®, VoxPro®, or other digital editing system. If a Wheatstone audio driver is installed on that PC, and that PC is connected to the Mix Engine's Ethernet switch, no analog or digital audio connection is required since the audio would connect over Ethernet for both recording and playback. Audio drivers are available to support play and record of one stereo channel up to twenty-four stereo channels of playback and recording.

Voice Tracking

Using the DMX to do voice tracking is easily done in a production room where the mic channel would be assigned to both PGM 1 and PGM 2. No other inputs on the console would be assigned to PGM 1, which connects to the recorder. The channel (or channels) with track playback would only be assigned to PGM 2. The talent then monitors PGM 2 to hear both the playback track(s) and their mic.

When voice tracking is done on an on-air console, PGM 2 and PGM 3 would be used to accomplish the same functionality as in a Production room since PGM 1 is feeding air. It's recommended that the talent mic be assigned to a second "voice tracking mic" fader, which is only assigned to PGM 2 and PGM 3. No other channels should be assigned to PGM 2, which is the voice tracking feed to the recorder. The playback audio channel(s) would then only be assigned to PGM 3. The talent monitors PGM 3 to hear track playback and their mic. If they need to go live to air while voice tracking, they just switch to monitor PGM 1 and use their normal board operator mic channel assigned to PGM 1.

Callers and Remotes

Basic caller and remote operations were covered in the *Caller/Remote Operation* section on page 47. This section presents specific settings for accomplishing tasks like answering the station hot line on the caller channel to take a song request or give a caller the sad news they're not the contest winner; to recording the caller for later broadcast because they are the contest winner; and, in the case of a talk show, to take the caller live-to-air. These same functions are also applicable to remote channels, although for the most part remotes are used to go live-to-air, but they may also be recorded for later broadcast.

If the callers will not go live-to-air, the caller channel's **PGM 1** button should be unassigned. The caller channel's **Cue** button should be lit so the board operator can hear the caller in the cue speakers and in their headphones. The board operator mic should be assigned to the PGM4/OL bus (no other channel should be assigned to PGM 4/OL) so they can carry on a hands-free conversation with the caller.

Recording Callers

If the callers will be recorded, we recommend using headphones and turning the cue speaker volume down, otherwise the board operator mic recording will also pick up the caller's audio.

Since the DMX doesn't have a dedicated call record bus you must get creative in connecting a call recorder since, ideally, you want the host mic on one channel and the caller on the other channel. The easiest method to do this is to dedicate PGM 3 for call recording.

To use the "PGM 3 method" you'll want the host mic on a second fader channel, so the original mic channel is still available to feed air. Typically, the host mic is also assigned to the adjacent channel to the phone, so typically channel 7 or 15 when the phone is assigned to channel 8 or 16. Assign the mic and phone channels only to PGM 3—making sure no other channels are assigned to PGM 3. Use **Navigator** to connect PGM 3 to the call recorder output. Use the Advanced Channel Feature **Pan** to pan the mic channel fully to the left channel and pan the phone channel fully to the right channel. The caller audio is now separated from the talent audio and each channel of the recording can be individually edited on the call recorder.

The advantage of using the "PGM 3 method" is that it allows the call recorder to connect via Ethernet, using WNIP drivers, or by using any stereo analog or digital output. It also means the call recorder can easily be set to record stereo programming using PGM 3. These "Telco Record" channel sources and pan settings can be saved as an Event to quickly reset the console for call recording by recalling that Event.

If you don't have a spare channel for a second mic, and are using an analog phone hybrid, then you could use two Y cables, one on the phone hybrid and the other on the console output that connects to the phone hybrid. One channel will be the caller audio while the other will be the bus-minus audio going back to the caller (e.g., the talent mic audio). If the call recorder's analog input is not balanced, a balanced-to-unbalanced balun, or a matchbox, will need to be used.

Going Live-to-Air

When a caller or remote will go live-to-air, their **PGM 1** and **Cue** buttons should be assigned. When their channel is off the caller/remote will hear the Off Line bus (any channel assigned to PGM4/OL minus their own channel). Typically, the board operator mic is assigned to PGM 4/OL, but a producer mic could be assigned as well to allow the board operator and the producer to talk with the caller/remote without having to press any buttons for a hands-free conversation.

When the caller/remote channel is turned on, their bus-minus audio switches to PGM 1 to hear the air signal, minus their own audio, so they can interact accordingly. If the board operator needs to talk to the caller/remote while they're on-air, they press the momentary **TB** button on the caller/remote channel to give them cues (a break is coming up, etc.). This action interrupts the return audio so using TB while live should be used sparingly, and certainly not while the caller/remote is talking.

