Dan Dugan Sound Design

Model D-3 Automatic Mixing Controller

User Guide

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Part Number: D-3M
Author: Rob Wenig

290 Napoleon St. Studio E
San Francisco, California 94124
Phone: 415-821-9776
Fax: 415-826-7699
Web: http://www.dandugan.com
e-mail: dan@dandugan.com

Important Safety Instructions and Warnings

The Model D-3's circuitry is made in the U.S.A. and meets applicable national safety standards.

Standards Compliance

The third-party power supply provided with this product has been certified to comply with UL and CE.

Safety Instructions

1. Read these instructions.
2. Keep these instructions.
3. Heed all warnings.
4. Follow all instructions.
5. Do not use this apparatus near water.
6. **WARNING!** To reduce the risk of fire or electric shock, do not expose this apparatus to rain or moisture.
7. Clean only with dry cloth.
8. Do not block any ventilation openings. Install in accordance with the manufacturer's instructions.
9. Do not install near any heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.
10. **Warning!** This is a Class 1 device. The electrical safety design of Class 1 devices depends on proper grounding. To maintain electrical safety ensure that a grounded mains lead is used and that it is properly connected to a grounded mains wall outlet.
11. Do not defeat the safety purpose of the polarized or grounding-type plug. A polarized plug has two blades with one wider than the other. A grounding type plug has two blades and a third grounding prong. The wide blade or the third prong are provided for your safety. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.
12. Protect the power cord from being walked on or pinched particularly at plugs, convenience receptacles, and the point where they exit from the apparatus.
13. Only use attachments/accessories specified by the manufacturer.
14. Unplug this apparatus during lightning storms or when unused for long periods of time.

15. **WARNING!** Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as power-supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped.

16. **WARNING!** To reduce the risk of electric shock, DO NOT REMOVE COVER. No user serviceable parts inside.

Warranty Statement

Warranty: one year parts and labor

Dan Dugan Sound Design warrants that Model D-3 hardware will be free from defects in components and workmanship for a period of 12 months from the date of invoice. During the warranty period, Dan Dugan Sound Design will cover the cost of all parts and labor to remedy the defect, or replace products which prove to be defective. Dan Dugan Sound Design is not obliged to honor this warranty if the hardware has failed to be maintained and operated as specified by Dan Dugan Sound Design, in the accompanying documentation, or other than in accordance with industry standards. Defects caused by unauthorized modifications, misuse, negligence, act of God or accident are not covered by this warranty. This Limited Warranty is exclusive and no other warranty is expressed or implied. Dan Dugan Sound Design does not warrant that Dan Dugan Sound Design software, or any third-party software, is error free. Third party branded or manufactured goods are supplied by Dan Dugan Sound Design with care but without responsibility and subject only to third party suppliers' warranties. In all other respects Dan Dugan Sound Design is not liable for consequential damages.

Quickstart

This section provides step-by-step instructions to help you get started quickly. It includes information to connect and install the D-3, make initial settings, and link Processors. See *Conventions* on page 13 before proceeding.

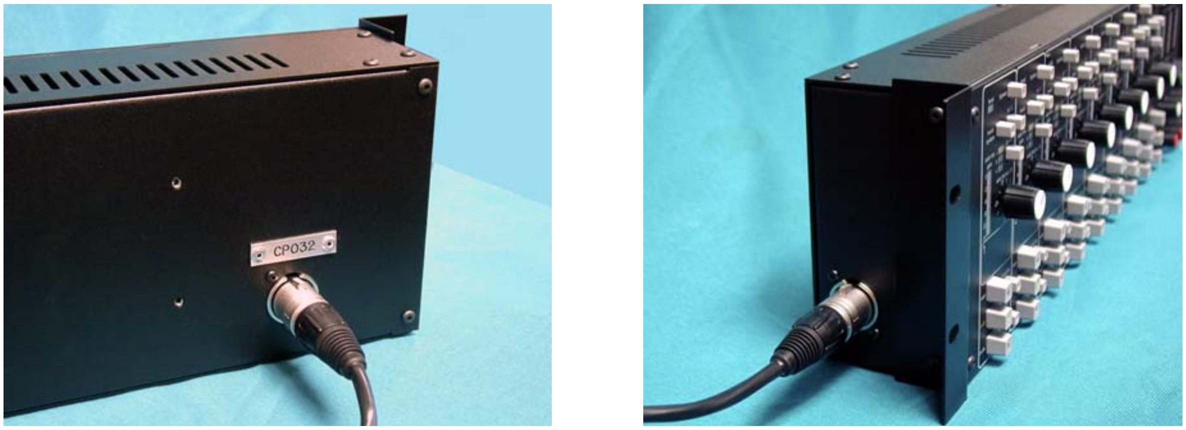
Connections

Q-1 Rear panel

1. Connect the DC power supply (provided) to the Processor.
2. Connect the Processor to the Control Panel with the five-pin XLR cable (provided).

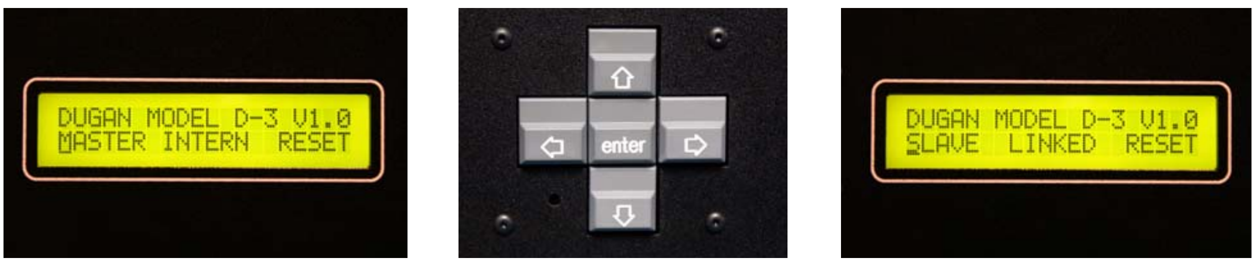
Q-2 Power and Control Panel connections

The Control Panel may be connected on its rear panel if the unit will be operated in an upright position, or on the left side if the unit will be laid on its back.



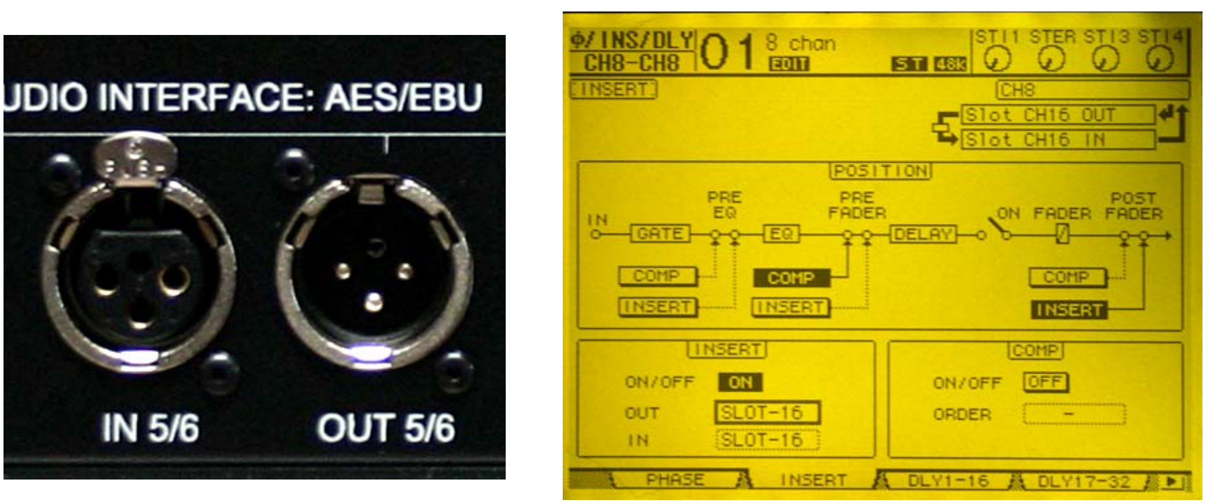
Q-3 Side and rear panel connections for Control Panel

3. If there are multiple Processors, make sure one Processor is set to Master mode; the other Processor(s) must be set to Slave mode. Position the cursor on **MASTER** or **SLAVE** using the left and right buttons on the front panel, then press the **ENTER** button to toggle the setting.



Q-4 Changing Processor from Master to Slave

4. Using insert cables wired appropriately for your console, connect each channel of the Processor into the insert loop (send and return) of each mic input channel.



Q-5 Audio connections

Settings

The following instructions pertain to the Speech System only. See *Dugan Music System* on page 33 for instructions on using the Music System.

***NOTE:** All the bypass LEDs blink together when the Model D-3 is not synchronized.*

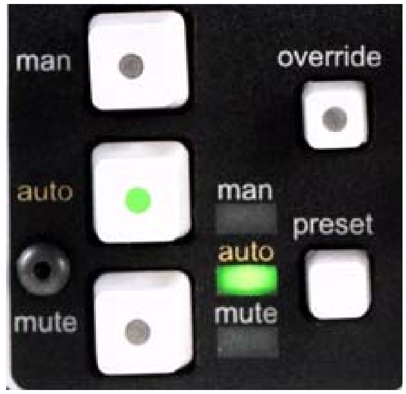
1. Set all rotary knob controls on their yellow marks.

