




mixIT

Logitek mixIT Audio Console System Setup Manual
Logitek JET67 Audio Engine Manual


Logitek

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Logitek Electronic Systems, Inc.
5622 Edgemoor Dr.
Houston, TX 77081
United States of America

Tel (713) 664-4470
Web logitekaudio.com
Email support@logitekaudio.com

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Chapter 1 – Introduction



Figure 1 - The mixIT-12 console.

Thank you for purchasing a **mixIT** Audio Console System. The **mixIT** is an affordable, easy to use Audio over IP system. With support for **Ravenna**, **Livewire+**, and **AES-67** AoIP built in and **Dante** support available as a plug-in option, the mixIT is easy to integrate into an existing AoIP facility or the perfect introduction to the world of AoIP.

The **mixIT** is powered by the **JET67 Audio Engine**. The JET-67 is where all of the audio and network connections are made. It also delivers power to the audio console. We designed the JET67 to have sufficient inputs and outputs to connect most studio equipment for the room based on fader size.

A 4 port Ethernet switch is built into the JET-67. It has two 1GB ports and 2 10/100 ports. The Ethernet connections are used to configure the engine and to connect Audio over IP devices.

Logitek customers with older JetStream Mini and JetStream Plus Audio Engines can also use the mixIT console in place of the JET-67 with a special power supply available from Logitek. The setup of these systems is covered in a different manual.

The mixIT console is not compatible with the Logitek AE-32 Audio Engine.

Model Numbers



Figure 2 - The mixIT-12 Console

mixIT-12

The **mixIT-12** console is a kit that contains the following:

MIXIT-C12 mixIT 12 Channel Surface

JET67 JET67 Audio Engine with 2 J67-I/O cards installed.

This system contains:

- 12 fader touchscreen controlled audio console
- 4 microphone inputs with phantom power
- 6 stereo analog inputs
- 2 digital inputs
- 8 stereo analog outputs
- 2 digital outputs
- 4 port Ethernet switch (2 x 1GB/2 x 10/100)



Figure 3 - The mixIT-6 Console

mixIT-6

The **mixIT-6** console is a kit that contains the following:

MIXIT-C6 mixIT 6 Channel Surface
 JET-67 JET67 Audio Engine with 1 J67-I/O cards installed

This system contains:

- 6 fader touchscreen controlled audio console
- 2 microphone inputs with phantom power
- 3 stereo analog inputs
- 2 digital inputs
- 4 stereo analog outputs
- 2 digital outputs
- 4 port Ethernet switch (2 x 1GB/2 x 10/100)

JET-67 DANTE

This is a card that plugs into the main DSP board of the JET-67 and adds 64 x 64 channels of Dante networking in addition to the AES-67 networking built into the JET67.

Practically speaking, it's more audio networking than you'll ever need for a 12 or 6 fader console, but that means you'll never run out of networking power, either.

If you did not order the JET-67 DANTE card when you bought your system and decide to add it later, it can be installed in the field. The only tool you will need is a screwdriver.

About Logitek

Founded in 1979 by two Rice University students, Logitek has made audio consoles for television and radio for more than 40 years. After helping found the campus radio station and installing a donated audio console at KTRU, the pair decided they could design a better one themselves and the company was born.

In the late 1990s, Logitek was the first North American company to offer a router-based digital console. Its first virtual PC-based console was introduced in 1999. By 2009, the company had pivoted to television, offering both physical and virtual consoles that seamlessly interface to production workflow automation systems.

mixIT represents Logitek's return to its radio roots, offering a simple user interface, modern control, and the latest in standards-based networking technology at an attractive price.

Our vision for Audio over IP is simple: we believe that over time, every piece of audio equipment will have an Ethernet jack on it, and it should be as easy to connect something to the audio network as it is to plug an XLR into an analog or digital board.



Chapter 2 – System Configuration



Figure 4 - Front view, JET67 Audio Engine

With its silent, fanless operation, the JET-67 Audio Engine is designed to be installed in the studio with the audio console.

System Requirements

The following is necessary to configure your system:

- A computer running Windows 10
- An Ethernet adapter to connect your computer to the JET67
- JetStream Server, available at <https://logitek.link/XXXXXXX>
- A web browser, such as Chrome, Internet Explorer, or Microsoft Edge. Firefox is not supported.

Connecting the Console to the JET-67 Audio Engine

Using the supplied standard CAT-5 cable, plug one end into the RJ-45 port on the back of the JET-67 and the other end into the RJ-45 port on the back of the console.

While we use a standard Ethernet cable for this connection, this is NOT an Ethernet connection, so don't plug it into an Ethernet switch! The console and JET-67 communicate via a RS-485 serial connection. We are also sending power on one pair of wires and AES-3 digital audio on another pair of wires to drive the cue speaker.

We recommend CAT-5 for this connection to adequately provide power to the console. CAT-6 and/or thin ribbon network cable generally use thinner wire which won't work for this purpose. Good old-fashioned CAT-5 or CAT-5e will always work, even at a distance of 25 feet.

Audio Connections Overview

The JET67 uses quick-connect Phoenix-style terminal connectors. Insert tinned wire into the slot and tighten the screw to secure the wire. Each terminal block has a positive terminal, negative terminal, and common ground.



Figure 5 - rear view, JET67 for mixIT12

The JET67 works best with balanced audio. If you have an unbalanced input, we recommend jumpering negative to ground.

Unlike a traditional audio console, the mixIT controls an audio router. Therefore, you'll notice there isn't an output that's labeled "program" or "monitor." You decide where you want to wire your inputs and outputs and then configure everything in the software.

Instead of having an A/B switch over the fader, you'll choose in the software what sources are allowed on each fader. You can have as many or as few choices as you'd like.

In the software, you will choose how many channels and input or output has (stereo or mono). You can split a stereo pair into two mono channels on either an analog or digital input or output.

Note that if you route a stereo mix to a mono output, the JET67 will NOT sum the two channels; you will hear left channel only. (However, if you're wiring up an AM radio station, we do offer a mono program mix output so we can sum program for you.) If you route a mono input (such as a microphone) into a fader, the JET67 will copy the left channel to the right so the input will feed both left and right.

Connecting Your Computer To The JET67

Overview

We use a program called JetStream Server to make an IP connection to the DSP card inside the JET67 to configure the engine.

JetStream Server, in turn, serves web pages to your web browser to enter information and upload to the engine.

Your configuration is stored in memory on the JET67. After you have configured the JET67 you can disconnect your PC and the audio engine will continue to control the computer. It is not necessary for JetStream Server to be connected to operate the console.

The configuration will remain in memory on the JET67 even if power is disconnected.

You can save the configuration in a JET67 to an .XLS file and keep it backed up and you can restore that .XLS file to the JET67 using JetStream Server.

Install JetStream Server

Connect Your PC to the JET67

1. Connect the mixIT directly to the JET67 jack marked “Console” with the supplied CAT-5 cable.
2. Connect the JET67 to power if not already plugged in.
3. Plug one of the Ethernet ports on the back of the JET67 to your local network OR plug an Ethernet cable from your PC directly into one of the Ethernet ports on the back of the JET67. The JET67 is set up by default to receive an IP address automatically via DHCP. If no DHCP server is present, your PC and the JET67 will automatically select IP addresses within the same subnet via AutoIP.
4. On the console touchscreen, tap the word “mixIT” next to the headphone selector and make note of the IP address. This is the IP address assigned to the JET67.
5. Launch JetStream Server.
6. After startup completes, go to the **External Comm** tab.