RAZOR I/O INTERFACE

The Razor I/O Interface is a 1RU device used to connect additional peripheral equipment to the DMX console and other devices on the WNIP network. Each Razor has eight stereo/16 mono inputs and eight stereo/16 mono outputs. A Razor can be used to connect satellite receivers, tuners, phone hybrids, codecs, internet streamers, and hardware recorders.

Its input signals often appear as sources on DMX fader channels while its outputs may be connected to codecs, hybrids, and other destinations.

The Razor's front panel has the same OLED display and large knob rotary encoder as used on the DMX fader channels. Rotating the encoder switches the display between showing Input Levels, Output Levels, and two screens displaying the technical details on that Razor (Figure 4-16).

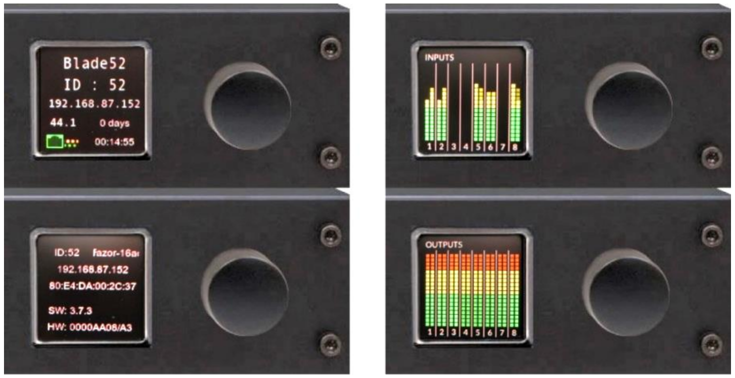


Figure 4-16 Razor Front Panel Displays

The two screens of interest to board operators are the Inputs and Outputs screens (two right screens in Figure 4-16). Unlike the DMX console meters, these displays only show average levels using vertical bar graphs since they're mainly to confirm a signal is present and its level appears "normal." Input levels should be as shown in the upper right screen in Figure 4-16. The output levels shown in the lower right screen show a 0 dBFS tone being sent to all sixteen outputs.

5 – DMX SERVICE INFORMATION

All DMX consoles are designed to yield many years of trouble-free 24/7 operation, which is why there's no power switch on a DMX Surface, Mix Engine, or Razor I/O Interface. If any DMX system component does require service, this section covers typical troubleshooting/service steps, getting replacement parts, and obtaining repair services.

PARTS AND REPAIR SERVICES

There are only a handful of field-replaceable parts on the DMX console, each of which is listed in this section. Most circuit boards and sub-assemblies are not readily field-serviceable due to the extensive use of surface-mount components and switching power supplies. The DMX Surface has the most field replaceable parts. It's recommended that the Mix Engine or Razor be returned to Wheatstone for repair.

DMX documentation: User Manual, Quick Guide, software revision information, wiring diagrams, application notes, and service bulletins, are always available for downloading from the Wheatstone web site: www.wheatstone.com. Most documentation is available in PDF file format so Acrobat Reader 6.0 or later is required.

Parts Ordering and Repair Services

Replacement parts can be purchased through your Audioarts dealer or by calling Wheatstone/Audioarts support at 252.638.7000 (New Bern, North Carolina, east coast USA time zone) or by sending an email to: techsupport@wheatstone.com.

To expedite part ordering, and to ensure the correct parts are ordered, use the Wheatstone part number when ordering. Some parts and assemblies may have long lead times, so order spares accordingly.

Any component or part returned to Wheatstone for service, exchange, or credit must have an RA (Return Authorization) tracking number issued prior to their return. Items received without an RA number written on the shipping label side of the packaging may be delayed and subject to additional handling fees.

To contact the Wheatstone Technical Support department to request an RA number, or for DMX console installation or technical support, call 252.638.7000 (8:30–5:30, Eastern time) or send an email to: techsupport@wheatstone.com.