Q-6 Channel settings

2. De-activate the **music system**, **NOM**, and **override** buttons (no lights) on each channel.

3. Select **group a** on each channel by pressing the **group** button until the **a** LED lights.

4. Activate the **auto** button for all channels in use; select the **mute** mode for unused channels.



Q-7 Preset setting

5. Press the **preset** button until the LED to its left matches the setting in step 4.

Gain Structure

1. Set the **auto mix weight** controls to **0**.

2. Raise your console preamp gains until the green **level** LEDs stay illuminated.

Q-8 Setting console preamp gain

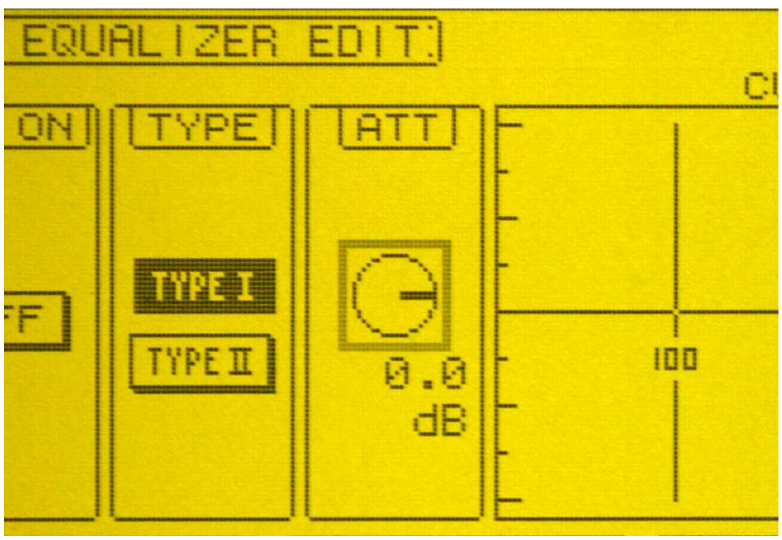
The **level** LED flashes red to indicate clipping and illuminates green when the input level is within the normal range.

There should be adequate gain to illuminate the green light all the time. When the green light goes out, the channel goes into an automatic safety downward expansion mode, which loses the smooth one-mic ambience. This out-of-range condition should be avoided.

3. Sufficient ambient sound level at the D-3 input is about -75 dBFS. If raising the preamp gains does not produce enough level to keep the **level** LED on, raise all the **auto mix weight** controls the same amount (Figure Q-9).

Q-9 weights at +10

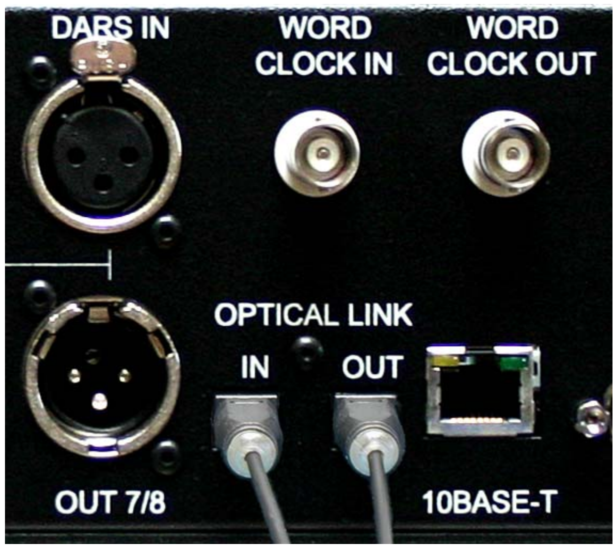
4. If raising the weight controls still does not produce enough level to keep the **level** LED on, raise the gain trim of the console equalizer (Figure Q-10).



Q-10 Using EQ trim for more gain

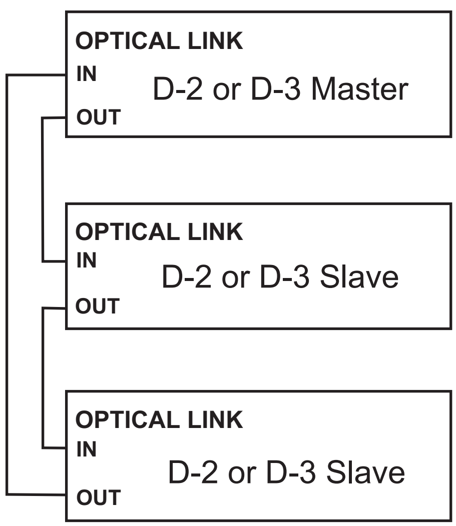
Linking Units

D-3 Processors can be linked in a ring with optical digital audio cables (i.e., ADAT lightpipe, Toslink). There is one master Processor and the rest are slaves. THEY MUST BE LINKED IN BOTH DIRECTIONS (i.e., it takes two cables to link a master and a slave). For three or more units, daisy-chain **LINK OUT** to **LINK IN** connectors, then loop back from the last to the first unit.



Q-11 Optical link connectors

The Model D-3 can be linked with Model D-3s and/or D-2s up to a maximum of 64 inputs.



Q-12 Linking analog and digital units

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Introduction

Conventions

We will use:

- *activate* to mean toggling on a button (LED lit).
- *de-activate* to mean toggling off a button (LED not lit).
- *select* to mean pressing a button repeatedly until the desired LED is lit.

Introduction to the Model D-3

The Model D-3 Automatic Mixing Controller helps professional audio mixers handle multiple live mics without having to continually ride their individual faders. The eight-channel Processor patches into the input insert points of an audio mixing console. The Model D-3 detects which mics are receiving input and makes fast, transparent cross-fades, freeing the mixer to focus on balance and sound quality instead of being chained to the faders.

The Model D-3 supports a broad spectrum of live mixing applications:

- Broadcast news, talk, and game shows
- Conference reinforcement and videotaping
- Dialogue recording for film and television
- Theaters, opera houses, and live performance stages
- Boardrooms and teleconferencing
- Distance learning

The D-3's voice-controlled crossfades closely track unpredictable dialogue, eliminating cuing mistakes and late fade-ups while avoiding the choppy and distracting effects common to noise gates.

The D-3 has many other features that dramatically improve the performance of live mixing with multiple mics:

- Reduces feedback and ambient noise pick-up
- Insert loop patching allows full use of mixing console features
- Separate Control Panel can be used horizontally on table top or vertically on meter bridge
- Three control groups for constant ambience with L-C-R panned mics
- Two separate automatic mixing algorithms optimized for speech and music
- Up to 64 live mics by linking eight units
- Convenient remote control capabilities
- Links with other Model D-3s and/or D-2 controllers

The Model D series marks inventor Dan Dugan's second major audio product innovation. In 1973, Dugan patented the first automatic microphone mixer. Automatic mixers have since become a major audio product category. The Model D-3 incorporates three Dugan-patented technologies:

- The Dugan Speech System™ automatically manages any number of live mics in unscripted talking situations.
- The Dugan Music System™ offers automatic downward expansion to help reduce feedback and noise pick-up in live music performances (i.e., opera, pop-rock back-up vocalists). Thresholds are adjusted automatically through continuous monitoring of ambient noise levels.
- Dugan Gain Limiting™ provides a continuous, stepless, NOM master gain system to prevent feedback and ambient noise build-up when used with either the Music System or with manual or remote mic switching.

The combination of these three Dugan technologies make the Model D-3 a unique audio industry tool. Professional connectors and default control settings marked on the unit allow it to be set up and running with any console in minutes.

Chapter 1: Testing Before Installation

This chapter is intended for experienced technicians to test the Model D-3 prior to installation. Those without test equipment can begin in Chapter 2: *Installation*.

1.1 Bench Tests Before Installation

Units shipped to the USA include a 120 VAC power supply. Contact the factory if you require a different power supply.

1. Place all front panel knobs on the yellow marks. Set the **music system threshold input** and **auto mix depth** controls fully counter-clockwise.
2. Use the right or left arrow on the Processor to move the cursor to **RESET** and press the **ENTER** button.

   This resets all parameters, except the front panel rotary knobs, to factory default settings.

To test the controls:

1. Try each channel's **manual**, **automatic**, and **mute** buttons. Selecting the **manual** button should produce a full gain display. Leave the channels in **auto** mode.

   The **auto mix depth** should produce full gain displays on all channels when rotated fully clockwise. Leave it there for the next two steps.
2. The master **mute** and **override** buttons should fade all channels out.
3. Return the **auto mix depth** control to full counter-clockwise (**auto**).
4. Press and hold down the master **preset** button. Verify that repeatedly pressing each channel's **preset** button causes the **preset** LEDs and channel mode buttons to cycle through their three values. Select **auto** using each channel's **preset** button.
5. Press and hold down the master **override** button. Verify that the **override** button on each channel causes the gains to turn full on and off. De-activate each channel's **override** button (no light).

   If a channel fails to respond normally, first recheck the positions of the front panel controls. Then cycle the power off for at least 10 seconds, and turn the power on again. If difficulties persist please contact the factory.

