Model 236
Announcer's Console

User Guide

Issue 8, June 2025

This User Guide is applicable for serial numbers
M236-00151 and later with Main Firmware version 3.09 and later
and STcontroller software application version 4.04.00 and later

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Revision History

Issue 8, June 2025:

- Documents addition of Any Talkback Active Dante Aux Output mode.
- Miscellaneous improvements and corrections.

Issue 7, January 2023:

- Documents revised encoder configuration and operation.
- Miscellaneous improvements and clarifications.

Issue 6, March 2022:

- Documents addition of Main – Encoder Action configuration function.

Issue 5, October 2021:

- Documents change to Dante Aux Output operation when Hot Mic is selected.

Issue 4, May 2021:

- Documents new analog microphone input limiter function.
- Documents revised configuration name from LED Intensity to LED Intensity and Action.
- Documents two LED Intensity and Action modes.

Issue 3, October 2020:

- Revises text to reflect Audinate's Inclusive Language Guidelines. The word "master," in reference to Dante® clocking, has been changed to "Leader."

Issue 2, September 2020:

- Documents changes to Main 1 and Main 2 inputs and associated headphone monitoring.
- Documents addition of Sidetone Channel Routing feature.
- Adds technical note regarding level attenuation in Dante Aux and Talkback transmitter (output) channels when Analog Mic Output is configured for Switched.

Issue 1, August 2020:

- Initial release.

Introduction

The Model 236 Announcer's Console provides a combination of high-performance audio with an extensive set of user features and configurable resources. The unit is intended for use in demanding broadcast sports, eSports, live event, entertainment, and streaming broadcast applications. With a few simple connections the Model 236 can provide one on-air talent with all the resources they need to support a wide range of applications. The compact, rugged steel enclosure is intended for tabletop use, small enough for use "court-side" or in a crowded booth, yet flexible enough for remote production deployment. The Model 236 supports Dante® audio-over-Ethernet digital media technology with AES67 compatibility for integration into contemporary applications.

The Model 236 can integrate directly into both Dante audio-over-Ethernet and standard analog audio environments. With just a Power-over-Ethernet (PoE) Gigabit network connection, a microphone, and a pair of headphones a complete broadcast on-air position can be created. With the addition of a second Ethernet connection Redundant Dante can be supported. And by using the Model 236's analog microphone output a connection to a microphone-level input on an associated remote I/O interface or audio console can be supported.

With ten Dante audio inputs and an integrated sidetone function, users can easily create their desired headphone audio mix. They'll enjoy clear and "click-free" monaural or stereo audio that can only help their on-air performance. Six independent Dante talkback channels allow users to communicate with a variety of support personnel. Among the range of operating capabilities includes the ability to create talent cue (IFB) channels. This was specifically included for REMI/At-Home applications. Provision has been made such that a technician can "customize" the Model 236's hardware. Option kits, purchased separately, can be added to support functions such as multi-pin headset connectors, remote control inputs, and a low-voltage, low-current DC tally output.

The Model 236 was designed to meet two main goals: supporting great audio quality and providing an extensive set of configurable features. Using the latest in audio integrated circuits and advanced 32-bit audio processing, the unit's audio performance should meet or exceed that of any audio console, standalone microphone preamplifier, remote I/O interface, or outboard A/D or D/A converter. With over 40 years of professional audio experience, Studio Technologies takes audio performance seriously! And while providing excellent technical specifications is a "must," a device also has to "sound" good before we feel its design is complete.

Figure 1. Model 236 Announcer's Console front and back views

The amount of flexibility provided in the Model 236 allows it to meet the needs of virtually all on-air announcer applications. And using the Studio Technologies' STcontroller software application makes "customizing" the operation of a Model 236 fast and simple. The unit's ability to handle both day-to-day and specialized situations makes it a unique product in the market. For example, the push-button switches and rotary encoders can be independently configured with multiple choices that range from simple to quite advanced. If a Model 236 can't seem to be configured to meet an application's goals please contact Studio Technologies' technical support for an application review.