DMX Console Parts

Wheatstone #	*Description or Use*
041390	DMX-8 Fader Panel Subassembly
041391	DMX-16 Fader Panel Subassembly
041394	DMX-8 Surface
041395	DMX-16 Surface
041396	DMX-8 expander (separate 8-fader Surface)
044300	Razor 16A with 16 Analog I/O and two mic preamps
044301	Razor 16D with 16 AES I/O
044302	Razor 16AD (8 Analog and 8 AES I/O
044303	DMX Mix Engine
083402	Complete DMX-8 console (Mix Engine & Surface)
083401	Complete DMX-16 console (Mix Engine & Surface)
059920	Control Surface Ribbon Cable
260005	HP Jack (TRS), soldered part
320003	Mic Preamp IC (SSM2019BNZ)
520149	Black Rotary Encoder Knob, 20mm
520141	Fader Knob, black
520412	Fader Knob, blue
520143	Fader Knob, green
520144	Fader Knob, gray
520144	Fader Knob, orange
520146	Fader Knob, red
520147	Fader Knob, white
520148	Fader Knob, yellow
530004	Large White Button Cap (On/Off, Soft, Timer, TB)
530396	Small White Button Cap (bus assign, cue, TB, etc.)
540043	Channel Fader (soldered part)
560002	Rotary Encoder (soldered part)
610029	Plug-in OLED Display
850088P	LN-21 Hex Driver (1/16") for control panel screws
980064	Surface In-Line DC Power Supply

DMX CONSOLE SERVICE

The DMX console contains two primary assemblies: the Mix Engine and the DMX Surface. The separate optional Razor I/O Interface, like the Mix Engine, has limited field-serviceable parts and should be returned to Wheatstone if service is required beyond replacing one of the two plugin mic preamp ICs (p/n 320003) in the Mix Engine or Analog Razor.

The DMX Surface can be powered down without affecting the Mix Engine's performance although, in most cases, this is more trouble than it's worth, so we recommend bypassing the DMX console if the fader panel must be serviced. The Surface button settings are maintained when the DMX Surface is powered down, and the Mix Engine will continue to deliver audio to the system. When the Surface is powered back up all fader positions will get updated. So, if the console is active, you must ensure the faders are set to the same positions as when the Surface was powered down, otherwise there could be large volume changes when the Surface powers up and the Mix Engine reads the fader positions.

The board operator's headphone jack can be replaced in the field. It screws into a threaded hole in the lower Surface chassis. The headphone cable goes to a 3-terminal Phoenix connector (HDPN) on the Host card (the card with the DC and Ethernet jacks). The fader panel must be removed to access the headphone jack and the rear cover must be removed to access the HDPN plug. Use an adjustable wrench to turn the square plastic jack housing to loosen it, then hand unscrew the jack. Be sure to use shrink tubing over the solder joints on the replacement jack.

Tools Required

A standard #1 Phillips screwdriver is used to remove and replace the three screws holding the rear cover panel on the meter bridge. The hex head screws, which fasten the fader panel to the chassis require a 1/16" hex driver. A small flat blade screwdriver (like a "Greenie") is needed to adjust the mic gain trim pots or to flip the phantom power switches on the mic preamps in the Mix Engine and Analog Razor.

If you're making up your own audio cables, an RJ45 crimper is required (and lots of RJ45 plugs). Having a category cable checker, with remote test plug so that already-run cables can be checked in situ, is also a must have tool.

Mix Engine Status lights

If the red ERROR LED lights up on the front of the Mix Engine, it indicates a fault with the Mix Engine's operation. Although audio may still be passing through the Mix Engine and there may be no outward signs of trouble, the error indicator is saying otherwise.

To clear the fault, first reboot the Mix Engine using **Navigator** (right-click on the Mix Engine icon in the System Dock and select *Reboot Blade*). If the Mix Engine reboots and the Error LED turns off, then the issue was resolved.

If the Mix Engine reboots but there is no change to the Error LED, the Mix Engine will have to be power cycled. Since this will cut off all audio through the console, the DMX will need to be bypassed before power cycling the Mix Engine. Unplug its AC cord and wait a minimum of ten seconds, to ensure power is fully removed from all internal circuits, before plugging the AC cord back in.

The Ethernet jacks for network connection have two integral green LEDs. Both blink to indicate Gigabit operation. If only one is blinking, that indicates the connection is working at 100MB speeds, which will occur on the DMX Surface cable and on an admin PC with a 100MB Ethernet NIC.

Swapping the Fader Panel

The Surface's Fader Panel Subassembly can be swapped out while the console is on-air since audio goes thru the Mix Engine, not the Surface. You'll lose audio control while the Surface is off-line, but it should take you less than five minutes to swap a fader panel. Here are the steps:

1. Unplug the AC cord on the Surface's DC power supply.
2. Use a 1/16" hex driver to remove the hex screws from around the fader panel (12 on a DMX-8, 18 on a DMX-16).
3. Lift the front edge of the fader panel and unplug the keyed ribbon cable(s) that connect the fader panel boards to the Host Card. There's one cable on a DMX-8, two cables on a DMX-16.
4. If the fader panel is being replaced "hot" make sure the faders on the replacement panel are set to the same positions as on the original panel. This is not important when the console is taken off-air, just move all faders to full off.
5. Set the replacement panel onto the rear support rail and plug in the keyed ribbon cable(s).
6. Set the panel into place on the chassis and fasten it using the hex screws removed in step 2.
7. Plug in the power supply's AC cord to power up the Surface.