NOTE: *A known bug in the Model D-3 system is that a short interruption in power can confuse the Processor and display an error message in the Processor window. If this happens, turn the power switch off, mix manually for ten seconds, and then turn it on again. Normal operation will resume.*

The Dugan Speech System can be tested with the following procedure:

1. Switch all channels to **speech** mode by de-activating the **music system** button.
2. Connect two different oscillators to inputs 1 and 2.
3. Set both to 1 kHz frequency and -20 dBFS output.
4. Fine-tune the frequency of one oscillator so that the Model D-3 gain indicators bounce up and down at about 2 Hz (1000/1002 Hz).
5. Adjust the oscillator levels so that the gain displays are equal.
6. Leaving one oscillator in input 1, move the second signal to input 3, 4, etc., observing the bouncing display. This is a good way to test linked systems.

NOTE: *The two inputs mix in the side chain control mix bus. The signal level on this bus beats between no signal (subtraction) and +6 dB (addition) as the waves phase against each other. When the two waves cancel in the control mix, both channels will rise to full. Gains bounce down to -6 dB when the two waves add.*

Test the Dugan Music System with the following procedure:

1. Activate each channel's **music system** button.
2. Verify that all channels are set to **group a**.
3. Connect one oscillator to the channel 1 input, and the other to the **MT IN A** connector on the rear panel (top-middle connector in Figure 1-1).

Figure 1-1 music system threshold input connectors

4. Set both oscillators to -20 dBFS.
5. Set the **music system threshold input** control to 0.
6. Raise and lower the **auto mix weight** control for channel 1.

   At 0 and above, the gain will be full and stay full. Below 0, gain will be reduced progressively as the knob is rotated counter-clockwise. Return the knob to 0.
7. Move the oscillator from channel 1 to channel 2, 3, etc. and repeat the process for each channel.
8. Test the **music system threshold input** B and C with the inputs switched to **groups b** and **c**.
9. Switch all channels back to **speech** mode by de-activating the **music system** button and select **group a**.

To test Dugan Gain Limiting:

1. Activate the **NOM** function for all eight channels.
2. Set the **NOM gain limit (group a)** control to 1.
3. Switch the channels one by one to manual mode.

   Observe the **master gain limiting** display. You will see 3 dB for two inputs, 6 dB for four inputs, and 9 dB for eight inputs.
4. Rotate the **NOM gain limit** control clockwise, which should reduce this display to 0.
5. Return the **NOM gain limit** control to 1, de-activate **NOM** on all eight channels, and select **auto** using each channel's **preset** button.

Chapter 2: Installation

All users should complete all steps in this chapter.

2.1 Setup

1. Install the Processor in a rack convenient for wiring to the console's insert points.
2. Position the Control Panel in a convenient place in the mixer's line of sight. Rack ears can be removed. The Control Panel can be placed flat on the table in front of the console, or on the meter bridge. There are four threaded holes on the rear panel for custom mounting brackets.

Figure 2-1 Control Panel in front of mixer

3. Connect the Control Panel to the **CTRL** connector on the Processor rear panel with the five-pin XLR cable. The Control Panel has two parallel connectors: one on the back and one on the left. Only one may be used at a time.

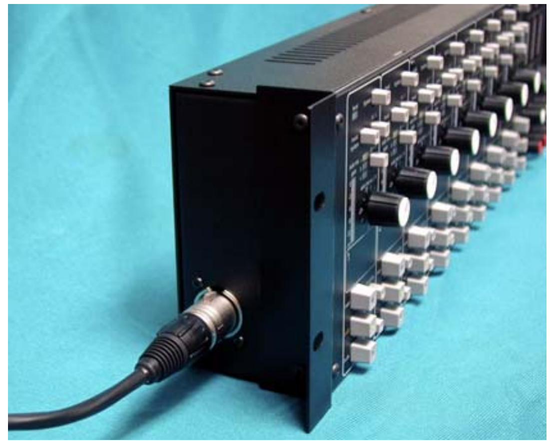

Figure 2-2 Side and rear control connections

4. Reset the Processor by positioning the cursor under **RESET** and pressing the center **ENTER** button.
5. Look at the front panel LCD display and check that the Processor is set to **MASTER** mode. If it is not, use the left and right keys to position the cursor under **SLAVE**, and press the **ENTER** key to toggle **SLAVE** to **MASTER**.
6. Set all rotary knob controls on their yellow marks.
7. De-activate the **music system**, **NOM**, and **override** buttons on each channel.
8. Select **group a** on each channel by pressing the **group** button until the **a** LED lights.
9. Activate the **auto** button for all channels in use.
10. Select the **mute** mode for unused channels.
11. Press the **preset** button until the LED to its left matches the setting in step 10.

2.2 Audio Wiring

2.2.1 Inputs

The eight channel inputs and the Music System threshold inputs are female AES XLR receptacles (pin 1 = shield, pin 2 = +, pin 3 = -). Average program levels from -42 to -16 dBFS are within the acceptable range; -20 dBFS is optimal. The **level** lights are green for all levels in normal operation, and flash red at 0 dBFS.

2.2.2 Outputs

The eight channel outputs are male AES XLRs (pin 1 = shield, pin 2 = +, pin 3 = -). The bypass relays will connect all three pins from input to output when the the power is off.

2.2.3 Connection

Patch the audio channels in the insert loop (send-return) of each console input strip. The Speech System works best when patched post-fader because channels can be muted by pulling their faders down. The Music System works best when patched post-EQ pre-fader so that fader and threshold adjustments remain independent.

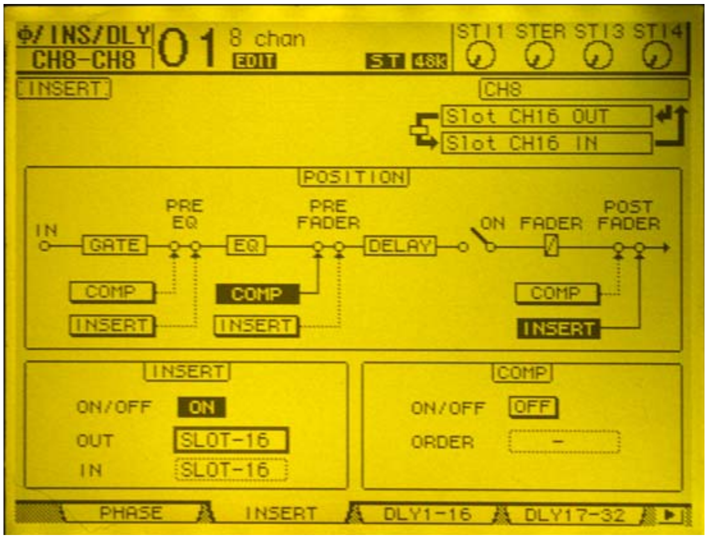

Figure 2-3 Insert post-fader for Speech System

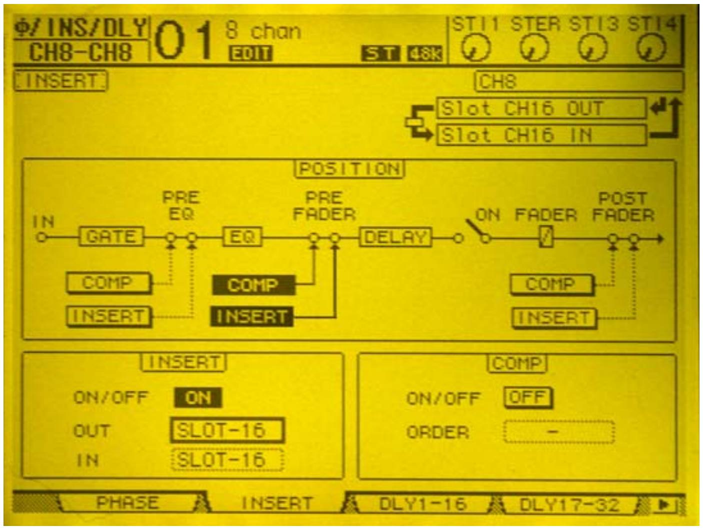

Figure 2-4 Insert pre-fader for Music System

Adjust the console input gain trimmers so that talking signal levels at the sends are between -20 and 0 dBFS, and room noise at the sends is higher than -75 dBFS. Console input gains should be high enough so the **level** lights stay green all the time, but do not turn red on peaks.

2.2.4 Synchronization

An external word clock can be connected to **WORD CLOCK IN**, or an AES digital audio signal can be connected to **DARS IN**. The sync signal should be *digital black* without audio modulation to minimize jitter. In any case, only the sync part of the signal to this input is read.

Word clock signals can be received from the:

- digital audio inputs;
- word clock input;
- DARS (Digital Audio Reference Signal) input.

The default setting after a reset is **INPUTS 1/2**. All bypass LEDs blink to indicate the controller is not synchronized with the incoming digital audio. Under this condition, the digital audio signals just pass through with no latency.

The D-3 performs its internal processing at 48 kHz but the inputs have sample-rate converters that can read any rate. However, the digital audio and word clock outputs operate at 48 kHz only.

NOTE: *All the bypass LEDs blink together when the Model D-3 is not synchronized.*

The best practice is to synchronize to a 48 kHz word clock from the associated mixer system. However, the D-3's inputs have sample-rate converters, so switching to internal synchronization will work if the console inputs can handle unsynchronized 48 kHz signals. That method could subject the signal to two sample-rate conversions, which is not desirable.

To change the synchronization source, use the left and right buttons on the Processor front panel to place the cursor under the center field in the data display. Pressing **ENTER** repeatedly cycles through the choices of **INPUTS 1/2, 3/4, 5/6, 7/8, WORD CLOCK INPUT, INTERNAL,** and **DARS**.