The Model 236 is part of a three-product family that differ only in the number of talkback channels provided. The Model 236 provides six talkback channels, while the Model 232 provides two and the Model 234 provides four. The Model 236, as with the other two models, has a compact enclosure with overall dimensions of 6.5 inches wide (16.5 cm), 2.9 inches high (7.4 cm), and 4.9 inches deep (12.5 cm). Weighing 2.1 pounds (0.95 kg), the enclosure is made of steel to provide some "heft" to minimize the chance of inadvertent movement. The Model 236's main, secondary, and FPGA (programmable logic) firmware can be updated using the USB port on the back of the unit; the Dante firmware can be updated via an Ethernet connection.

Setup, Configuration, and Operation

Set up, configuration, and operation of the Model 236 is simple. The unit includes two Neutrik etherCON RJ45 connectors which allow interfacing with single or redundant Gigabit Ethernet networks. The primary network connection can provide power to the unit using a port on a Power-over-Ethernet (PoE) network switch. Model 236's power can also be supplied from a 12 volt DC source that is connected using a 4-pin XLR connector. A broadcast headset or handheld ("stick") microphone can be directly connected to the unit's 3-pin female XLR microphone input connector. The input is compatible with dynamic or condenser microphones. A P48 phantom power source allows support for a wide range of condenser microphones. A 3-pin male XLR connector provides an analog microphone output for integration with inputs on related devices. A configuration choice allows this output to be active all the time (hot mic operation) or muted or unmuted following the main output function. A pair of stereo headphones, the headphone connection from a stereo or monaural headset, or even earbuds can be connected to the Model 236's headphone output jack.

Audio signals are routed to and from the Model 236 using the Dante Controller software application. This is available, free of charge, from Audinate®, the creators of Dante. All Model 236 operating features are configured using the Studio Technologies' STcontroller software application. The extensive set of parameters allows the unit's functions to be tailored to meet the needs of many, many applications. STcontroller is available at no charge from the Studio Technologies' website. Versions are available that are compatible with the Windows® and macOS® operating systems. STcontroller is a fast and simple means of configuring, revising, saving, and loading a unit's operating parameters.

The Model 236's front panel includes eight rotary controls (encoders) which are used to adjust the level of the Dante input signal sources as they create an audio mix that is fed to both the analog and Dante digital headphone outputs. Using RGB (red-green-blue) LEDs, each rotary encoder is illuminated and can display whether or not signal is present on their associated audio input channels. Two additional rotary controls allow adjustment of sidetone level and the overall level of the analog headphone output.

Seven pushbutton switches allow the user to control the status of the main outputs (digital and analog) as well as the six talkback functions. Extensive configuration choices allow the operation of the pushbutton switches and associated output channels to be optimized to meet the needs of specific applications. For example, the talkback functions can be independently configured for talkback, talent cueing (IFB), call signal (20 kHz tone), and other related actions.

The Model 236's one main and six talkback pushbutton switches are illuminated to display their operating status. For performance confirmation an integrated sidetone function allows audio coming from the headset's microphone to be returned to the headset output.

Audio Quality

The Model 236's audio performance is completely "pro." A low-noise, wide dynamic-range microphone preamplifier ensures that microphone audio quality is preserved while minimizes the chance of signal overload. The gain of the microphone preamplifier can be adjusted over a range of 20 to 65 dB in 1-dB steps. A digitally controlled analog limiter function allows the signal level to be automatically adjusted to reduce the chance of signal overload. A setting in STcontroller allows the limiter function to be enabled or disabled as desired. The limiter function applies only to signals that are connected to the analog microphone input.