Logitek JETSTREAM SERVER 5.2 JetNet 3 LOG-ENG-D-01 Exception 12:10:21 PM

External Communications

Auxiliary Com Ports
☐ Aux Ports Active Lowest Aux Port: 0 Number of Aux Ports: 0

Source Com Port	Connected Destinations	Baud Rate	Data Bits	Stop Bits	Parity	Source Description

vGuest Com Ports
☐ vGuest Ports Active Lowest vGuest Port: 0 Number of vGuest Ports: 0

vGuest Com Port	vGuest Protocol	Baud Rate	Data Bits	Stop Bits	Parity	vGuest Profile	vGuest Description

JetStream Com Ports

Com Port	Status	Connected To
1	COM 1	Open
2		
3		
4		
5		
6		
7		
8		

External Protocols
 Protocol Name: **Logitek LP2**

Set up TCP / IP Com Port

Surface Number: 1
 JetStream First Fader: 1
 Number of Faders: 6
 Master Fader: None

Connection Enabled:
☐ as TCP/IP Client
☐ as TCP/IP Server
☐ Com Port

☒ Show in Network Log
☒ Show ExtPro in Log

Figure 6 - JetStream Server External Protocols

7. In the lower right corner, locate External Protocols.
8. Change the Protocol Name from None to Logitek LP2.

- Click on the TCP/IP tab.

The screenshot shows the 'External Protocols' window with the 'Logitek LP2' protocol selected. The 'TCP / IP' tab is active, showing the 'TCP/IP Client' section with 'Remote IP Address' set to 192.168.3.20 and 'Remote TCP/IP Port' set to 9500. The 'TCP/IP Server' section shows 'Local TCP/IP Port' set to 8003. The 'Set up' and 'Com Port' tabs are also visible.

Figure 7 - JetStream Server, External Protocols, TCP/IP tab

- Under TCP/IP client, type the IP Address of the JET67 from step 4 in the Remote IP Address field. Set the Remote TCP/IP port to 9500
- Return to the Set up tab.

The screenshot shows the 'External Protocols' window with the 'Logitek LP2' protocol selected. The 'Set up' tab is active, showing fields for 'Surface Number' (1), 'JetStream First Fader' (1), 'Number of Faders' (6), and 'Master Fader' (None). The 'Connection Enabled' section has checkboxes for 'as TCP/IP Client' (checked), 'as TCP/IP Server' (unchecked), and 'Com Port' (unchecked). The 'Show in Network Log' and 'Show ExtPro in Log' checkboxes are also checked.

Figure 8 - JetStream Server, External Protocols, Set up tab

12. Check the box for Connection Enabled as TCP/IP Client. A green box will appear on the TCP/IP tab showing that the connection is good. (If you run into any issues, you may need to open the Windows firewall to traffic on TCP port 9500.)
13. Open your web browser. (Any modern browser except for Firefox will work.)
14. Enter 127.0.0.1 into the address bar of your browser and hit enter.
15. You will see the login screen. For username, enter user. Leave the password blank. Click Connect.