The new fader panel will take about a minute to boot up. The same audio sources, assignment button settings, and monitor settings as the original panel had will appear on the replacement panel. If the faders were set as on the original panel, and the DMX console has continued playing on air, there will be momentary audio level changes as the new Surface takes control—especially when EQ or Dynamics are active since there will be a couple seconds where the audio will be flat and then switch to use the EQ and Dynamics settings on any channel that's currently turned on.

A stand-alone DMX console can be networked with up to three other WNIP-compatible devices (PC or audio server, Razor I/O Interface, Blade, talent station, etc.) using the five-port gigabit switch in the Mix Engine. To network additional WheatNet-IP (WNIP) devices or if you want to network your DMX with additional WNIP consoles, one or more Wheatstone-approved gigabit Ethernet switches must be used to support the additional audio streams propagated over the WNIP network.

DMX MIX ENGINE'S ETHERNET SWITCH

The Mix Engine's gigabit Ethernet switch (Netgear GS105Ev2 5-port) is factory configured for stand-alone operation so if the DMX will not be connected to an external switch you don't need to edit any configuration settings. But, when the DMX will be networked with other WNIP devices, which means it will need to connect to a WNIP system managed switch, so the Netgear configuration settings may need to be changed to stream audio between the built-in switch and the managed WNIP system switch.

Editing the switch configuration is done using the Netgear ProSafe Plus Utility, v2.7.2 (link to ProSafe Plus Utility installer). It can be installed on the admin PC that's running the Navigator and DMX Surface Setup apps. When the ProSafe Plus Utility starts up it searches for and displays any Netgear switches it finds. The IP address assigned to each Netgear switch at the Wheatstone factory is 192.168.87.239.

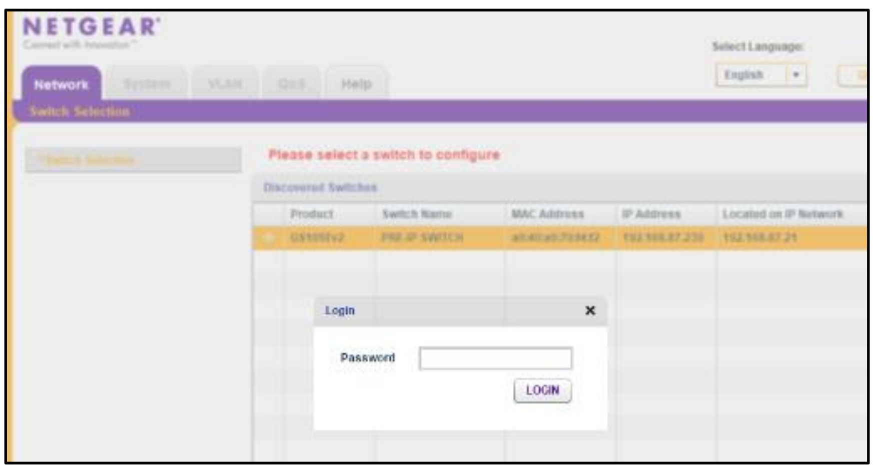


Figure A-1 Netgear Login Page

If you are networking multiple DMX consoles you may need to connect directly to each Netgear switch to change its default IP address since all have the same IP address. When you highlight the switch row, as shown in Figure A-1 and click *Apply*, a log-in window opens. The factory set password is *password*.

After entering the password, the *System > Switch Status* page is displayed. Click on *Multicast* then click *IGMP Snooping* to show the **IGMP Snooping Configuration** shown in Figure A-2. Confirm the settings for *IGMP Snooping Status* is **Enable** and the *IGMP Snooping Static Router Port* is set to 01 (Port 1 on the Mix Engine switch). This sets up Port 1 to network the DMX switch to an **Access Port** on a managed gigabit system switch. Since the Netgear switch is an unmanaged switch it will not work correctly if connected to a Trunk port.