The output of the microphone preamplifier is routed to an analog-to-digital conversion (ADC) section that supports sampling rates of 48 and 96 kHz. The audio signal, now in the digital domain, routes through a 32-bit microprocessor and associated programmable logic and then on to the Dante interface section where it is packetized and prepared for transport over Ethernet. Ten Dante transmitter (output) channels are provided: one main, one aux, six talkback, and two headphone monitor output channels.

Audio input signals arrive into the Model 236 by way of eleven Dante receiver (input) channels and pass into the Model 236's 32-bit logic circuitry. Four channels are associated with the main section, six are used for the talkback listen sources, and one can serve as an alternate microphone audio source. In addition to being used for typical headphone monitoring, the audio input associated with each talkback channel can be used as part of a talent cue (IFB) function. All channel routing, headphone level control, signal detection, and sidetone functions are performed within the digital domain. This preserves audio quality, enhances performance, and provides flexibility, allowing precise level control and mixing as well as eliminating the need for analog audio signals to pass through passive level controls (potentiometers). The audio signals destined for the analog headphone output are sent to a high-performance 2-channel digital-to-analog converter and then on to robust driver circuitry.

Dante Audio-over-Ethernet

Audio data is sent to and received from the Model 236 using Dante audio-over-Ethernet media networking technology. As a Dante-compliant device, the Model 236's ten Dante transmitter (output) channels and eleven Dante receiver (input) channels can be assigned (routed or "subscribed") using the Dante Controller software application. To minimize the chance of a flow limitation, the Model 236's Dante transmitter and receiver channels support 32 Dante flows, 16 in each direction. The digital audio's bit depth is up to 32 with a sample rate of 48 or 96 kHz. Bi-color LEDs, located on the unit's back panel, provide status indications related to the Ethernet network and Dante interface performance.

The Model 236 is compliant with the AES67 interoperability standard. In addition, the unit is compatible with Audinate's Dante Domain Manager™ (DDM) software application. DDM offers an enhanced set of network control and monitoring features, making it ideal for security-conscious applications. This may also allow the Model 236's audio channels to support SMPTE® ST 2110-30.

Ethernet Data and PoE

The Model 236 connects to one or two local area networks (LANs) by way of two Gigabit (GigE) twisted-pair Ethernet interfaces. These 1000BASE-T interconnections are made by way of Neutrik® etherCON RJ45 connectors. While compatible with standard RJ45 plugs, etherCON allows a ruggedized and locking interconnection for harsh or high-reliability environments. The two Ethernet interfaces can be configured, using the Dante Controller software applications, to serve in the Dante Switched or Redundant modes.

The Model 236's operating power can be provided by way of its Primary-PoE Ethernet interface using the 802.3af Power-over-Ethernet (PoE) standard. PoE allows fast and efficient interconnection with an associated data network. Alternately, an external source of nominal 12 volts DC can be connected to power the unit. If both are connected then PoE will serve as the active power source.

Future Capabilities and Firmware Updating

The Model 236 was designed so that in the future its capabilities and performance can be easily enhanced. A USB type A connector, located on the unit's back panel, allows the main, secondary, and FPGA firmware files to be updated using a USB flash drive. The Model 236 uses the Broadway™ integrated circuit from Audinate to implement its Dante interface. The firmware in this integrated circuit can be updated via an Ethernet connection helping to ensure that its capabilities remain up to date.

Getting Started

What's Included

Included in the shipping carton are a Model 236 Announcer's Console and instructions on how to obtain a copy of this guide. As a device that can be Power-over-Ethernet (PoE) powered, no external power source is provided. This is because in most applications an Ethernet switch with PoE capability will be utilized. It's also possible to connect an external source of 12 volts DC to power the Model 236. If this is the case then a power supply would need to be obtained separately.