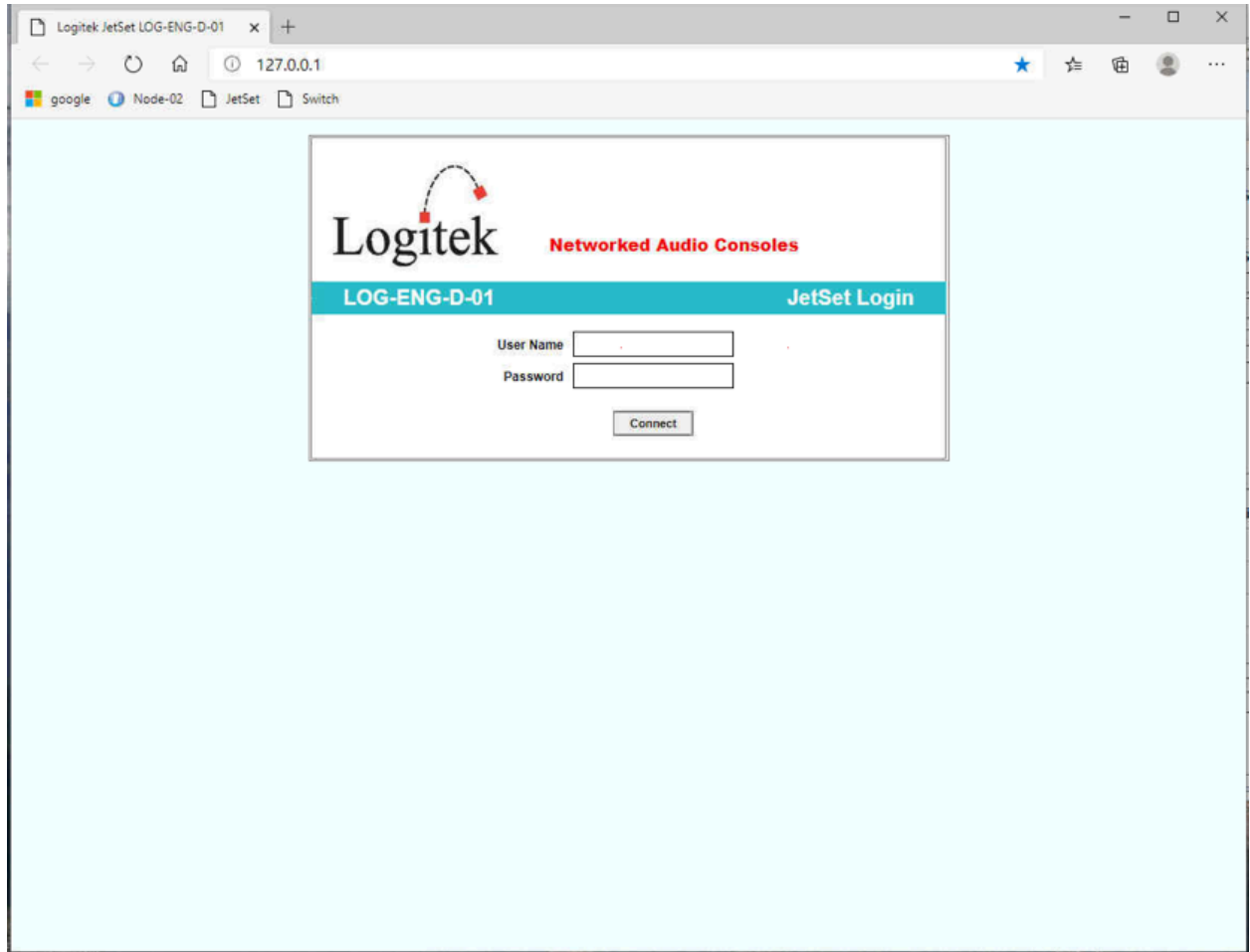


Figure 9 - The login screen

16. You will be on the Local IO Page.

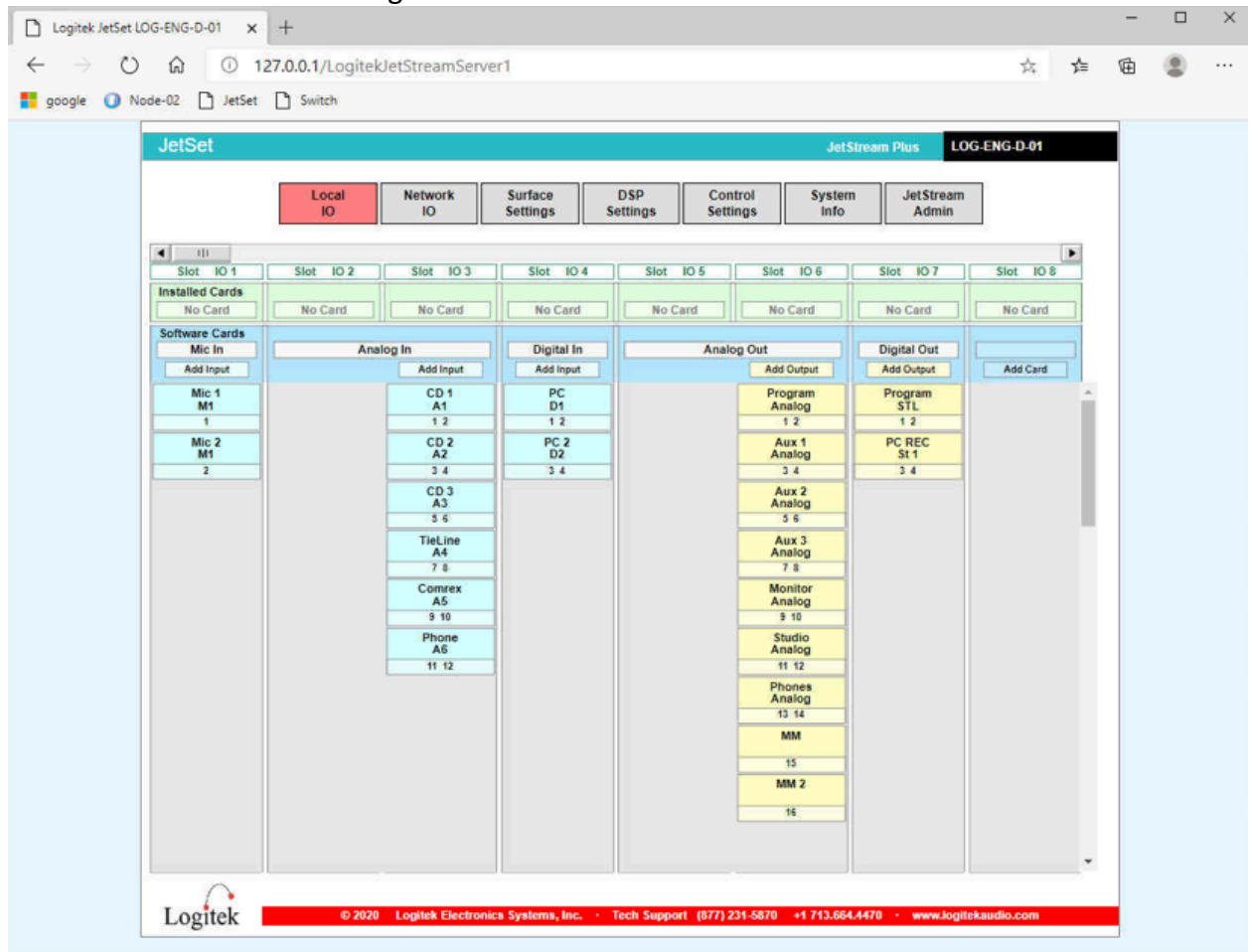


Figure 10 - The Local IO page

The first column is the microphone inputs. The second is the analog inputs. The third is the digital inputs. The fourth is the analog outputs. The fifth is the digital outputs. To add an input or output, click the 'Add Input' or 'Add Output' button over the respective column. To edit an input or output, click on the box for the desired input or output.

Creating or Editing a Microphone Input

To create a new input, click Add Input in the Mic In column. To edit an existing input, click the box for the input. You will be taken to the input editor.

Figure 11 - The Local Input editor

You will need to fill in the following information:

1. The **Surface Label** is what the operator will see on the console. There are two fields of 8 characters for the top and bottom line on the display.
2. Enter a **Unique Name** to identify the input.
3. Set the **Number of Channels** to 1.
4. Set the **Lf** channel to the input number. Valid numbers on a mixIT-12 are 1-4. Valid numbers on a mixIT-6 are 1-2.
5. If you are using a GPI Input to turn the microphone on or off, enter the GPI input number in the start or stop box. If you are using a momentary cough switch, enter the GPI input number in the **Cough** box. Valid numbers are 1-4.
6. For a tally light that will turn on and off **ONLY** for this microphone (example, a tally light ring on a microphone arm) enter the GPI Output number in the Audio Tally box. Valid GPI output numbers are 1-8.
7. Assign the microphone to a **mute group** if desired to set up studio muting and a on air light tally that will turn on when the first microphone turns on and off when the last

microphone turns off. (example: to put all microphones in the first **mute group**, put a 1 in the **Mute Group** box for each microphone.) To assign the bus to be muted and the GPI outputs to the **Mute Tally**, you will configure the **Mute Group** under **Control Settings**.

8. To adjust the gain of the mic preamp, enter the value in DB in the IO Gain box. Negative numbers reduce gain, positive numbers increase it. **The valid range is xxx**
9. To enable Phantom Power for the input, set Phantom Power to Yes.
10. To send the audio from this input to the audio network so other AES-67 devices can use it, check Yes for **Send to Network**. Set **Network Latency** to Low. The JET67 will stream this source to the network and announce it to other devices using the surface label that you entered.
11. Click **Upload to JetStream** to save your changes.
12. To delete this source, click **Delete this local input**.
13. To cancel and return to the Local IO page, click **Cancel edit changes**.

Creating or Editing an Analog Input

To create a new analog input, click Add Input under the Analog In column. To edit an existing input, click the box for the desired input to open the editor.

Figure 12 - The Local IO editor for an analog input

You will need to fill in the following information:

1. The **Surface Label** is what the operator will see on the console. There are two fields of 8 characters for the top and bottom line on the display.
2. Enter a **Unique Name** to identify the input.
3. Set the **Number of Channels** to 1 for mono, 2 for stereo.
4. Set the **Lf** channel to the left analog input number and the **Rt** to the right analog input number. (if mono, there will only be the Lf box to fill.) Valid numbers on a mixIT-12 are 1-12. Valid numbers on a mixIT-6 are 1-6.
5. The **Mode** setting determines how the fader will handle the audio when the source is routed to the fader. The choices are:
Default: if this source is two channels, it will be stereo. If this source is one channel, it will be mono.
Mono: if selected, a two channel source will be summed when it is routed to the fader.
Phase: phase reverse a two channel source

- LL: send the left channel only to both left and right
- RR: send the right channel only to both left and right
- Lx: left channel on left, silence on right
- xR: silence on left, right channel on right
- 5.1: this function is not supported by the JET67
- 6. If you are using a **GPI Input** to turn the fader on or off, enter the GPI input number in the start or stop box. The Cough box will mute the input when that GPI input turns on. Valid numbers are 1-4.
- 7. For **GPI Outputs**: Audio Start pulses a relay when the channel turns on. Audio Stop pulses a relay when the channel turns off. Audio Tally latches a relay on when the channel turns on and turns the relay off when the fader turns off. Valid numbers are 1-8.
- 8. **UDP Control**: This sends ASCII text via UDP to the specified IP address using the specified port, usually for automation control. The **Start String** is the text sent when the channel turns on and the **Stop String** is the text sent when the channel turns off.
- 9.
- 10. To adjust the gain of the input, enter the value in DB in the **IO Gain** box. Negative numbers reduce gain, positive numbers increase it. **The valid range is xxx**
- 11. **Mix Minus Bus**: type in the number of the mix minus bus to have this input removed from the mix. (Example: if mix minus 1 is derived from program, then mix minus 1 will be program minus every input tagged with a 1.) More information about mix minus is found in DSP Settings, Mix Minus.
- 12. To send the audio from this input to the audio network so other AES-67 devices can use it, check Yes for **Send to Network**. Set **Network Latency** to Low. The JET67 will stream this source to the network and announce it to other devices using the surface label that you entered.
- 13. Click **Upload to JetStream** to save your changes.
- 14. To delete this source, click **Delete this local input**.
- 15. To cancel and return to the Local IO page, click **Cancel edit changes**.