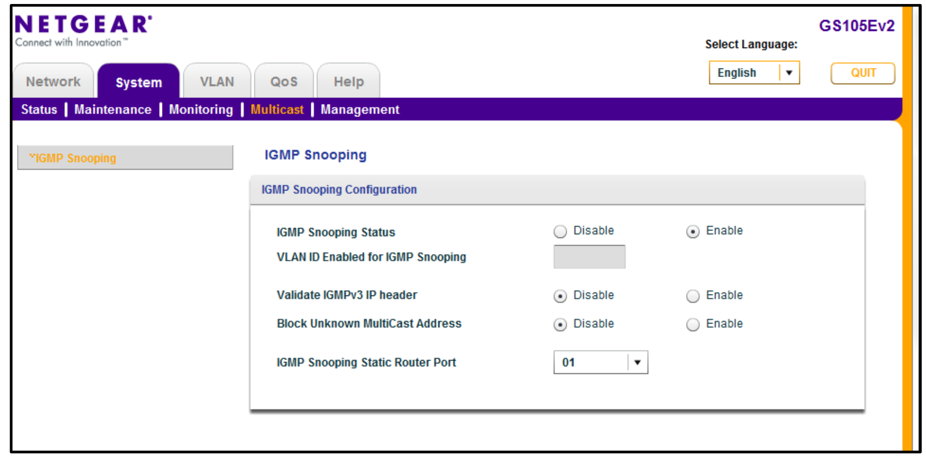


Figure A-2 Netgear Switch System Management Page

IP Addresses & Device ID

The default IP addresses assigned to "factory-fresh" DMX consoles and Razors should be reserved so that new devices can be networked without causing any IP addressing conflicts. The default settings can be edited, as required, to fit into your system using Navigator. Here are the default addresses assigned to the various DMX devices at the factory:

192.167.87.50 (Razor-16A, Device ID 50)
192.167.87.60 (Razor-16D, Device ID 60)
192.167.87.70 (Razor 16-AD, Device ID 70)
192.168.87.101 (Mix Engine, Device ID 1)
192.168.87.201 (DMX Surface)

We recommend that prior to networking any devices, that a system-wide IP address table be created using a spreadsheet or a handwritten list to record every device that is connected—or will connect in the future, to your WNIP network: Mix Engines, Surfaces, Razors, Blades, PCs running

Navigator or other apps, PC Blades running the WNIP audio driver, AES67 devices, talent stations, source selector panels, etc. Each device must be assigned a unique IP address since a WNIP network, just like every AoIP network that uses multicast, does not work with nor support DHCP.

When multiple DMX consoles and Razors are ordered at the same time you can have the factory assign unique ID# and IP addresses to those units so they can be networked without needing to change any ID# or IP addresses. However, when a single unit is ordered it will arrive with the default settings listed on the previous page.

If you already have devices in the system using the factory default settings, you'll need to use Navigator to factory reset each of the new devices so you can assign a new unique ID# and IP address prior to networking the new device with your existing system. You could use the Mix Engine's built-in switch to create a stand-alone network to connect the devices you want to configure and a PC running Navigator.

To factory reset a device, first click on it in Navigator's System Dock to select that device, then view the Blade Admin tab, as shown in Figure A-3. The Factory Reset option must be enabled, so click the checkbox to activate the Full Blade Reset button. You will get a warning popup about enabling this option since when you click *Full Blade Reset*, that device's settings will be deleted, and the Razor or Mix Engine will then reboot.

Figure A-3 Factory Resetting a Device

After a minute or so, the device will reappear in the System Dock under the Detached Blade category as shown, and outlined, in Figure A-4. Right-click on that detached device icon to display a context menu. Select *Initialize Blade…* to open the Initialize window where you then assign the Device ID#, the Signal Template (normally Stereo I/O), the Blade name, its Location, and its IP and Gateway addresses.

Clicking *Configure Blade* uploads those settings to that device and closes the window. The Mix Engine or Razor icon will disappear as it is again rebooted. In a minute or so it will reappear in the System Dock with the factory default settings incorporating the blade ID you selected into the signal names. So, if you assigned it as Blade 5, then the source and destination names will each begin with BL005.