Connections

In this section signal interconnections will be made using the connectors located on the back of the Model 236's enclosure. One or two Ethernet connections will be made using either standard RJ45 patch cables or etherCON protected RJ45 plugs. The Ethernet data connection associated with the Primary-PoE RJ45 connector is compatible with Power-over-Ethernet (PoE). If desired, an Ethernet connection made to the Secondary Ethernet connector can provide access to a Redundant Dante network. Or it can be utilized as an active "loop through" connection associated with the Primary-PoE Ethernet connection. If PoE is not going to be utilized, or a redundant source of power is desired, a source of 12 volts DC can be connected by way of a 4-conductor XLR connector.

A microphone can be connected using a cable-mounted 3-pin male XLR connector. A set of headphones, the headset portion of a broadcast-style headset, or an earpiece will be connected by way of a ¼-inch plug. If desired, the Model 236's analog microphone output may be interfaced with other equipment using a cable terminated with a standard 3-pin female XLR connector.

For special applications the Model 236's internal circuit boards contain 3-pin "header" connectors that allow access to various functions. In addition, there are two spare connector locations located on the Model 236's back panel. A technician can add connectors and wire them to the headers such that application-specific needs can be supported. For example, a 6- or 7-pin XLR connector can be added to support direct connection of broadcast-style headsets. Contact closures can be interfaced to the Model 236's circuitry, allowing external switches to activate the main and talkback functions. A low-voltage, low-current tally output associated with the main function is also available. Refer to the Technical Notes section for details.

One or Two Ethernet Connections

One 1000BASE-T Gigibit Ethernet (GigE) connection is required for Model 236 operation and can provide both Ethernet data and power by way of Power-over-Ethernet (PoE) for the Model 236's circuitry. A 10BASE-T (10 Mb/s) or 100BASE-TX (100 Mb/s) connection is not sufficient.

A second 1000BASE-T (GigE) connection can be made if Redundant Dante is desired. (For this functionality to be active the Model 236's network configuration must be set for *Redundant* within the Dante Controller software application.) PoE is not supported on the Secondary Ethernet connection. Again, a 10BASE-T or 100BASE-TX connection is not sufficient. The Secondary Ethernet connection can also be used as a "loop through" port such as would be provided by an Ethernet switch. (This requires that the Model 236's network configuration within the Dante Controller software application be set for *Switched*.) Switched mode is the default setting but using the Secondary port in this mode for applications other than troubleshooting is not recommended. The unit will function reliably but "daisy chaining" Ethernet signals can limit flexibility and present a failure point; it's best if each Ethernet interface connects directly to a port on an Ethernet switch.

The Ethernet connections are made by way of two Neutrik etherCON protected RJ45 connectors that are located on the back of the Model 236's enclosure. These allow connection by way of cable-mounted etherCON connectors or standard RJ45 plugs. The Model 236's Ethernet interfaces support auto MDI/MDI-X so that crossover cables are not required.

External 12 Volts DC Input

An external source of 12 volts DC can be connected to the Model 236 by way of a 4-pin male XLR connector which is located on the unit's back panel. While the requirement for the external source is nominally 12 volts, correct operation will take place over a 10 to 18 volts DC range. The Model 236 requires a maximum of 550 milliamperes (0.55 amperes) at 12 volts DC for correct operation. The DC source should be terminated on a 4-pin female XLR connector with pin 1 negative (–) and pin 4 positive (+).

If an external power source is required, the PS-DC-02 power supply, available as an option from Studio Technologies, is directly compatible with the Model 236. Its AC mains input allows connection to 100-240 volts, 50/60 Hz with an output of 12 volts DC, 1.5 amperes maximum. Its DC output is terminated on a 4-pin female XLR connector.

As previously discussed, an Ethernet connection that provides Power-over-Ethernet (PoE) can serve as the Model 236's power source. Alternately, an external 12 volts DC source can be connected. For redundancy, both PoE and the external source can be connected at the same time. If both PoE and an external 12 volts DC source are connected, power will be drawn only from the PoE supply. If the PoE source becomes inoperative the 12 volts DC source will provide the Model 236's power. No interruption in operation will occur during a power source transition.