Creating or Editing a Digital Input

Digital inputs are handled exactly the same as analog inputs in the editor. The only difference is in the channel assignment.

AES audio carries two channels on one wire. Therefore, to assign a stereo input on AES input #1, assign left to channel 1 and right to channel 2. To assign a stereo input to AES input #2, assign left to channel 3 and right to channel 4.

You can also split the mono channels into separate inputs. (For example, a satellite receiver might have separate programming on left and right channels or a AES mic preamp might have microphone 1 on left and microphone 2 on right.) In this case you would set number of channels to 1 and select 1, 2, 3, or 4 as desired.

For all other settings, follow the analog input instructions.

Creating or Editing a Local Output

To create an output, click Add Output in either the Analog or Digital column.

To edit an output, click on the box for the output that you wish to edit.

You will see the output editor.

Fill in the following information:

1. Surface Label – two fields of 8 characters to identify the source internally on the JetStream State page
2. Unique Name – give the source a unique name
3. Number of channels – select 1 for mono, 2 for stereo.
4. Assign the channels. For mono, set the Lf channel to the desired output number. For stereo, set the LF and Rt channels to the desired outputs for left and right. If this is analog, the channel number will correspond with the output number. If this is digital, 1 and 2 are the left and right channels of AES output 1; 3 and 4 are the left and right channels of AES output 2.
5. Default Source – choose what audio should be fed to the output. This pick list has each input along with the console mixes. Available console mixes:
 - Program Out – the main program mix
 - Aux 1 Out – the aux 1 mix bus
 - Aux 2 Out – the aux 2 mix bus
 - Aux 3 Out – the aux 3 mix bus
 - Studio Out – the studio monitor output. This corresponds to the Studio/Guest volume knob on the console. Usually used for talk studio monitor speakers or for guest headphones.
 - Phones Out – the headphone monitor output. This corresponds to the Headphone volume knob on the console. Usually used to feed the headphone amp of the console operator.
 - Phones PF – pre-fader output of the headphone monitor. Often used to feed guest headphone amps with individual volume controls. The audio will follow what the operator selects in the headphone monitor input, but the volume knob will be ignored. This way the board operator who is listening to Phones Out can turn the volume up and down without disturbing the guest.
 - Monitor Out – the control room monitor output. This corresponds to the Monitor volume knob on the console.
 - MM 1 – MM 24 – Mix Minus outputs 1-24. See the Mix Minus section under DSP Settings to learn how to configure mix minuses.
 - Mixer A Out – This is a three channel submixer that sums the source routed to MixA1, MixA2, and MixA3 on the Surface Settings page. Commonly used to sum together microphones for telephone feeds, it's a lot easier to tick 3 boxes on a web page than bury a Shure mic mixer under the console like we all used to do to feed the phone!
 - Mono PGM – the program bus, summed to mono
 - Mono Ax1 – the Aux 1 bus, summed to mono
 - Router 1 Out – the output to Router 1, the left widget on the mixIT fader display.
 - Router 2 Out – the output to Router 2, the middle router widget on the mixIT fader

display

Router 3 Out – the output to Router 3, the right router widget on the mixIT fader display

The Available Sources column on this page does NOT have a function on the JET67.

If you route a stereo mix to a mono output, the JET67 will send left channel only.

The Network IO Page

Click on the **Network IO** button to look at the network audio. The default view when you click on this page is the **Outputs To Network** tab.

While the AES-67 standard got various manufacturers to agree upon timing and packet size, it did not create a standard way for audio devices to tell other things on the network the address of a stream or its name. Therefore, when you create an output to the network, the JET67 will announce the stream three different ways: via the **Livewire+** method, the **Ravenna** method, and via the **Dante** method of announcing AES-67 streams.

Note that Dante has two different flavors of AoIP. There's Dante, which communicates with other Dante devices only, and there's Dante's implementation of AES-67 where you can create a "flow" in Dante Controller to announce a stream as AES-67.

Dante devices that can handle AES-67 will see AES-67 streams from the JET67 in the AES-67 side of Dante Controller, whether you have a J67-DANTE card installed in the JET67 or not. Dante devices that cannot handle AES-67 and only speak Dante (for example, Audinate's inexpensive AVIO devices or their \$29 sound card driver) will require having a J67-DANTE card installed in the JET67.

Outputs To Network

To add an output to the network, click the Add Output button. To edit an existing output, click on the line you wish to edit. The editor will appear.

Fill in the following information:

1. **Surface Label:** two fields of 8 characters. This is the information to be sent to a console display.
2. **Unique Name:** Give the output a unique identifier. **Note: if a J67-DANTE card is installed and you wish to send this output as Dante instead of AES-67, this field must begin with the following D| (D, the pipe symbol (usually shift + backslash key), and a space followed by a unique name**
3. **Number of channels**
4. **Default Route:** select the audio you wish to send to the network. This drop-down includes all inputs plus console mixes. Available console mixes:
 - Program Out – the main program mix
 - Aux 1 Out – the aux 1 mix bus
 - Aux 2 Out – the aux 2 mix bus
 - Aux 3 Out – the aux 3 mix bus
 - Studio Out – the studio monitor output. This corresponds to the Studio/Guest volume knob on the console. Usually used for talk studio monitor speakers or for guest headphones.
 - Phones Out – the headphone monitor output. This corresponds to the Headphone volume knob on the console. Usually used to feed the headphone amp of the console

operator.

Phones PF – pre-fader output of the headphone monitor. Often used to feed guest headphone amps with individual volume controls. The audio will follow what the operator selects in the headphone monitor input, but the volume knob will be ignored. This way the board operator who is listening to Phones Out can turn the volume up and down without disturbing the guest.

Monitor Out – the control room monitor output. This corresponds to the Monitor volume knob on the console.

MM 1 – MM 24 – Mix Minus outputs 1-24. See the Mix Minus section under DSP Settings to learn how to configure mix minuses.

Mixer A Out – This is a three channel submixer that sums the source routed to MixA1, MixA2, and MixA3 on the Surface Settings page. Commonly used to sum together microphones for telephone feeds, it's a lot easier to tick 3 boxes on a web page than bury a Shure mic mixer under the console like we all used to do to feed the phone!

Mono PGM – the program bus, summed to mono

Mono Ax1 – the Aux 1 bus, summed to mono

Router 1 Out – the output to Router 1, the left widget on the mixIT fader display.

Router 2 Out – the output to Router 2, the middle router widget on the mixIT fader display

Router 3 Out – the output to Router 3, the right router widget on the mixIT fader display

5. Click Upload to JetStream to save. Click the Cancel Edit Change button to exit without saving. Click Delete this output to remove the output from the network.

The Available Sources column on this page does NOT have a function on the JET67.

If you route a stereo mix to a mono output, the JET67 will send left channel only.

Inputs From Network

Click on this tab to view and edit inputs coming from the audio network.