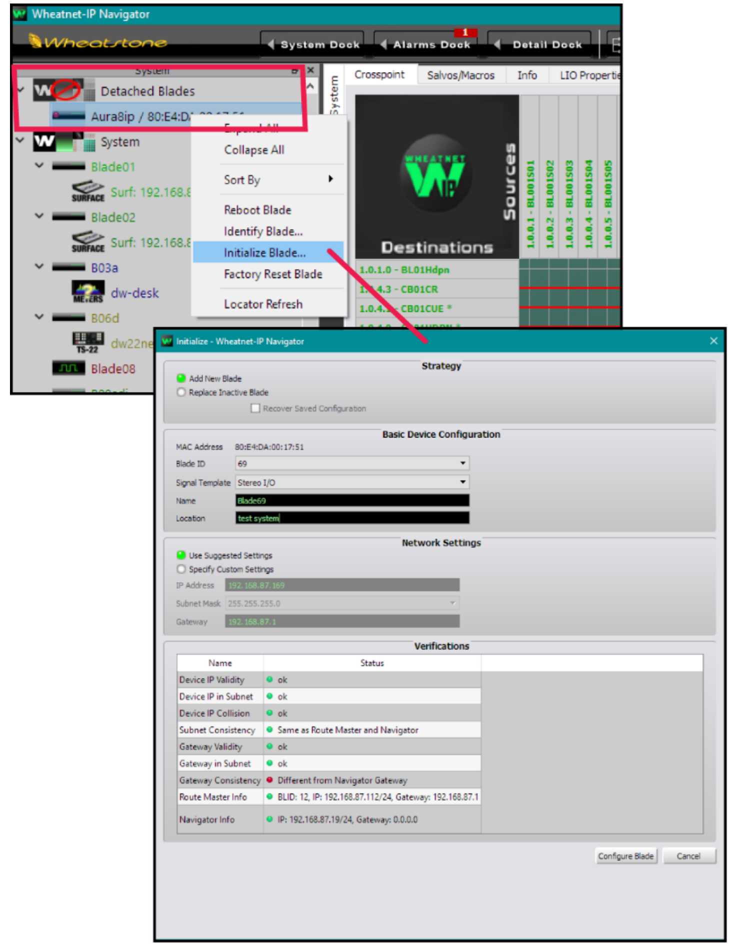


Figure A-4 Initializing the Device

Alternately, if you've already edited signal names and/or added logic settings to your signals and don't want to do a factory reset, you can manually change the Device ID, blade name, and IP address of any Blade by using the Locator tab, in either Navigator or the DMX Surface Setup app. Since changing these settings will affect system operation, you'll want to edit these settings prior to releasing the device for daily operation.

To view the Locator tab in Navigator, the Navigator Preferences option *Enable Advanced Controls* must be checked. Select the device you want to edit by clicking on its row in the top part of the Locator tab (see A-5 on the next page). If a new device is not listed, click *Refresh* to rescan the network. Clicking on a device highlights its row (a Mix Engine is highlighted in Figure A-5) and its current properties are shown in the Set Selected Device Properties pane.

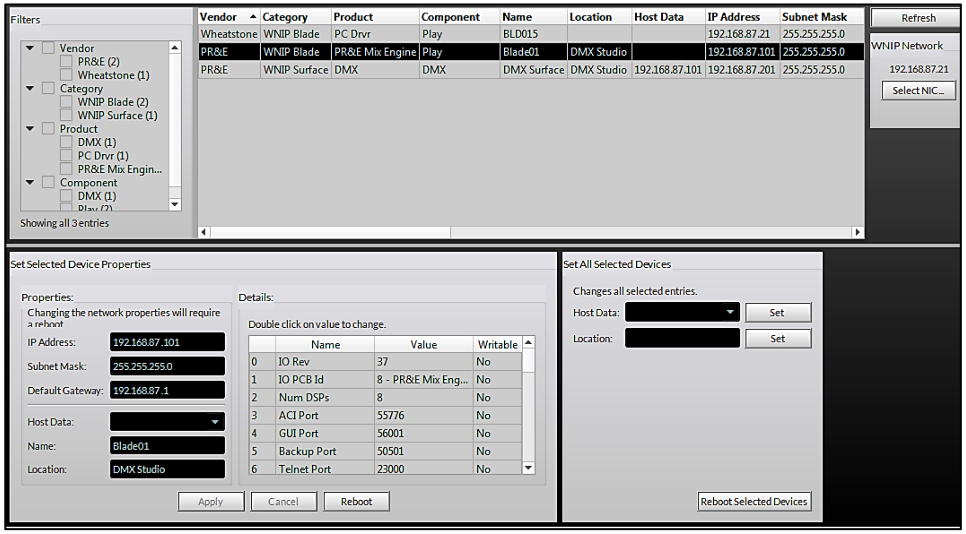


Figure A-5 Setting Device Properties in the Locator Tab

In the *Properties section* the assigned IP address, Name, and for Razors, the Host Data (Host IP address) can be edited. For a Mix Engine, Razor, or a Blade, the assigned Device ID Number can be edited in the Details section. Once the settings are edited click *Apply* updates the device. For most settings the device must then be rebooted to incorporate the new settings, so click *Reboot* to restart that device to uses the new settings.

Typically, you want the IP address and ID# for each device to be related. For example, setting a Blade to ID9 means its IP address is typically set to 192.168.87.109. If that's a Mix Engine or a Console Blade, its Surface's IP is typically related, so for Blade 9 its Surface IP address would typically then be set to 192.168.87.209.