Analog Microphone Input

Using a configuration selection within the STcontroller software application, the Model 236's microphone audio source can be selected. In most cases a local microphone will be connected and *Analog* will be selected. Alternately, an audio signal can be provided by way of a Dante receiver (input) channel. In this section, details on connecting an analog microphone will be provided.

The Model 236 provides a 3-pin female XLR connector, located on the back panel, to allow an analog microphone to be connected. The microphone can be a standalone handheld ("stick") type, a stand- or arm-mounted type, or can be part of a broadcast-style headset. The Model 236's analog microphone input is directly compatible with balanced dynamic or P48-compatible "phantom" powered microphones. A microphone should be connected such that its associated 3-pin male XLR connector has pin 1 as common, pin 2 as signal high (+), and pin 3 as signal low (–).

A configuration setting in STcontroller allows the Model 236's P48 phantom power source to be enabled or disabled as desired. Another setting allows the gain of the preamplifier circuitry associated with the analog microphone input to be selected. In addition, a configuration setting allows the limiter function to be enabled or disabled as desired. Details on configuration settings will be described in later sections.

While the Model 236 provides an excellent source of P48 phantom power, it's possible that an input on an associated piece of equipment is already providing microphone power. This input could be connected to the Model 236 by way of the analog microphone output connector. This would not create a problem since the circuitry that passes audio from the Model 236's analog microphone input connector to the Model 236's analog microphone output connector will also pass microphone power from output-to-input without interruption. This situation could be relevant in cases where the Model 236's analog microphone output connector is interfaced with a microphone input on an associated ENG camera, audio console, microphone mixer, remote I/O interface, or similar equipment. The microphone inputs on such devices will often provide phantom power capability which may, or may not, be enabled.

Analog Microphone Output

A 3-pin male XLR connector on the Model 236's back panel provides an analog microphone-level output that's directly related to the analog microphone input. Technically this output is identical to the signal that's connected to the analog microphone input but with a solid-state muting circuit in series with the interconnection. A configuration choice in STcontroller allows the operation of the analog microphone output to be selected. The choices are to have the analog microphone output active at all times or for it to also mute whenever the Dante Main transmitter (output) channel mutes.

If the analog microphone output is configured to be active at all times it will provide what is effectively an analog "hot mic" output; a signal that is connected to the analog microphone input will continuously "pass through" to the analog microphone output. No Model 236 action will impact this signal. If the analog microphone output is configured to follow the status of the Dante Main transmitter (output) then the analog microphone input signal will pass through to the analog microphone output connector only when the Dante Main transmitter (output) function is active. Whenever the Dante Main transmitter (output) channel is muted the analog microphone input signal will not pass through to the analog microphone output connector; it is muted in an essentially click-free manner. Refer to Appendix C for a block diagram of the analog microphone input and analog microphone output circuitry.

The analog microphone output can be connected to a balanced (differential) analog microphone-level input on a variety of devices. These include the microphone input connections on a remote I/O interface associated with a networked audio console. An example of such an interface would be the Calrec® Hydra2®. The microphone inputs on these devices typically offer microphone power, high-quality amplification, and conversion to the digital domain. The output signals from an I/O interface's preamplifier channels are typically transported to the main electronics or console surface using a fiber optic interface. In this type of application, the Model 236's Dante Main transmitter (output) channel would not be used, or would only be used as a backup path.

No preamplifier or other active circuitry impacts the path from the Model 236's analog microphone input connector to the Model 236's analog microphone output connector. But the signal does pass through 200-ohm resistors in each "leg" (pin 2 and pin 3) as well as connecting to a solid-state relay contact. The result is that the source impedance of a connected microphone will be 400 ohms greater than the nominal impedance of the analog microphone. This will slightly raise the theoretical noise floor of the microphone signal vis-à-vis a directly connected microphone but shouldn't impact most real-world applications. In addition, when the analog microphone output is in its muted state a 400-ohm load (from the two 200-ohm series resistors) will be connected in parallel with the microphone. This additional load will typically attenuate the microphone output level by 1 to 3 dB. This will then impact the level of the Dante talkback output channels by the same amount.