JetSet JetStream Plus LOG-ENG-D-01

Local IO **Network IO** Surface Settings DSP Settings Control Settings System Info JetStream Admin

Outputs to Network Inputs from Network

Red Inputs from Network are NOT active

	Stream Handle	Stream Name	Stream Source	Source Type	Stream IP Address	Num Channels	Latency
1	0x0200	Livewire 1	JetNet83		0.0.0.0	2	Low
2	0x0201	Livewire 1	JetNet82		0.0.0.0	2	High
3	0x0202	Net Tone	JetNet82		0.0.0.0	2	High
4	0x0203	AES-3 Lw2	JetNet82		0.0.0.0	2	High
5	0x0204	AES-4 Lw2	JetNet82		0.0.0.0	2	High
6	0x0205	Iyo1616M-145b30	JetNet83		0.0.0.0	2	Low
7	0x0206	Analog L	JetNet68		0.0.0.0	1	High
8	0x0207	Analog R	JetNet68		0.0.0.0	1	High
9	0x0208	Net67-17a294 : 3	JetNet83		0.0.0.0	2	Low
10	0x0209	Pgm 21 AES	JetNet83		0.0.0.0	2	Low
11	0x020A	Iyo1616M-145b30	JetNet83		0.0.0.0	2	Low
12	0x020B	R3LAY01 - GRP	JetNet82		0.0.0.0	2	High
13	0x020C	Digital L	JetNet68		0.0.0.0	1	High
14	0x020D	Digital R	JetNet68		0.0.0.0	1	High
15	0x020E	MM-John	JetNet68		0.0.0.0	1	High
16	0x020F	Mic/Line In 6	JetNet68		0.0.0.0	1	High
17	0x0210	Mic/Line In 7	JetNet68		0.0.0.0	1	High
18	0x0211	Telos	JetNet68		0.0.0.0	1	High
19	0x0212	Mic/Line In 9	JetNet68		0.0.0.0	1	High
20	0x0213	Mic/Line In 10	JetNet68		0.0.0.0	1	High
21	0x0214	Mic/Line In 11	JetNet68		0.0.0.0	1	High
22	0x0215	Mic/Line In 12	JetNet68		0.0.0.0	1	High
23	0x0216	Mic/Line In 13	JetNet68		0.0.0.0	1	High
24	0x0217	Mic/Line In 14	JetNet68		0.0.0.0	1	High

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Figure 13 - Inputs From Network

Standards are wonderful. That's why we have so many to choose from. While AES-67 created standard packet sizes and timing methods, it did not specify which method manufacturers should use to announce to other devices the names and addresses of their sources.

At Logitek, our solution to this dilemma was to identify the three biggest AoIP systems on the market, license them, and then create an internal channel guide that searches for all three and builds a list in the JET67 of what's available on the network. Therefore, if your other AoIP devices use **Livewire+**, **Ravenna**, or a **Dante** card in AES-67 mode, we will automatically find those streams and put them into the Inputs From Network list.

Native Dante streams will appear in the network pages when the Dante card is installed. More information about Dante is in the Dante setup chapter.

One more thing to note on this page before we go into more detail about how to use it: you will see at the top of the screen that **"Red Inputs from Network are NOT active."** **Disregard this!**

It's left over from the JetNet audio networking protocol that has been retired. EVERY input on this page will be red, and it's perfectly okay.

Since this page is showing you what it has found from scanning the network, you'll notice there is no "Add Input" button.

This page is important because this is where you tell the JET67 how to handle these sources when they are routed to the console. Remember in the local IO section where you could assign an input to a mix minus bus, associate an input with GPIO or set the mode? This is how you connect those same features to a source that came in from the network.

Click on the line of the input you wish to edit and we'll begin.

Figure 14 - The Network Input Edit Window

Note that the Stream Name, Unique Name, Source, Type, Number of Channels, and Latency fields are greyed out. This information is supplied by the source machine, so we can't edit those things.

What we can do is tell the console what to do when handling this source:

GPIO Inputs (valid numbers are 1-4)

Audio On – enter the GPIO Input where a contact closure will turn the channel on.

Audio Off – enter the GPIO Input where a contact closure will turn the channel off.

Cough – enter the GPI input where when closed will turn on the cough mute.

GPI Outputs (valid numbers are 1-8)

Audio Start – enter the GPI output where we will pulse the relay when the fader is turned on

Audio Stop – enter the GPI output where we will pulse the relay when the fader is turned off

Audio Tally – enter the GPI output where we will latch the relay when the fader is on

Mute Group – enter the mute group that this microphone will belong to, where the first mic on mutes the monitors and turns on the mute tally as defined in Control Settings, and the last mic off un-mutes the monitors and turns off the mute tally as defined in Control Settings.

Tip: as with the local inputs, put the on air light for the studio in the Mute Tally on the Control Settings page so it follows all of the mics in the mute group. Use the Audio Tally for things that follow just this one source, such as a LED glow ring for a single microphone.

UDP Control

To have the JET67 send ASCII text via UDP for device control enter the following:

UDP Port Number: a valid port number

UDP IP Address: the IP address of the device to which we are sending the text string

Start String: the text to send when the fader turns on

Stop String: the text to send when the fader turns off

The following checkboxes are relevant to the mixIT:

Fader Start: if set to yes, turn the channel on when the fader is raised from zero and off when the fader is returned to zero.

Cue at Inf: If set to yes, put the fader in cue when the fader is returned to zero

Click Upload to JetStream to save your work.

Click Cancel Edit Changes to exit

If you click Delete this input from network, you will remove it from the list of sources in the system, but it will return the next time the stream is announced by the sending device.

Surface Settings

Here is where you assign sources to the faders

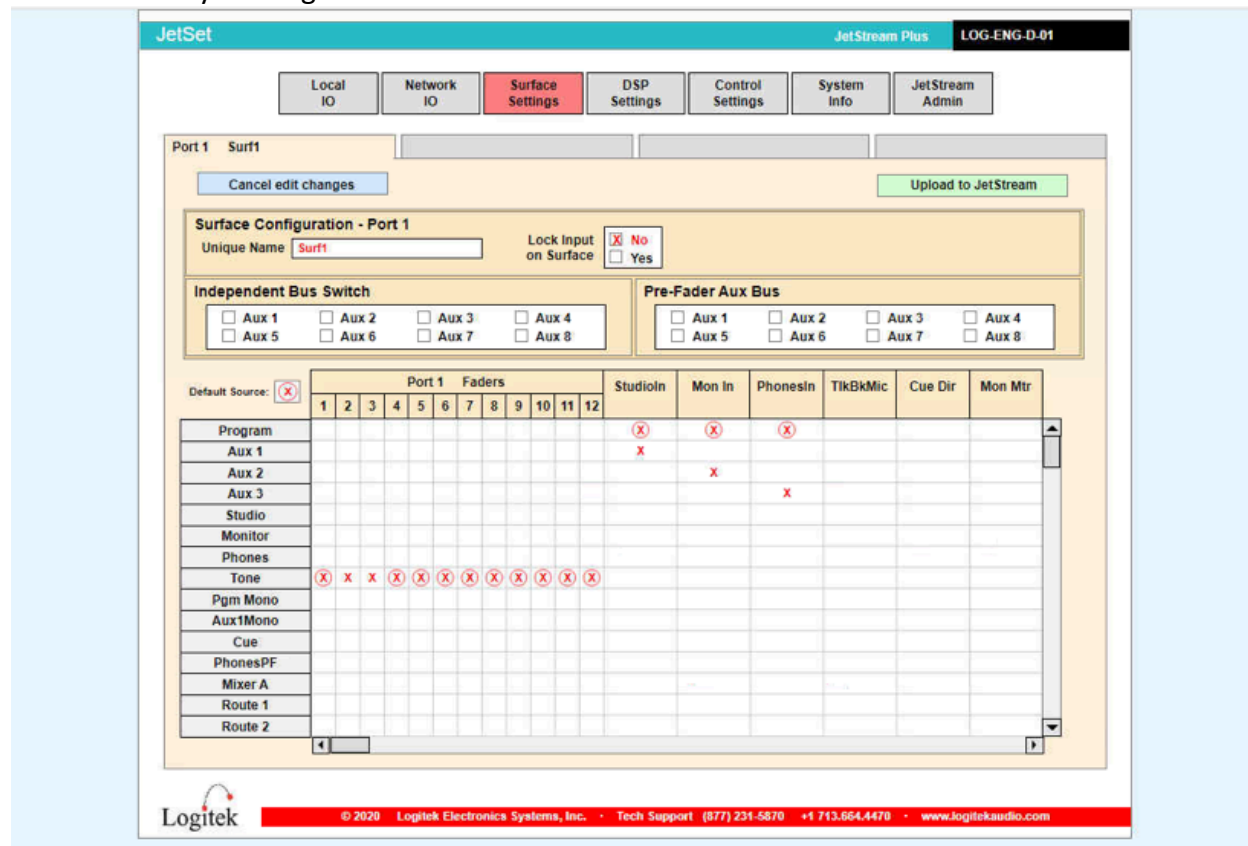


Figure 15 - The Surface Settings Page

The columns of this grid are the available sources. The lines in gray are mixes from the DSP such as program, aux, mix minus, and the built in tone generator. The lines in blue are local inputs. The lines with a beige background are network inputs.