Razors and I/O Blades can be set for sequential ID numbers and IP addresses, so Blade 51 could be assigned to IP address 192.168.87.151. Blades are typically sequentially numbered up to ID99 and IP address up to 192.168.87.199, skipping over the default IP addresses.

PC Blades (PCs running the WNIP audio driver) are typically assigned IP addresses starting at 192.168.87.100 and moving down. Ethernet core and edge switches (talked about in the Multi-Station Expansion section) are typically assigned IPs from 192.168.87.1 up to 192.168.87.20 reserving 192.168.87.11 for the PC running Navigator and the DMX Setup app. Talent stations and other accessories would be assigned IP addresses above the switches and below the Blades and PC Blades (e.g., in the .30 to .70 range).

Note: On larger systems, like those with more than about 30 consoles, the IP addresses assigned to the Mix Engine's internal Ethernet switches should not use the .240 to .254 range. Rather they should be assigned along with the network switches, using the .1 up .20 range skipping over .11 assuming it's assigned to the PC running Navigator and the DMX Setup app.

Networking Multiple DMX consoles

The DMX Mix Engine's switch works fine to network one DMX console with three other WNIP devices. But one cannot directly connect a DMX Mix Engine switch port to another DMX Mix Engine switch port. To network two or more DMX consoles or other WNIP consoles together requires adding a managed gigabit switch which is setup to function as the WNIP system's IGMP Querier to control the flow of multicast traffic through the system.

WNIP NETWORK EXPANSION

Here's an overview of what's involved in building out a WNIP network, starting with one stand-alone DMX console and adding one or more Gigabit Ethernet switches configured to support networking WNIP devices.

Just about any Ethernet gigabit switch can be configured to support multicast audio streaming. The switches used on a WNIP system are generically called AoIP (Audio over Internet Protocol) switches to differentiate them from your facility's switches that network your facility PCs, servers, printers, phone system to one another and to your internet access point. AoIP switches and the facility's Ethernet switches may be the same model since it's the switch configuration settings which set whether a switch is suitable for AoIP traffic or is intended for standard Ethernet traffic. Thus, the AoIP switch definition is arbitrary and only refers to a switch that can be specifically configured for multicast streams.

For most installations this means using separate switches to "air-gap" traffic on the AoIP network from the facility's Ethernet network. One could use multiple large Ethernet switches, configured into multiple VLANs or virtual networks, to isolate the AoIP traffic from the facility's Ethernet traffic, but creating VLANs is not covered in this manual since we whole-heartedly recommend using separate AoIP switches to create an isolated WNIP network. Doing this simplifies switch configuration and makes network troubleshooting a whole lot easier down the road.

Creating a Small WNIP Network

Once the built-in switch in the Mix Engine has five devices plugged into it, the only way to add additional WNIP devices is to add a Wheatstone-recommended AoIP switch and then connect the built-in Mix Engine switch (after verifying the switch is setup for IGMP Snooping, as covered on page 61) from Port 1 to an **Access Port** on one an AoIP switch.

Various Ethernet switch models from Cisco and HP have been used in WNIP systems over the years, but their model numbers change on a regular basis, so it's best to visit the Wheatstone web site (click on Support & Downloads then select Compatible switches for WheatNet-IP) or else call or email Wheatstone tech support to get a list of the currently recommended switches.

One current Cisco switch (as of December 2021) that we can recommend are the 2960CX-series of layer 3 managed switches. The 2960CX-8TC-L eight port switch can be used as a studio edge switch or even as a core switch for a small system. For larger systems, the 2960CX-24 twenty-four port switch can be used as an edge switch to combine multiple studios together so one CAT6 cable can be run back to the Technical Operations Center (TOC). That model can also serve as the WNIP system core switch to connect various smaller edge switches with the Blades and other networked equipment in the TOC. A configuration sheet on these switches can be downloaded from the Wheatstone web page: compatible-switches-for-wheatnet-ip.

For a single station facility with two DMX consoles in an on-air studio and a production room, several Razor I/O Interfaces, and two PC Blades for your audio servers, a small eight-port switch may be all that's needed beyond the ports on the Mix Engine switch. Two in-room WNIP devices like a Razor I/O Interface, a Mic Processor Blade, a Talent Station, or a PC Blade can plug into the built-in switch in each studio, using Ports 2 and 3, reserving Port 1 on each Mix Engine to connect to your main AoIP switch that's typically located in the TOC.