2.3 Linking

Multiple Model D-3 units are linked in a ring so they operate as one system. Two standard optical audio cables (i.e., ADAT lightpipe or Toslink) are required to link two mixers.

NOTE: *This is different from the way the Models D and D-1 are linked.*

2.3.1 Linking Two Model D-3s

Set one Processor to Master mode and the other to Slave. Locate this setting on both Processors by using the right and left arrow buttons on the Processor front panel to position the cursor under the word **MASTER** or **SLAVE**. Press the **ENTER** button to toggle the setting.

Use the optical cables to connect the **LINK OUT** of the first Processor to the **LINK IN** of the second, and the **LINK OUT** of the second Processor to the **LINK IN** of the first.

2.3.2 Linking Three or More Model D-3s

Set one Processor to Master mode and all others to Slave. Locate this setting on each Processor by using the right and left arrow buttons on the front panel to position the cursor under the word **MASTER** or **SLAVE**. Press the **ENTER** button to toggle the setting.

Use the optical cables to connect the **LINK OUT** of the first Processor to the **LINK IN** of the second, the **LINK OUT** of the second Processor to the **LINK IN** of the third. Connect the **LINK OUT** of the last Processor back to the **LINK IN** of the first Processor to complete the ring.

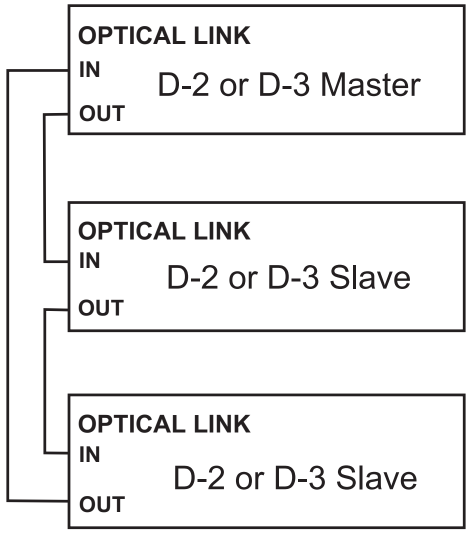

Figure 2-5 Linking analog and digital units

2.3.3 Legacy Linking to the Analog Models D and D-1

Multiple Model D and D-1 analog controllers may be daisy-chained together and then linked into the 15-pin **LINK INPUT** connector on a Model D-2. The D-2 must be the master and the analog units must be slaves. Analog mixers are in **group a** only.

The following diagram illustrates linking Model D, D-1, D-2, and D-3 units.

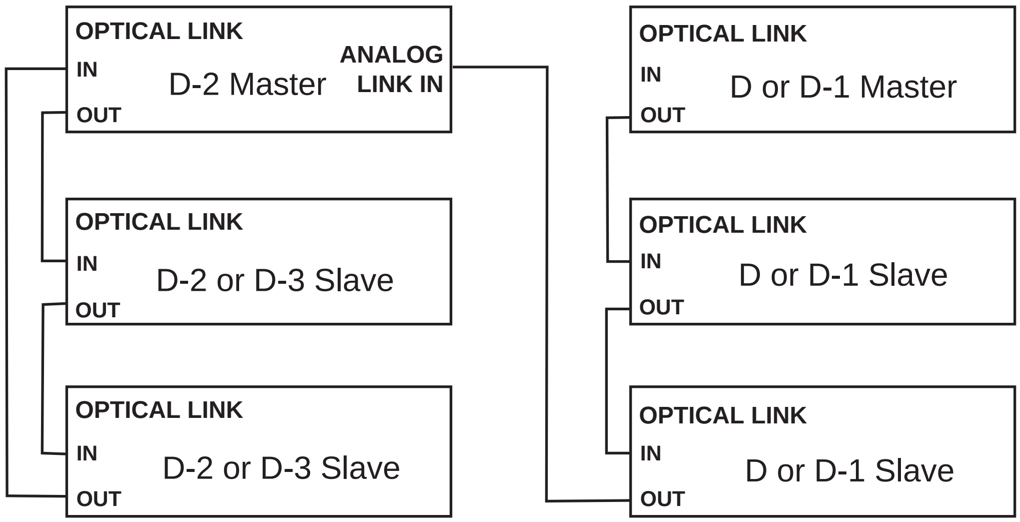

Figure 2-6 Linking analog and digital units

2.3.4 Level Matching Linked Analog Controllers

Model D controllers with serial numbers above 24 and all Model D-1s have a 7 dB higher **auto mix weight** level. In a system with the D-3, start out with the D-3 weight controls at 0, and the **threshold** or **weight** controls of the analog models at -7 dB. Then adjust for balanced attenuation in all channels when no one is talking.

Chapter 3: Operation

3.1 Preset Modes

Set the unused channels to the **mute** mode. Use each channel's **preset** button to set the preset mode lights to match the modes of the channels.

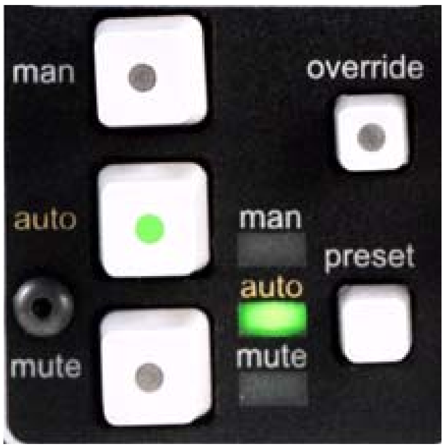

Figure 3-1 Channel (left) and Master (right) preset buttons

The preset indicators should mirror your normal working combination of input modes. The normal condition can then be restored by pressing the master **preset** button. This is also the state in which the system powers itself up.

3.2 Operation Overview

Figure 3-2 shows one channel of the D-3 control panel with the master section.

Figure 3-2 Channel and Master sections

The Dugan Model D-3 Automatic Mixing Controller has three patented and trademarked automatic mixing functions: the Dugan Speech System™, the Dugan Music System™, and Dugan Gain Limiting™.

The Dugan Speech System is the best choice for "talking heads." The Dugan Music System and Gain Limiting are usually used together for vocalists or musical instruments. The Dugan Music and Speech Systems are independent: one group of mics can be processed by the Speech System with another by the Music System. Gain Limiting can be turned on for any combination of channels.

In addition to these main functions the D-3 provides the following useful amenities:

- Pressing the **manual**, **auto**, and **mute** buttons causes a smooth audio transition for each channel between modes.
- The momentary master **override** button fades up the channels whose **override** function is active and fades down the other channels. The automatic mix resumes when the button is released.
- The master **preset** button sets each channel to the mode indicated by its **preset** mode LED. Powering up the system also performs this function.
- The **last hold** function is only functional for the Music System. It can be used to maintain constant ambience by keeping the last mic used at full gain until another mic is used.
- The D-3 can be divided into three **groups**, each an independent automatic mixing system.

3.3 Dugan Speech System

The Dugan Speech System distributes the gain of one open microphone over the entire system, maintaining a natural one-mic ambience. It is essential to distinguish this behavior from the annoying fluctuation of levels and uneven ambience in a conventional gating system.

When one person speaks into the Dugan Speech System, it rapidly fades that mic’s gain up and the others down. When the speaker pauses, all mics fade down so they sum to equal one mic at full gain. The result sounds like handing one mic around among the speakers. When several people talk at once, the gain is shared. This makes all mics sound normal when used but prevents buildup of noise or feedback.

3.3.1 How the Speech System Works

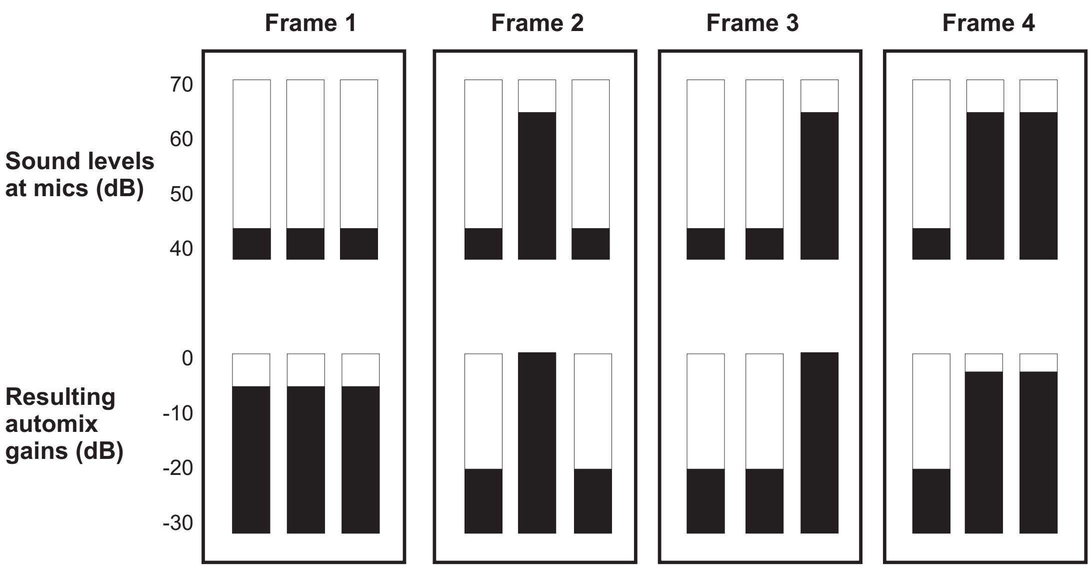

Figure 3-3

The Dugan Speech System automatically manages any number of live mics in an unpredictable dialogue situation. Figure 3-3 shows the Speech System in action, with four snapshots of a three-mic system under simulated conditions.