Headphone Output

The Model 236 provides a 2-channel ("stereo") headphone output interfaced by way of a 3-conductor ¼-inch phone jack. Devices such as stereo headphones or dual-channel ("dual-muff") broadcast-style headsets can be directly connected using a 3-conductor ¼-inch plug. Following the usual convention, the left channel should be terminated on the tip lead of the plug, the right channel on the ring lead, and common on the sleeve lead.

It's also possible to use a single-channel, monaural ("single-muff") headset or a broadcast-type single earbud but in these cases, care must be taken. If a 3-conductor ¼-inch plug is used by such a device's transducer it should be wired to the tip and sleeve leads; the plug's ring lead should be left unconnected. But it's also possible that the monaural device will be terminated on a 2-conductor ("tip and sleeve") plug. When such a plug is inserted into the Model 236's headphone output jack the right headphone output channel will be shorted; the ring lead will be directly connected to the sleeve lead. This can result in stress on the right channel headphone output circuitry as well as extra current draw. To prevent this undesirable condition, ensure that no audio signal is configured such that it's routed to the right headphone output channel. Refer to the Configuration section for details on using STcontroller to select the desired headphone audio routing.

Dante Configuration

For audio to correctly pass to and from the Model 236 requires, at a minimum, that several Dante-related parameters be configured. These configuration settings will be stored in non-volatile memory within the Model 236's Dante network interface circuitry. Configuration will typically be done with the Dante Controller software application which is available for download free of charge at audinate.com. Versions of Dante Controller are available to support Windows and macOS operating systems.

The Model 236 is also compatible with the Dante Domain Manager (DDM) software application. Refer to DDM documentation for details on what Model 236 and related parameters may have to be configured.

Audio Routing

The Model 236's ten Dante transmitter (output) channels must be assigned to the desired Dante receiver (input) channels on associated equipment. This will route the Main, Aux, Talkback, and Phones transmitter (output) channels to the devices that will be "listening" to them. Within Dante Controller a "subscription" is the term used for routing a transmitter flow (a group of up to four output channels) to a receiver flow (a group of up to four input channels).

The Model 236 uses the Broadway integrated circuit to implement its Dante functionality. The number of transmitter flows associated with this integrated circuit is 16 and, as such, the chance of a flow limitation is minimal. These flows can either be unicast, multicast, or a combination of the two. (Note that when operating in the AES67 mode the Dante transmitter (output) channels will function only in multicast; unicast is not supported.)

Using Dante Controller, the desired Dante transmitter (output) sources can be routed to the eleven Dante receiver (input) channels associated with the Model 236. The exact number utilized will depend on the specific application. Eight of the Dante receiver (input) channels are assigned to the Main and Talkback level rotary encoders and can be used for listening to associated mix, mix-minus, intercom, or general audio channels. The ninth Dante input channel can be used as the Model 236's microphone input source.

Unit and Channel Names

The Model 236 has a default Dante device name of **ST-M236-** along with a unique suffix. The suffix identifies the specific Model 236 that is being configured. The suffix's actual alpha and/or numeric characters relate to the MAC address of the unit's Broadway integrated circuit. The eleven Dante receiver (input) channels have defaults names of **Main 1 L**, **Main 1 R**, **Main 2 L**, **Main 2 R**, **Talkback 1, Talkback 2**, **Talkback 3**, **Talkback 4**, **Talkback 5**, **Talkback 6**, and **Mic In**. The ten Dante transmitter (output) channels have default names of **Main**, **Aux**, **Talkback 1**, **Talkback 2**, **Talkback 3**, **Talkback 4**, **Talkback 5**, **Talkback 6**, **Phones L**, and **Phones R**. Using Dante Controller, the default device and channel names can be revised as appropriate for a specific application.