Note: A PC Blade is a PC that's running a licensed copy of the WNIP audio driver to play and/or record audio over their WNIP network connection. Audio drivers, to support from 1-channel up to 24-channels of stereo playback and recording can be purchased through any authorized Wheatstone or Audioarts dealer.

An 8-port or larger AoIP switch is typically located in the TOC along with your I/O Blades or Razor Interfaces and your audio playback PCs/servers since it takes a single CAT6 cable, from Port 1 on each Mix Engine, to connect each studio to an Access Port on the system's central AoIP switch. The Razors, Blades, and PC Blades in the TOC connect to other ports on the main AoIP switch, again using one CAT6 cable per device.

If the on-air studio has a separate talk studio, a small AoIP switch could be placed in that studio to network that studio's WNIP devices and to connect that studio to the main switch. If there are four mics in the talk studio, an M4IP-USB Blade could be installed since it has four mic preamps along with four USB ports for connecting laptops for playback or recording. If a Talent Station is used at each mic position for mic control and to add a headphone jack, then an eight-port switch would be used in that room.

Since Talent Stations can be powered by PoE (Power-over-Ethernet) using a switch that supports PoE is recommended for talk studios since that means the talent stations only need a single CAT5 cable connected to them—no wall-wart supply required.

Multi-Station Network Expansion

When two or more stations share a single facility, we recommend you follow the small WNIP network expansion model for each station where that station's studios connect to one "Edge Switch," typically located in that station's main on-air studio. Each station's main edge switch then connects to a "Core Network Switch" located in the central rack room/TOC. Depending upon how many stations, and how much TOC equipment is being connected, the Core Network Switch might physically consist of a couple medium-sized (24-port) network switches rather than a single large 48- or 96-port switch. Standardizing on one size or type switch means you'll only need one type of "on-the-shelf" backup switch.

If a multiple station facility is being planned, Wheatstone offers system configuration and system programming to create salvos, scripting for programmable buttons, etc. These engineering services are charged on a per-studio or on a per-hour basis. Other services, including factory proof of performance as well as on-site commissioning and training are also available. These services can all be arranged through your dealer or directly thru Wheatstone sales.

Wiring Practices

Since most studio wiring is now using straight-thru four-pair category (i.e., CAT5, CAT5e, CAT6) cables, we recommend taking advantage of the available category cable jacket colors to differentiate the cables carrying various system signals. This can both simplify installation and ensure future system troubleshooting is also a bit easier.

Here are our recommended cable jacket colors, by signal type:

Orange	Blade-to-Access Port cables
Green	PC Blades-to-Access Port cables
Yellow	KVM-over-IP cables
Blue	Facility IT Ethernet cables (since they're probably already in place and are probably blue!)
White	VoIP phone system cables
Black	Audio and logic wiring (Blade I/O to/from peripheral audio and logic devices)

Most users have standardized on an automation server company—most of whom offer WNIP support for SLIO (Software Logic I/O). The automation servers are typically located in the TOC since it's climate controlled and server noise is not an issue. KVM IP extenders are available from various vendors, like AdderLink, to allow using a single category cable to directly connect the server to a studio-located monitor, keyboard, and mouse over cable runs of up to 160 feet (48 meters).

Signal Connection Control

Most users have several types of codecs which may be shared between studios or even between different stations in a cluster. These codecs might include older ISDN interfaces along with newer Tieline and Comrex codecs, but no matter what type of codec, hybrid or other two-way device is used, Navigator's *Associated Connections* feature can be used to automatically connect a bus-minus or a mix-minus signal to each codec when it's taken on a console to simplify codec connection.

The only downside of setting up automatic bus-minus or mix-minus return signal connections on shared codecs is that this can lead to a user changing the return feed to a codec that's still actively in use. To prevent this from occurring, many users opt to force codec users to physically go to a rack selector or to a touchscreen, often mounted in a hallway outside TOC, where they must identify whether a codec they want to use is actively being used and, if not, to then connect the codec to their studio.

This "Public Codec Rack" is typically setup with a touchscreen monitor running ScreenBuilder (a Wheatstone scripting app configured using a GUI interface) which is setup so that any user can quickly switch codec routing without needing to use Navigator's Crosspoint grid, fire off a system salvo, or use an Associated Connection to switch the return connections for a shared codec.

The ScreenBuilder app is a very powerful tool that can be licensed to run on one or multiple PCs. The app uses standardized on-screen elements like faders, meters, labels, buttons, clocks, timers, and other widgets to physically control WNIP network devices. Arranging these on a PC screen allows one to create custom control panels with quick-access buttons to monitor and control codecs, to setup recorder feeds, and to control sources feeding your Internet streams.