- The first frame shows no one speaking; the sound levels at all mics are low. The system fades all channels to a medium gain that sums to the equivalent of one mic at full gain.
- The second frame shows one person speaking. The system automatically fades his/her gain to full, while the other two inputs are turned down.
- The third frame shows a different person speaking. The system automatically fades his/her gain to full, while the other two inputs are turned down.
- The fourth frame shows two people speaking simultaneously. The system automatically shares the gain between them, while the other input is turned down.

3.3.2 Auto Mix Weight

The **auto mix weight** controls set the relative sensitivity of the automatic mix for channels in use. When the **auto mix weight** controls are balanced, each mic has an equal opportunity to take over the system: when one person talks into one mic, he/she gets all the gain and the others get turned down.

It is important to understand that the Speech System works by detecting the ratios of the levels between channels, not their absolute levels. The **auto mix weight** control is not a gate threshold!

The following example explains how **auto mix weight** works:

If all mics are the same type and the console's input trim controls and fader levels are identical, raising the **auto mix weight** control for one channel (Frame 1 in Figure 3-4):

- increases that channel's **auto mix gain** display during ambience and decreases it slightly for the others;
- makes it more difficult for others to speak when someone speaks into the channel with the higher **auto mix weight** setting.

  It is important to understand that this does not set that mic's level in the mix when that person speaks alone, but only its relative level when others speak simultaneously.

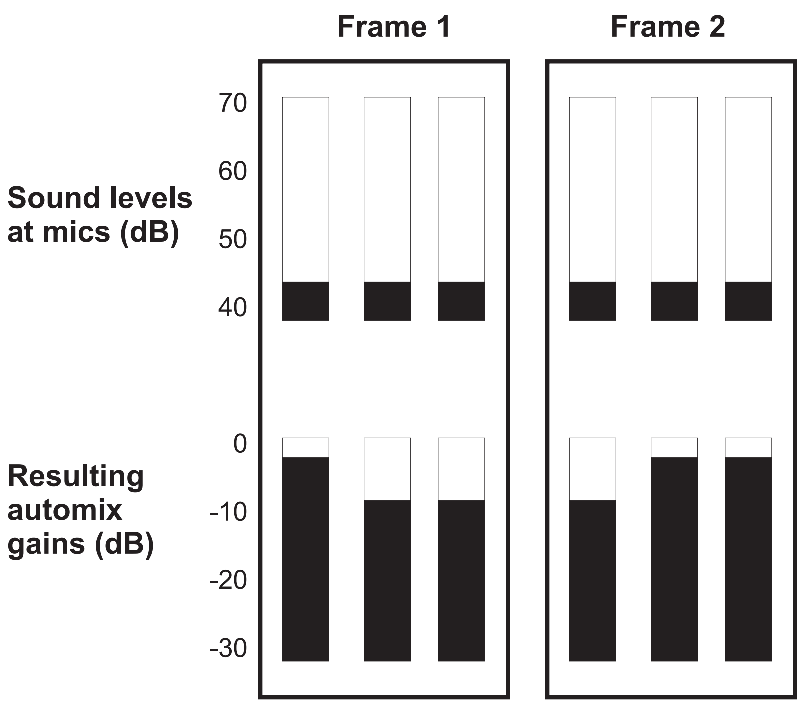

Figure 3-4 Frame 1: Raising **auto mix weight**; Frame 2: Decreasing **auto mix weight**

Lowering the **auto mix weight** control for one channel (Frame 2 in Figure 3-4):

- decreases the **auto mix gain** display during ambience for that channel and increases it slightly for the others;
- makes it more difficult for that channel's talker to be heard over the others.

For optimal performance, balance the **auto mix weight** controls so the channel gains display approximately equally when no one is talking. The only exception is if there is ongoing noise near one mic like a computer fan or air vent, it can be suppressed by reducing that channel's **auto mix weight**.

3.3.3 Setting up the Speech System

1. Put all the live mic channels in **auto** mode.
2. Activate the **mute** buttons for all unused channels.
3. For all unused channels, press the **preset** button until **mute** mode is selected.
4. De-activate the **music system** button.

   This puts the channel in the Speech System.
5. Set all the **auto mix weight** controls to 0.
6. Adjust the console input trim controls to the highest possible gain without clipping, assuming a normal speaking volume.

   There should be enough gain for the **level** LED to remain green during silences.
7. Set the console faders to your normal operating positions.
8. Balance the **auto mix weight** controls so the ambient noise registers approximately equally on the **auto mix gain** displays.

   Flickering is normal in the ambient noise condition, but try to get the channels to average about the same gain on the displays. Note that raising the **auto mix weight** of one channel causes its gain to rise and the others to fall. It's a balancing act. Now all your mics are set for equal access to the system gain.
9. When the event begins, you may discover that some mics must be set to lower gain to accommodate louder voices. Turn down the console's input trim controls to avoid clipping, then when no one is talking, turn up the **auto mix weight** of that channel to re-balance the **auto mix gain** displays.

NOTE: *If the* **auto mix weight** *controls are turned down too far, the inputs drop below the minimum computing level, the gains expand downward, and the green* **level** *LED is not lit. THIS IS NOT DESIRABLE! If you want gating, use the Music System.*

Figure 3-5 During ambience with 4 mics, **auto mix gain** displays should hover around -6 dB

Figure 3-6 During ambience with 8 mics, **auto mix gain** displays should hover around -9 dB

TIP: *When there is an unwanted noise in the mix, use the gain displays to locate the offending channel and activate its* **mute** *button.*

3.3.4 Pre-fader Speech System Insert

The Speech System works best if patched post-fader. If this is not possible, this section discusses some of the operational issues you will encounter. One must use different techniques to accomplish some familiar tasks, such as muting channels and pre-listening to muted channels.

Muting Channels

Mics must be muted by using the Dugan's **mute** buttons. Pulling a fader down on the console will not properly mute a channel. Although that mic is no longer audible in the mix, it still contributes to the gain computations and causes ambient noise fluctuation. In the worst case, this could cut off a speaker.

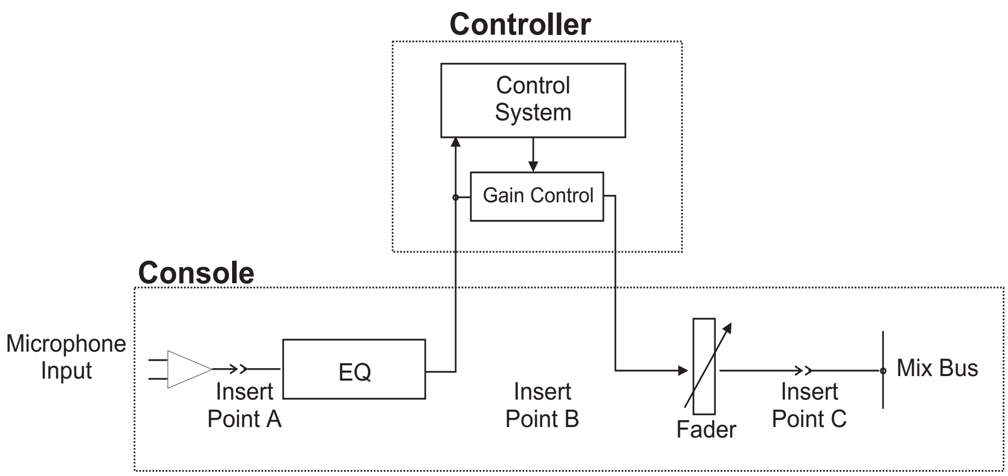

Figure 3-7

To properly mute a channel:

- Leave the console faders up, adjust levels while people are talking, and mute a mic by pressing the Dugan's **mute** mode button. Enable the mic when needed by putting it back into **auto** mode.

  Muting mics does not alter the overall ambient sound mix: the Speech System compensates by slightly raising the ambient gains of the other mics to compensate for the gain subtracted by muting a mic. Note that this behavior is during ambience without input to any mic; while one or more mics receive speech input, the gain shifts to the active mics.

  OR

- Use the **bypass** switch with the console fader pulled down. This keeps the mic instantly available on the fader but that channel is no longer in the control mix.

  Be aware that **bypass** may generate a click if it interrupts room rumble, whereas the **mute** button does a quick-fade. The **mute** button can be used with the fader up; **bypass** is best used with the fader down.

Pre-listening to Muted Channels

If your board allows listening to a mic before the insert point, you can mute on the Dugan. If signal through the Dugan is required to pre-listen to a mic, pull the fader down and put the channel in **bypass** mode. When the person with that mic is about to speak, switch the channel out of **bypass** and raise the fader.

NOTE: *A known bug in the Model D-3 system is that a short interruption in power can confuse the Processor and display an error message in the Processor window. If this happens, turn the power switch off, mix manually for ten seconds, and then turn it on again. Normal operation will resume.*

3.4 Dugan Music System

The Dugan Music System is a soft-gating or ducking system with an automatic threshold that follows the ambient noise level. Each channel has a 2:1 expansion ratio below the floating threshold, which is controlled by line level audio signal sent to one of the **music system threshold inputs** (**MT IN**). This signal is needed only for the Music System because the Speech System has no threshold. This signal can be derived from a sensing mic in the stage area.