Left click in the columns 1-12 to assign a source to a fader. This will place an x in the box.

Right click in the columns 1-12 to set the default route on a fader. This is the route that the console will return to when the system is sent to its default settings. This will place a circle x in the box. Only one default can be set per column. If no default is set, the first source in the column is automatically made the default.

In the column for Studio In, set what selections to choose from in the Studio/Guest monitor selector. Common selections here are program, aux 1-3, air monitor, or an input with a processed feed where the talent can hear the output of the on-air processing chain if they can't listen to air because of profanity delay or HD Radio diversity delay.

In the column for Monitor In, set what selections to choose from in the Control Room monitor selector. Common selections here are program, aux 1-3, air monitor, or an input with a

processed feed where the talent can hear the output of the on-air processing chain if they can't listen to air because of profanity delay or HD Radio diversity delay.

In the column for Phones In, set what selections to choose from in the Headphone monitor selector. Common selections here are program, aux 1-3, air monitor, or an input with a processed feed where the talent can hear the output of the on-air processing chain if they can't listen to air because of profanity delay or HD Radio diversity delay.

In the Talkback Mic column, select the microphone you want sent as talkback to a mix minus when you hit the talkback button on the console. This is also the audio sent to the caller when using the Add Mic When Off feature on the mix minus bus. You can only select one source as the talkback mic.

If you want to mix microphones together to feed to talkback, here are your options:

Assign one mic to MixA In 1, a second mic to MixA In 2, and a third mic to MixA In 3. Then set MixerA as the talkback mic.

or

Tick the Pre-fader aux bus box for one of the auxes (such as aux 3) and tick the box for the same aux bus for Independent Bus Switching. Then set the output of that aux bus as the talkback mic.

Cue Dir and Mon Mtr are not used on the mixIT.

MixA1 In, MixA2 In, MixA3 In. This is a three channel submixer used to combine three sources. Assign one source to the MixA1 column, one source to the MixA2 column, and one source to the MixA3 column. These will be summed together and output as Mixer A.

Route 1, Route 2, Route 3: These are the three router widgets on the right-hand side of the mixIT-12 or the three router widgets on the mixIT-6. Assign tick marks to the sources you want available to be routed by these router widgets the same way you assign sources to the console faders.. Assign the output to either a local or network output.

Tip: you can rename these routers to what they are feeding in the DSP Settings section under Inputs. The label will appear both in the Surface Settings and on the router widget on the console.

Click Upload to JetStream to save your work. If you leave the webpage without saving, any changes will be lost.

DSP Settings

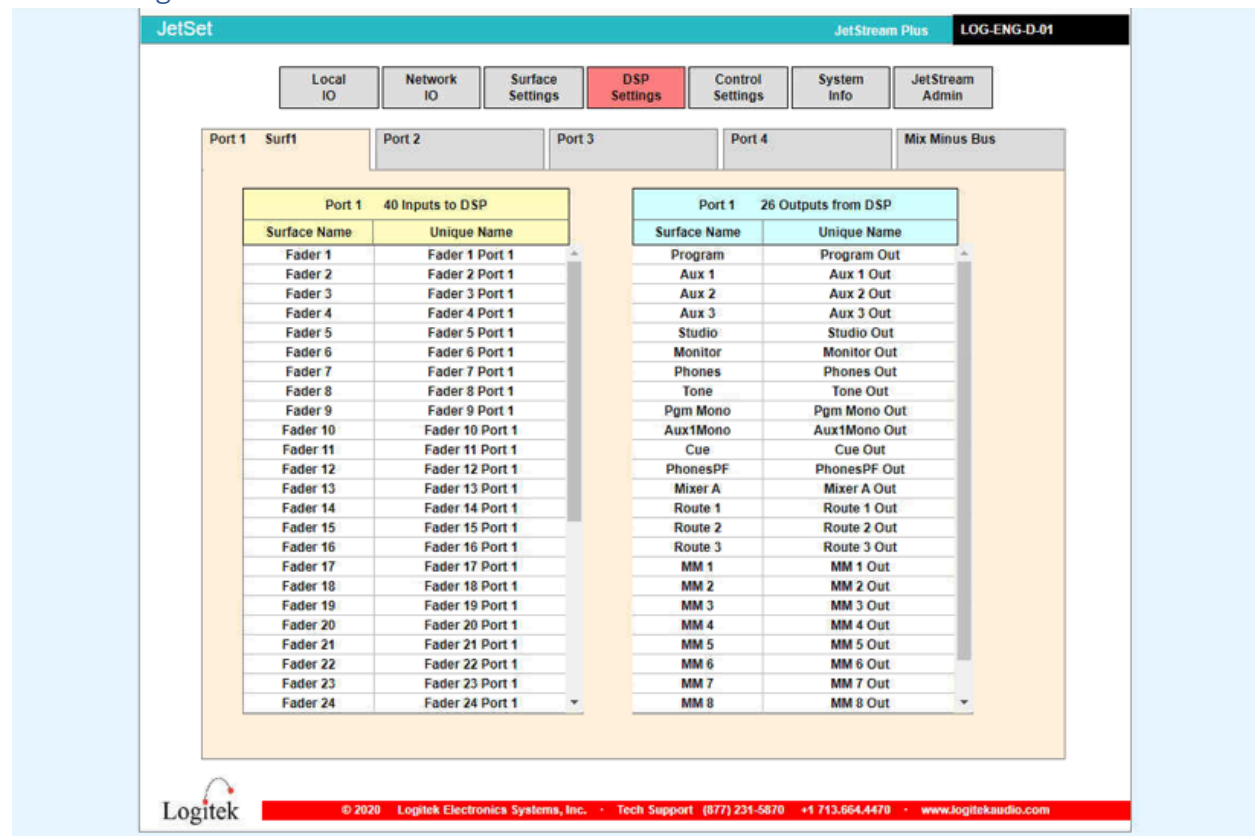


Figure 16 - The DSP Settings Page

Port 1 Tab

This is where we can modify the DSP settings. Changes allowed:

Under Inputs, click on a fader to select which buses are turned on by default. Defaults for the faders is the program bus. If you wanted to make certain faders always turn aux on by default, click on the fader and put an x in the box of the bus you want turned on by default.

Also, under inputs, click on Router 1/Router 2/Router 3 to change the name displayed under the router widget. This is useful to name the router for its destination.

Under outputs, you can click on a DSP output to change the surface label of the mix bus. If you want to change Program to instead read “WKRP Program” here’s your chance.