Device Configuration

The Model 236 supports audio sample rates of 48 and 96 kHz with no pull-up/down options available. The digital audio data is in the form of pulse-code modulation (PCM) samples. Encoding choices within Dante Controller are PCM 16, PCM 24, and PCM 32, but in most cases the default selection of PCM 24 would be appropriate. Clocking and Device Latency Parameters can be adjusted if required but the default values in Dante Controller are typically correct.

Network Configuration – Dante Redundancy

The Model 236 allows connection of two Ethernet signals. Two RJ45 jacks are located on the unit's back panel and are labeled Primary-PoE and Secondary. How these ports function can be selected in the Network Config, Dante Redundancy section of Dante Controller. The choices are *Switched* or *Redundant*.

If *Switched* is selected (the factory default) then the Model 236 can establish one connection with an Ethernet network. It doesn't matter which RJ45 jack is utilized, although for PoE powering the jack labeled Primary-PoE must be utilized. The other RJ45 jack, labeled Secondary, can be used to interconnect with another piece of networked equipment.

If the Model 236's network configuration is selected for *Switched* ensure that only one of the RJ45 jacks on the back panel is connected to the LAN associated with the Dante devices. If both of the Model 236's RJ45 connections are routed to ports on the same LAN this will typically "crash" the network! (Although some of the latest, most-advanced Ethernet switches will automatically detect and prevent such a "network bridging" issue from occurring.)

If *Redundant* is selected then Dante's redundant networking capability will be enabled. In this case, separate network connections should be made to the Primary-PoE and Secondary RJ45 jacks. A personal computer that's running STcontroller will typically be associated with the network that is connected to the Primary-PoE RJ45 jack.

Network Configuration – Addresses

When the Model 236 has been configured for the *Switched* network mode one Dante IP address will be associated with the network connection that is made to either the Primary-PoE or the Secondary RJ45 jack. If the network configuration has been selected for *Redundant* then separate IP addresses and related network parameters will be assigned to the Primary-PoE and Secondary Ethernet ports.

By default, the Model 236's Dante IP address and related network parameters will be determined automatically using the DHCP or, if that's not available, link-local network protocols. If desired, Dante Controller allows the IP address and related network parameters to be manually set to a fixed (static) configuration. While this is a more-involved process than simply letting DHCP or link-local "do their thing," if fixed addressing is necessary then this capability is available. But in this case, it's highly recommended that every unit be physically marked, e.g., directly using a permanent marker or "console tape," with its specific static IP address or addresses. If knowledge of a Model 236's IP address or addresses has been misplaced there is no reset button or other method to easily restore the unit to a known (default) IP setting.

Note that if the Model 236's network configuration has been set for *Redundant* then the Primary and Secondary Dante IP addresses and related parameters can be independently configured. This allows both interfaces to be configured automatically, both interfaces to be configured manually, or one interface to be configured automatically and the other to be configured manually.

AES67 Configuration – AES67 Mode

Dante Controller allows a Model 236 to be configured for AES67 operation. This requires the AES67 Mode to be set for *Enabled*. By default, AES67 mode is set for *Disabled*.

Model 236 Clocking Source

While technically the Model 236 can serve as a Leader clock for a Dante network (as can all Dante-enabled devices) in virtually all cases the unit will be configured to receive its timing reference ("sync") from another device. As such, Dante Controller's check box for Preferred Leader associated with the Model 236 would typically not be enabled.

Model 236 Configuration

The Studio Technologies' STcontroller software application is used to configure the way in which the Model 236 functions. No DIP switch settings or other local actions are used to configure the unit. This makes it imperative that STcontroller be available for convenient use on a personal computer that's connected to the related LAN.