Install the sensing mic as near the program mics as possible, but not close to a voice or instrument. We recommend using the same type for the sensing mic as the program mics. The goal is to represent the ambient sound detected by the mics on stage.

- Do not point the sensing mic at a pit orchestra, but rather in the same direction as the stage mics.
- Do not put the sensing mic in the back of the house because of the time delay.

Patch the preamplified sensing mic to the **MT IN A** of the Processor. Typically, the direct output of a console input strip is used, with the mix buses deselected. This input normally should be set to 10–15 dB more gain than the program mic inputs.

Another use for the Music System is to duck an audience mix when the people on stage talk. In this application, an aux send of the vocal mix is the threshold signal.

3.4.1 How the Music System Works

The following example illustrates the function of the Music System with three background vocalists.

- Frame 1 (Figure 3-8) shows no one singing. The system keeps all channels at a low gain.
- Frame 2 shows one person singing. The system automatically fades his/her gain to full, while the other two inputs stay down.
- Frame 3 shows a different person singing. The system automatically fades his/her gain to full, while the previous singer's mic fades down and the other input stays down.

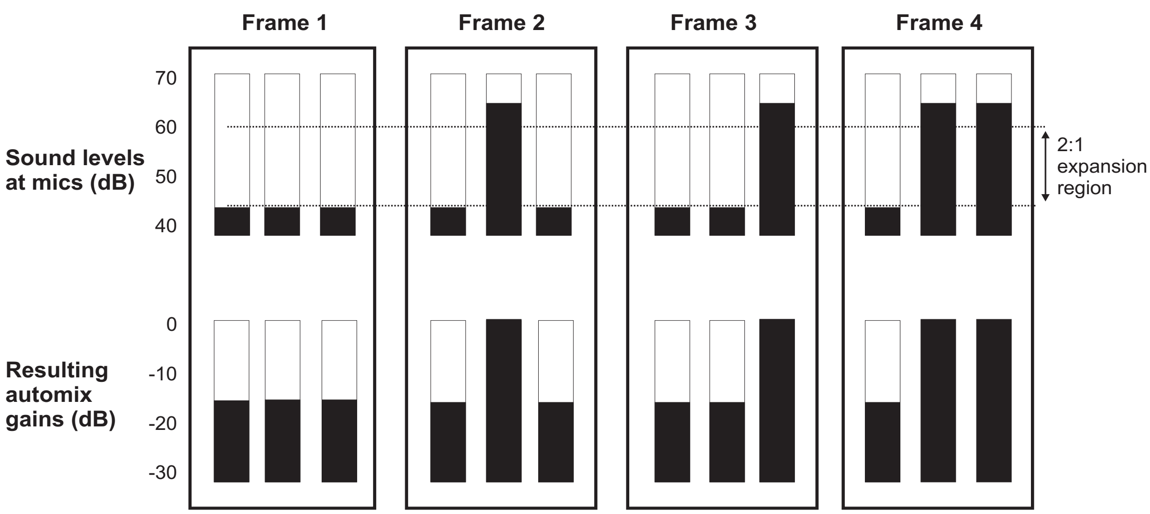

Figure 3-8

- Frame 4 shows two people singing together. The system automatically gives their channels full gain, while the other input stays down.

3.4.2 Setting up the Music System

The Music System works best when patched post-EQ pre-fader so that fader and threshold adjustments remain independent.

To place all desired channels in the Music System:

1. Activate the desired channels **music system** buttons.
2. Activate the **auto** buttons for all live channels.
3. Activate the **mute** buttons for channels not in use.
4. Press the **preset** button to select **auto** mode for the channels in use, and **mute** mode for the unused channels.
5. Adjust the **auto mix weight** for each channel so the bottom LED of the **auto mix gain** display flickers.

The master **music system threshold input** pot controls all the channel thresholds, as does the sensing mics's input gain trim control. If all the channel **auto mix gain** displays stay at full gain, the threshold signal is missing or too low.

If not using the **NOM** (Gain Limiting) function, it is OK to mute the channels on the board. However, if **NOM** is on, mute on the controller otherwise noise at muted mics can cause unwanted attenuation of the active mics.

The Music System does not maintain constant ambience like the Speech System. If gating of the ambience is annoying, the effect can be smoothed two ways:

- Keep one mic always open by turning **last hold** on: press both the master **override** and **preset** buttons at once. Their LEDs blink to indicate when **last hold** is on; toggle it off the same way. The **last hold** function keeps the last mic on that stays above -3 dB gain for 1/2 second. When another mic passes the same criteria, it becomes the **last hold** mic.

  OR

- Reduce the maximum attenuation depth by turning the **depth** knob clockwise to around -10 dB. This limits the range of downward expansion and mixes in a steady ambience.

NOTE: *The Music and Speech Systems operate independently but the* **depth** *control is global to both. Use different groups if necessary.*

3.5 Gain Limiting

The acoustic gain increases as each mic comes on, whether this occurs manually by placing that channel in **man** mode or automatically by using the Music System. Gain Limiting manages new mics entering the system to avoid overwhelming the mix with feedback or noise. Gain Limiting is controlled by the **NOM** (Number-of-Open-Mics) channel buttons and the master **NOM gain limit** pot. Since the Model D-3 has no master audio channel, Gain Limiting reduces the gain on all input channels for which **NOM** is active. Gain Limiting is the only automatic mixing function that can be active when the channel is in **man** mode.

Gain Limiting doesn't just count how many mics are on; it actually sums the gains of all the channels and compares them to a threshold. This also accounts for partially attenuated channel gains. Gain Limiting sums the gain for all channels with **NOM** active and reduces the gain to keep it below the threshold.

Conventional automatic mixing implementations always assume an NOM of 1. This can be effected on the Dugan by setting the **NOM gain limit** pot to *1*, which maximizes gain for each mic. Activate the **NOM** switches for each channel you wish to be included in the gain computation. The Master Gain Limiting display shows 3 dB of gain reduction when two mics are on, 6 dB of reduction with four mics on, etc.

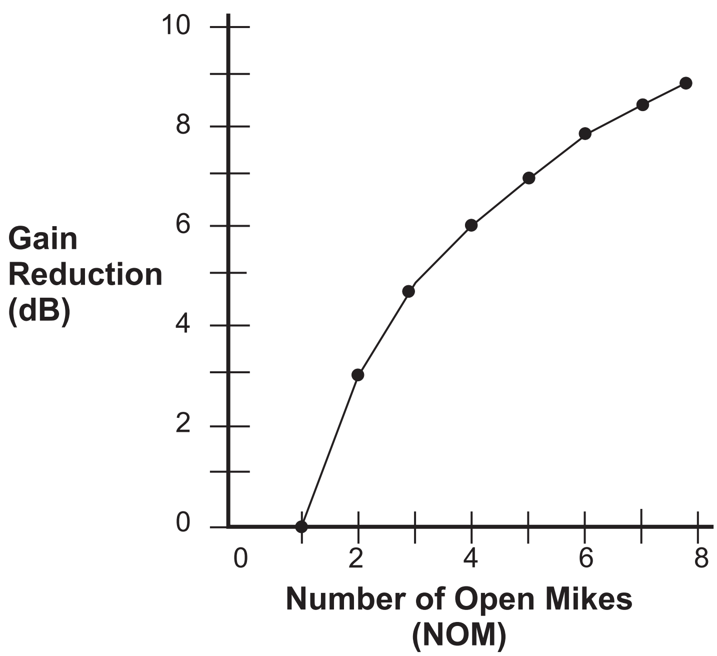

Figure 3-9

The Dugan offers the unique function of allowing 2–10 mics to activate before gain limiting occurs. Set the **NOM gain limit** pot to the desired number. For example, if there is enough gain to tolerate four open mics, set the **NOM gain limit** to four. Gain reduction begins only when the fifth mic turns on.

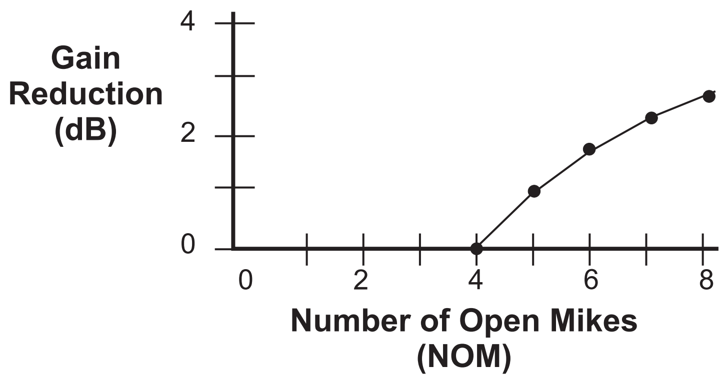

Figure 3-10 NOM is 4

3.5.1 Bypass and Mute

When Gain Limiting is active, we recommend using the **bypass** switch with the fader down to pass signal through the Dugan to preview mics on the console. This sends the signal through without activating the gain limiter.

Gain Limiting may be used with manual mic switching. Press the **mute** buttons for all channels. With **NOM** active, switch one, two, four, and eight mics to manual. Note the appropriate master gain reduction on the master display.

3.5.2 Gain Limiting and the Speech System

The Speech System always maintains NOM = 1, which prevents Gain Limiting from sensing significant excess gain. It normally does not matter how the **NOM** controls are set when using the Speech System. However, to avoid feedback when mics are switched to manual or the master **override** button is pressed with multiple mics selected, activate the NOM switches and set the **NOM gain limit** pot to 1.