Mix Minus Tab

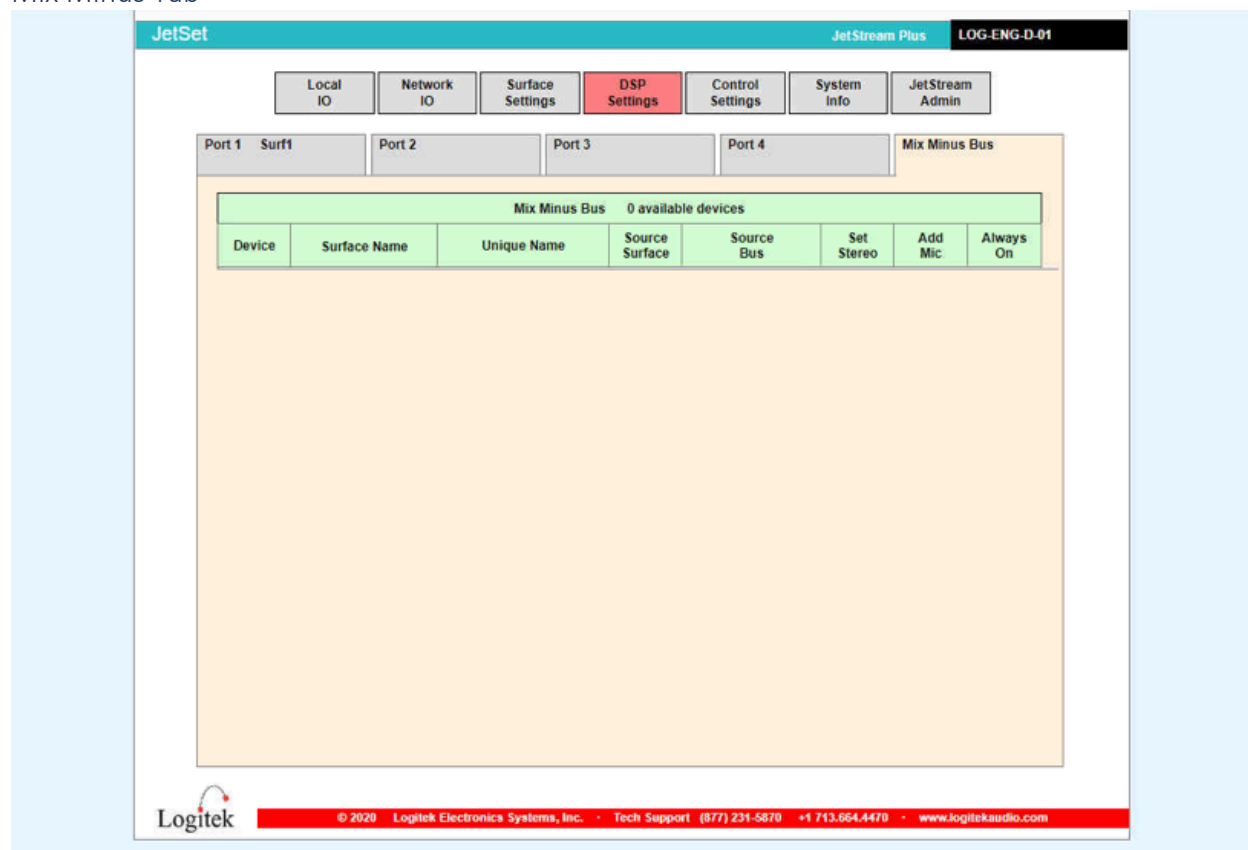


Figure 17 - The Mix Minus Tab

Click to edit each mix minus bus. On the mixIT, 12 mix minuses are available.

How To Set Up A Mix Minus

A mix minus starts with a mix bus such as program, aux 1, aux 2, or aux 3 and subtracts certain inputs from it. This is primarily used with remote broadcasts where there’s transport latency. You want the person in the field to hear everything in the mix but themselves. This prevents an echo coming back to the person in the field and making it difficult for them to talk.

The mixIT has 12 mix minus buses. You will tag the inputs, either local or network by editing them and entering the mix minus number and saving it. Then here on the DSP settings page you will define the mix minus bus and turn it on. Finally, you will route the output of the mix minus bus either to a local or network output to feed your equipment.

Here is what to do on the DSP Settings Page:

- Click on the line to edit the mix minus bus

- Set the Source Surface number to 1 (there is only one console).
- Set the Source Bus. This is the mix you are removing audio from. Usually this is program.
- If this is a stereo mix minus, Set Stereo to yes. If it's a mono return feed, leave this as no. NOTE: if you set a mix minus bus to stereo, the next mix minus bus in the list will not be available as it is being combined with this bus to create 2 channels of audio.
- The final choice is "Add Mic When Off" or "Bus Always On." Add Mic When Off is usually used for telephone hybrids. When the telephone fader that's associated with this mix minus is off, the console will send the talkback mic as defined in Surface Settings to the caller. When the telephone fader is turned on and the caller goes on air, the console will feed program minus the telephone.

Bus Always On is usually used with remote broadcast codecs. When selected, the console will always feed program audio to the remote minus the codec. That way when someone connects in the field they will immediately hear return audio and be sure they're connected.

Once you have saved this, the mixIT system will be creating a mix minus and when you route it to an output you will hear the mix.

Things to remember: you can only assign one mix minus number to an input. If you assign the same number to more than one input, all of those inputs will be subtracted from the mix. You must turn on the mix minus here in the DSP settings page and either enable "add mic when off" (and have a talkback mic defined in Surface Settings) or enable "bus always on" for audio to come out of a mix minus output.

Control Settings

This is where we define microphone mute groups.

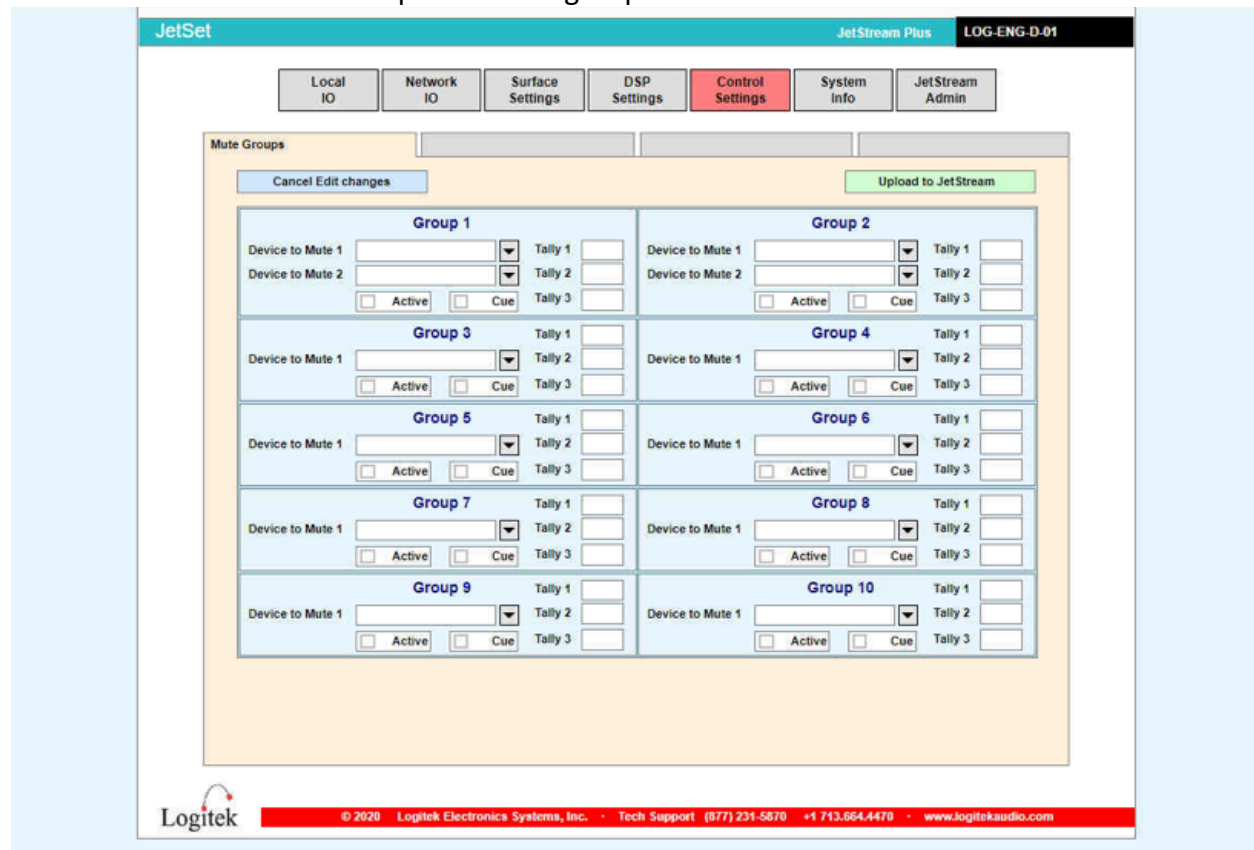


Figure 18 - The Control Settings Page, where Mute Groups are defined

For a mute group you must choose the following:

- Which mix bus you wish to mute (either Monitor (control room) or Studio/Guest)
- Which tally or tallies you wish to turn on (numbers 1-8 are valid)
- If you check the cue bus, the monitors will be muted with a mic that's a member of this mute group is put in cue
- The Active button must be checked for the mute group to be enabled. If left unchecked the console will ignore the mute group.

Once you have defined a mute group on this page, go to each microphone input (local or network) and put the mute group number in the box named Mute Group.

When the first microphone in the group is turned on, the monitors will mute and the tally relays will turn on and remain so until the last microphone in the group turns off.

There are 10 available mute groups in the system, although most stations use one or two. The first two mute groups are capable of muting more than one monitor bus. Up to three tallies can be defined per mute group to turn on multiple on air lights, logging skimmers, and the like.

Dante Setup

Coming Soon

Dante for the JET67 is in development as of this writing. This section will be updated when the Dante card is shipping.

Wiring Diagrams



Figure 19 - Rear view of the JET67

Audio Connections

The JET67 uses 5 pin European-style terminal blocks, commonly known as Phoenix connectors. These solderless terminal blocks are easy to connect.



They are grouped as follows:

+	-	Gnd	+	-
Dig	In 1		Dig	In 2

mixIT-6 and mix-IT-12

+	-	Gnd	+	-
Dig	Out 1		Dig	Out 2

mixIT-6 and mix-IT-12

+	-	Gnd	+	-
Mic	In 1		Mic	In 2

mixIT-6 and mix-IT-12

+	-	Gnd	+	-
Ana	In 1		Ana	In 2

mixIT-6 and mix-IT-12

+	-	Gnd	+	-
Ana	In 3		Ana	In 4

mixIT-6 and mix-IT-12

+	-	Gnd	+	-
Ana	In 5		Ana	In 6

mixIT-6 and mix-IT-12

+	-	Gnd	+	-
Ana	Out 1		Ana	Out 2

mixIT-6 and mix-IT-12

+	-	Gnd	+	-
Ana	Out 3		Ana	Out 4

mixIT-6 and mix-IT-12

+	-	Gnd	+	-
Ana	Out 5		Ana	Out 6

mixIT-6 and mix-IT-12

+	-	Gnd	+	-
Ana	Out 7		Ana	Out 8

mixIT-6 and mix-IT-12

+	-	Gnd	+	-
Mic	In 3		Mic	In 4

mix-IT-12 only

+	-	Gnd	+	-
Ana	Out 9		Ana	Out 10

mixIT-12 only

+	-	Gnd	+	-
Ana	Out 11		Ana	Out 12

mix-IT-12 only

+	-	Gnd	+	-
Ana	Out 13		Ana	Out 14

mixIT-12 only

+	-	Gnd	+	-
Ana	Out 15		Ana	Out 16

mix-IT-12 only

GPIO

DB-15 Connector (female jack mounted on chassis)

- | | |
|---------------|-------------|
| 1. Out-1 | 9. +3.3 VDC |
| 2. Out-2 | 10. Out-7 |
| 3. Out-3 | 11. In-1 |
| 4. Out-4 | 12. In-2 |
| 5. Out-5 | 13. In-3 |
| 6. Out-6 | 14. In-4 |
| 7. Out-Common | 15. Out-8 |
| 8. Ground | |

Inputs grounded to activate. The maximum input is 12 volts. The outputs are opto-isolated electronic relays. The maximum load is .5 amps at 50 volts.

Console Operation

The mixIT is very easy to use. The console uses touchscreens to select sources and bus assignments and knobs and buttons to adjust levels and turn faders on and off. This gives the operator the best of both worlds: simple menus and fewer buttons while still allowing someone to operate the board on air by feel without looking at where their hands are going.

Fader Assignments

Any source can route to any fader. The sources are assigned during the setup process using the Surface Settings menu in the web browser.



Tap the name of the source over the fader to change it. A list of choices appears. Tap the arrow buttons to scroll through the list. Tap the name of the desired source to route it to the fader. When you tap the source, the menu will close.

Bus Assignments

The console has 4 main mix buses: Program, Aux 1, Aux 2, and Aux 3. Usually Program is the main on air output.



As with changing the route, tap the name of the source over the fader you wish to change. Tap P (Program), 1 (Aux 1), 2 (Aux 2), or 3 (Aux 3) to assign the fader to the mix bus. When you have made your selections, tap "Done."

Fader Operation



To raise the audio level of a source, move the fader up.

To decrease the audio level of a source, move the fader down.

The dashes silkscreened on the panel marked "0" represents unity gain. This is the normal level for most sources. At this setting, there is still 10 dB of gain available on the fader if needed.

To turn a channel on, press the top push button under the fader. The on button will glow green.

To turn a channel off, press the lower push button under the fader. The off button will glow red.

The Cue Bus



To audition a source before putting it on the air, tap the “Cue” button. This will send the source’s pre-fader audio to the cue speaker built into the console. Tap the cue button again to take the source out of the Cue bus.

Talkback

There is a talkback button over each fader. If a mix minus is associated with this source (for example, a codec for remote broadcasting), tapping the **Talk** button will route the studio talkback microphone (as defined in surface settings) to the mix minus output. Tap the button again to turn talkback off and resume sending mix minus to the output.

There is also a Talk button next to the Guest/Studio selector. This is often used to feed a monitor to a separate studio. Tapping this Talk button will insert the same talkback microphone into the Guest/Studio monitor feed.

Studio/Guest, Headphones, Monitor Controls



The Studio/Guest, Headphone, and Monitor selectors are on the far right. Tap the source to change what audio is being fed to these outputs. The volume knobs below the screen will adjust the level.

Utility Routers



There are three utility routers on the console that can be used to change what is fed to the air chain, recorders, or remote codecs. Tap the box for the router to change the input.

Scene Selects



There are 4 scene select buttons on the mixIT-12. Different console profiles can be stored to those buttons in setup. Press the button to recall the scene.

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LOGITEK will repair or replace, at its option, at its factory without charge professional equipment if a defect in materials or workmanship develops during the first two years following purchase, when the equipment is returned to the factory or LOGITEK authorized service centers freight prepaid with a description of the nature of the failure. No reimbursements can be made for repair charges that are not factory authorized. After repair or replacement, LOGITEK will return the equipment to the purchaser freight prepaid.

In the event that any part of this professional equipment becomes defective during the first two years following purchase, and purchaser wishes to attempt repair, purchaser may obtain a replacement part by notifying LOGITEK of the part of the equipment which has failed. LOGITEK will thereafter ship a replacement part, freight prepaid. LOGITEK may require the purchaser to return the defective part to LOGITEK freight prepaid as a condition of such replacement, either before or after LOGITEK ships the replacement part. LOGITEK shall not be responsible for any other charges or liabilities associated with purchaser-made repairs.

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