If the unit has channels set to both Speech and Music Systems, turn the **NOM** switches *off* for the Speech System channels so the master gain reduction of the Music System mics does not affect those in the Speech System.

3.6 Groups

The Model D-3 uses three groups to implement the functions of three independent automatic mixers. Inputs may be assigned to any of three processing groups, each with its own **depth**, **NOM gain limit**, and **music system threshold input** controls. Grouping continues through linked units, except that analog mixers linked with a Model D-2 are always assigned to **group a**. The only functions that do not split into groups are the master **override**, **preset**, and **mute** buttons, which always work on the whole system.

Groups may be used to split controllers between simultaneous events in different rooms. Assign the mics for each room to their own group.

If mics are assigned to different console busses (like LCR), assign each mic on the D-3 to a group corresponding to its console bus. Otherwise the ambience will shift around in the audio image when different people talk.

3.7 Logic Functions

3.7.1 Override Function

The **override** button in the master control section fades all the channels full up or mute, depending on the states of the **override** switches for the individual channels. Override is a momentary function, active only while the master **override** button is held down.

- Activate the desired channel's **override** switch (LED is lit) so these channels come fully on when the master **override** button is held down.
- Deactivate the desired channel's **override** switch (LED not lit) so these channels are muted when the master **override** button is held down.

The **override** button may be accessed remotely so a chairperson can silence all other mics.

3.7.2 Preset Function

The master **preset** button sets the modes (manual, auto, mute) of the channels to the setting on the channels preset LEDs. Use these switches to store your most commonly used pattern of channel usage, which may be restored by pressing the **preset** button. The channels are in their preset modes when the unit is powered up.

3.7.3 Mute Function

The master **mute** button is a momentary function which fades out all channels while held down. It may be accessed remotely so a chairperson at the podium can shut off all mics.

3.7.4 Bypass Mode

Each channel has a bypass mode. A channel may be patched directly through the Dugan with the **bypass** switch. The bypassed channel's displays lights extinguish and the **bypass** LED illuminates. All channels are hard-wire bypassed when the power is off so it is not necessary to unpatch the Processor when not in use.

NOTE: *After all controls have been set, the Control Panel can be removed. The Processor remembers all settings, even after the power is turned off.*

Chapter 4: Remote Control

There are four methods for controlling the Model D-3 Processor:

- the D-3 Control Panel (see Chapter 3: *Operation*);
- by contact closures at the rear panel **REMOTE** connector (page 41);
- by ASCII commands to the RS-232 **HOST** connector (page 42);
- by interfacing a virtual control panel via the RS-422 **CTRL** connector. A virtual control panel can include both controls and displays (contact the factory for assistance).

In any case, the Processor continues to perform automatic mixing if the Control Panel is disconnected.

4.1 Remote Connector

The Processor's back panel **REMOTE** connector provides:

- Mute inputs (contact closures pulling to ground) for the input channels.
- Contacts for master functions Override and Mute. Override and Mute are momentary the same as on the Control Panel.
- Logic outputs for the channels. Contacts are pulled to ground when channel gain is above -3 dB. See the next section for details.

4.1.1 Logic Output

The model D-3 has one logic output per channel that is *true* (low) when the gain of an individual channel is high (above -3 dB). This may be used to switch off speakers near the mic, or to switch to a different camera. The logic output normally has about 0.5 s delay when turning from *false* to *true*. For camera switching, the delay prevents inadvertent switching from momentary noises like paper flaps. Since instantaneous response is required for speaker switching, contact the factory to learn how to adjust the software for this mode. The logic output is an open collector transistor switch, grounded when active. There is an internal 51 kΩ pull-up to 5 V.

To use the logic outputs, we recommend using the Music System for positive action. However, to use the logic outputs with the Speech System, the following considerations apply:

- Since the Speech System's logic output flickers when a person speaks, it cannot be used directly for control. Users have successfully programmed Crestron systems to convert the Dugan's momentary outputs to the necessary held signals for camera control.

- The logic output is *true* (low) whenever the gain of a channel is above the logic threshold, set at the factory at -3 dB. If you are using the logic outputs, you may need to adjust the logic thresholds (internally) depending on the number of mics in the system. If there are only two mics in auto mode, each mic's average gain during ambient noise would be around -3 dB, so the logic outputs would both be *true* much of the time. In this case the logic threshold should be set to -1 or -0.5 dB.

  Conversely, in a system with 16 or more mics and high gain to speakers in the room, feedback may prevent channels from coming all the way up. If the logic threshold is set to -3 dB, it may not activate when someone speaks. This case might require setting the logic thresholds down to -4 or -5 dB. Contact the factory to set the threshold to the optimal level for your installation.

4.2 RS-232 Remote Control Commands

4.2.1 Introduction

The Model D-3 provides an RS-232 link to control the unit remotely with third-party control systems. Every control available on the Control Panel, except the knobs, can be accessed with the RS-232 link. The RS-232 transmission protocol is: 38400 baud, 8-bit, no parity, and 1 stop-bit. A straight serial cable (not a null-modem cable) is required to operate from a PC.

Two command formats are supported:

- ASCII text format extended with a binary byte at the beginning and a binary check-sum byte at the end. The check-sum enhances the reliability of the transmission.

  0xFE ID Command <CR> check-sum

  0xFE means an 8-bit byte with the hexadecimal value FE, 254 in decimal, 0111 1111 in binary (sent left to right).

- ASCII text format:

  ID Command <CR>

  This format is appropriate for quickly testing a command from a terminal. We recommend using the extended format for highest reliability.

In both formats, *ID* is the Processor's serial number, which can be found on the serial number plate on the rear panel (ignore leading 0s), or in the Factory page of the LCD front panel. The Factory page can be displayed by poking a small screwdriver or a paper clip once into the small hole on the bottom-left corner of the keypad. The *Command* is a text command string (7-bit ASCII characters only). *<CR>* is the carriage return character (ASCII value 13). Each command requires a carriage return at the end of the line. We will not show this in the examples.

The second format requires two extra bytes. The first byte is 0xFE (254 in decimal) and is the only 8th-bit-set number accepted by the Model D-3. The second extra byte is the *check-sum* byte that is the lower 7 bits of the 8-bit sum of all the bytes from 0xFE to <CR>, inclusive.

The two formats may be mixed. Whenever 0xFE is detected, the D-3 automatically switches to the extended format. When an extended format packet is sent, the reply is also in the extended format. All white spaces and tab characters are automatically removed before parsing and command strings can be in either upper or lower cases. The following characters result in the same command:

1NE3,1

1 Ne 3, 1

1n E3, 1

The Model D-3 replies to every serial command. The format of the reply echoes the command. For example, if **1NE3,1** is sent from the user, the reply from the D-3 when the command is successfully executed is: **1NE3,1**

If **1NE3,** is sent from the user, the reply is **1NE Syntax error,** indicating a syntax error in the command (parameter value missing).

4.2.2 Commands

This section describes all the available command strings for both formats. The following abbreviations are used:

id: Processor serial number, not including leading 0s

ch: 1–8

grp: 1–3 for groups A–C.

Each command has the form:

***<id> <command> <group/channel> <value*>**

The command string will have a group or a channel, but not both.

The following examples are in plain ASCII text format but also apply to the extended format. We use spaces for clarity but they are not necessary and are not present in the replies. The commas are a required part of the command strings.

Auto Mix Depth

This read-only command reports the Auto Mix Depth for the given group in multiples of 0.01 dB. For example:

	Example	ID	Command	Group	Value
Sent	15 AD 2	15	Auto Mix Depth	B	
Reply	15 AD 2, -300	15	Auto Mix Depth	B	-300 = -3.00 dB

The Auto Mix Depth cannot be changed via the serial link since it follows the knob position on the Control Panel.

Channel Bypass

Use this command to query or set the given channel's bypass mode. For example:

	Example	ID	Command	Channel	Value
Sent	15 BP 2	15	Bypass Mode (query)	2	
Reply	15 BP 2, 0	15	Bypass Mode (response)	2	0 = not in Bypass Mode

	Example	ID	Command	Channel	Value
Sent	15 BP 2, 1	15	Bypass Mode (set)	2	1 = Activate Bypass Mode
Reply	15 BP 2, 1	15	Bypass Mode (response)	2	1 = Bypass Mode Active

Channel Mode

Use this command to query and set the mode of the given channel. The numbers 1–3 corresponds to Manual, Automatic, and Mute channel modes, respectively.

For example:

	Example	ID	Command	Channel	Value
Sent	15 CM 2	15	Channel Mode (query)	2	
Reply	15 CM 2, 2	15	Channel Mode (response)	2	2 = Automatic Mode

	Example	ID	Command	Channel	Value
Sent	15 CM 2, 3	15	Channel Mode (set)	2	3 = Activate Mute Mode
Reply	15 CM 2, 3	15	Channel Mode (response)	2	3 = Mute Mode Active

Channel Override

Use this command to query and set a given channel's **override** status. For example:

	Example	ID	Command	Channel	Value
Sent	15 CO 2	15	Channel Override (query)	2	
Reply	15 CO 2, 0	15	Channel Mode (response)	2	0 = Channel Override disabled

	Example	ID	Command	Channel	Value
Sent	15 CO 2, 0	15	Channel Override (set)	2	0= Disable Channel Override
Reply	15 CO 2, 0	15	Channel Mode (response)	2	0 = Channel Override disabled

Channel Weight

This read-only command reports the channel **auto mix weight** for the given channel in multiples of 0.01 dB. For example:

	Example	ID	Command	Channel	Value
Sent	15 CW 2	15	auto mix weight (query)	2	
Reply	15 CW 2, -300	15	auto mix weight (response)	2	-300 = -3.00 dB

You may not change the channel weight from the serial link since the weight must follow the knob position on the control head.

Factory Preset

This command, like the **RESET** command on the LCD front panel, resets the unit to its factory presets. For example:

	Example	ID	Command	Channel	Value
Sent	15 FP	15	reset to factory presets		
Reply	15 FP ok	15	response		ok = reset accomplished

Group Assign

Use this command to query and assign a channel's group assignment. For example:

	Example	ID	Command	Channel	Value
Sent	15 GA 2	15	Group Assignment (query)	2	
Reply	15 GA 2, 3	15	Group Assignment (response)	2	3 = Group C

	Example	ID	Command	Channel	Value
Sent	15 GA 2, 2	15	Group Assignment (set)	2	2= Group B
Reply	15 GA 2, 2	15	Group Assignment (response)	2	2 = Group B

Last Hold

Use this command to query and set the **last hold** function. For example:

	Example	ID	Command	Channel	Value
Sent	15 LH	15	last hold (query)		
Reply	15 LH 0	15	last hold (response)		0 = off

	Example	ID	Command	Channel	Value
Sent	15 LH 1	15	last hold (set)		
Reply	15 LH 1	15	last hold (response)		1 = on

Master or Slave Mode

Enable master or slave mode for the Processor. For example:

	Example	ID	Command	Channel	Value
Sent	15 MM	15	master or slave mode (query)		
Reply	15 MM 1	15	master or slave mode (response)		1 = master mode

	Example	ID	Command	Channel	Value
Sent	15 MM 0	15	slave mode (set)		
Reply	15 MM 0	15	slave mode (response)		0 = slave mode

Dugan Music or Speech System

Use this command to query or set the given channel to Dugan Music or Speech System. For example:

	Example	ID	Command	Channel	Value
Sent	15 MR 2	15	music system (query)	2	
Reply	15 MR 2, 1	15	music system (response)	2	1 = music system enabled

	Example	ID	Command	Channel	Value
Sent	15 MR 2, 0	15	speech system (set)	2	
Reply	15 MR 2, 0	15	speech system (response)	2	0 = speech system enabled

Music Threshold

This read-only command reports the gain of the **music system threshold input** for the given group in multiples of 0.01 dB. For example:

	Example	ID	Command	Group	Value
Sent	15 MT 2	15	speech system (set)	2	
Reply	15 MT 2, -300	15	speech system (response)	2	-300 = -3 dB

The Music Threshold cannot be changed from the serial link since it must follow the knob position on the Control Panel.

NOM Function

This command queries the current NOM function of the given channel. For example:

	Example	ID	Command	Channel	Value
Sent	15 NE 2	15	NOM function (query)	2	
Reply	15 NE 2, 1	15	NOM function (response)	2	1 = enabled

	Example	ID	Command	Channel	Value
Sent	15 NE 2, 0	15	NOM function (set)	2	0 = disable
Reply	15 NE 2, 0	15	NOM function (response)	2	0 = disabled

NOM Gain Limit

This read-only command reports the **NOM gain limit** threshold for the given group in multiples of 0.01. For example:

	Example	ID	Command	Group	Value
Sent	15 NL 2	15	NOM gain limit threshold (query)	1	
Reply	15 NL 2, 100	15	NOM gain limit threshold (response)	1	100 = 1.00

The NOM limit threshold may not be changed from the serial link since it must follow the knob position on the Control Processor.

Channel Mode Preset

This command queries and sets the current channel mode **preset**. The *mode* can be 1, 2, or 3 corresponding to **manual**, **automatic**, or **mute**, respectively. For example:

	Example	ID	Command	Channel	Value
Sent	15 PS 2	15	channel mode preset (query)	2	
Reply	15 PS 2, 2	15	channel mode preset (response)	2	2 = auto mode

	Example	ID	Command	Channel	Value
Sent	15 PS 2, 3	15	channel mode preset (query)	2	3 = mute mode
Reply	15 PS 2, 3	15	channel mode preset (response)	2	3 = mute mode

Master Mute

This command queries and sets the master (system) mute. For example:

	Example	ID	Command	Channel	Value
Sent	15 SM	15	system mute (query)		
Reply	15 SM 0	15	system mute (response)		0 = system not muted

	Example	ID	Command	Channel	Value
Sent	15 SM 1	15	system mute (set)		1 = mute system
Reply	15 SM 1	15	system mute (response)		1 = system muted

System Override

This command queries and sets the master (system) override. For example:

	Example	ID	Command	Channel	Value
Sent	15 SO	15	master override (query)		
Reply	15 SO 0	15	master override (response)		0 = override off

	Example	ID	Command	Channel	Value
Sent	15 SO 1	15	master override (set)		1 = override system
Reply	15 SO 1	15	master override (response)		1 = override on

System Preset

This command sends a master (system) preset command causing all channels to go to their channel preset mode. For example:

	Example	ID	Command	Channel	Value
Sent	15 SP	15	master preset (set)		
Reply	15 SP ok	15	master preset (response)		ok = executed

Chapter 5: Model D-3 Specifications

Control Panel	
Connectors	Two five-pin male XLRs for combined control and power cable. Cable may be connected to either the left side or the rear of the Control Panel.
Dimensions	3RU **H** = 2.5 in (6.4 cm) **D** = 5.0 in (13.0 cm) **W** = 17.0 in (43.0 cm) 19 in (48 cm) with rack ears
Weight	7.2 lb (3.25 kg) 9.1 lb (4.10 kg) shipping w/box and control cable
Processor	
Audio I/O	Eight AES/EBU digital inputs on four female XLR connectors. Eight AES/EBU digital outputs on four male XLR connectors
Gain	unity
Sample Rate	**Input**: 32 kHz – 96 kHz (integral sample rate conversion) **Output**: 48 kHz
Bit Depth	24 bits
Audio Latency	1.2 ms
Frequency Response	10 Hz to 22 kHz, ±0.0075 dB
Output Noise	-125 dBFS (20 Hz to 20 kHz), -128 dBAFS
Distortion	-125 dB
Synchronization	Sources (48 kHz only) AES input pair 1/2 (default), 3/4, 5/6, or 7/8 Internal Word Clock in DARS input Word Clock Output 48 kHz
Linking	optical ring, up to 8 units (64 channels)

Connectors	**Audio** AES/EBU XLR (female inputs, male outputs) **DARS input** XLR female **Word Clock I/O** BNC coaxial **Control Panel** 5-pin female XLR **10 BASE T** RJ-45 **Host RS-232** DB-9 **Linking** ADAT optical **Remote (Logic I/O)** DB-25 **Power** plug-in screw compression terminal strip
Logic Inputs	Eight channel mutes, System Override, System Mute (all momentary, while held). TTL, active low (contact to ground).
Logic Outputs	Eight channel active flags. Active low, maximum applied voltage 30 V, maximum current sinking 300 mA momentary, 100 mA sustained, inductive load permitted. Internal 51 kΩ pull-up to 5 V.
Power	12 W; 9–24 VDC either polarity, or 9–18 VAC. Supplied power supply unit: input 90–264 VAC, 47–63 Hz, output 12 V, 5 A Approved UL, CSA, TUV
Dimensions	2RU **H** = 3.5 in (8.8 cm) **D** = 10.0 in (25.0 cm) **W** = 19.0 in (48.0 cm)
Weight	11.5 lb (5.2 kg) 14.5 lb (6.5 kg) in shipping box with power supply 23.6 lb (10.6 kg) entire system in two shipping boxes

Chapter 6: Connector Pinouts

Table 6-1 AES digital audio inputs (female XLR): 4 channel pairs, 2 Music System Threshold, DARS

Pin	Signal
1	Ground (chassis)
2	Signal in +
3	Signal in -

Table 6-2 AES digital audio outputs (male XLR)

Pin	Signal
1	Ground (chassis)
2	Signal out +
3	Signal out -

Table 6-3 Control Panel (Female XLR, RS422 half-duplex data 38.4 kHz)

Pin	Signal
1	Ground
2	Data +
3	Data -
4	5 V power
5	Ground

Table 6-4 Host (Female DB9, RS-232)

Pin	Signal
1	DCD in
2	DSR in
3	TXD out
4	RTS out
5	RXD in
6	CTS in
7	DTR out
8	RI in
9	GND

Table 6-5 10 BASE T (Ethernet, TCP/IP, to be implemented as needed)

Pin	Signal
1	T2
2	T1
3	R2
4	n/c
5	n/c
6	R1
7	n/c
8	n/c

Table 6-6 LOGIC (DB25 logic I/O)

Pin	Signal
1	Mute in 1
2	Mute in 2
3	Mute in 3
4	Mute in 4
5	Mute in 5
6	Mute in 6
7	Mute in 7
8	Mute in 8
9	Master Mute in
10	Override in
11	n/c
12	n/c
13	Ground
14	Logic out 1
15	Logic out 2
16	Logic out 3
17	Logic out 4
18	Logic out 5
19	Logic out 6
20	Logic out 7
21	Logic out 8
22	n/c
23	n/c
24	n/c
25	Ground