Changes made using STcontroller will be immediately reflected in the unit's operation; no Model 236 reboot is required. Each time a change is made the eight LEDs associated with the multi-purpose display on the Model 236's front panel will light orange in a distinctive pattern to indicate that a command from STcontroller has been received.

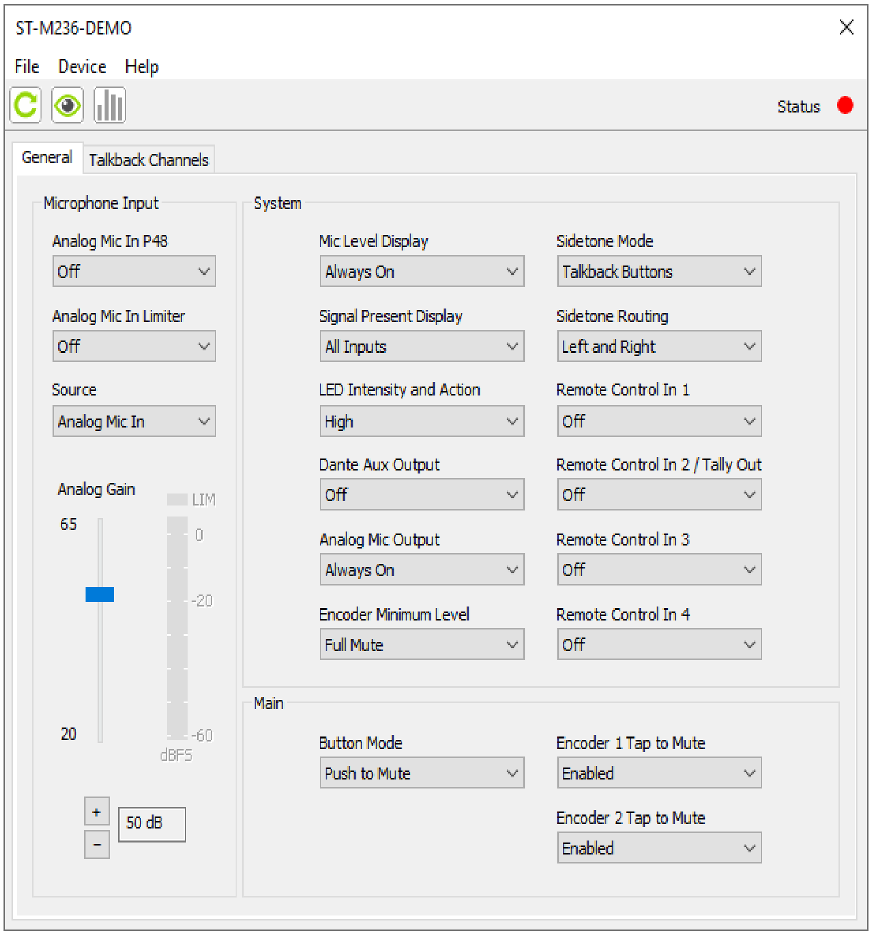
Installing STcontroller

STcontroller is available free of charge on the Studio Technologies' website (studio-tech.com). Versions are available that are compatible with computers running the Windows and macOS operating systems.

If required, download and install STcontroller onto the designated personal computer. This personal computer must be on the same local area network (LAN) and subnet as the Model 236 unit(s) that are to be configured. Immediately after starting STcontroller the application will locate all the Studio Technologies' devices that it can control. The one or more Model 236 units to be configured will appear in the device list. Use the Identify command to allow a specific Model 236 unit to be easily recognized. Double-clicking on a device name will cause the associated configuration menu to appear. Review the current configuration and make any changes that are desired.

General Menu Page

The following configuration selections are available in STcontroller's General menu selections tab:



Microphone Input – Analog Mic In P48: This section has two configurable choices: *Off* and *On*.

STcontroller allows the analog microphone input's P48 phantom power source to be enabled or disabled. The real-time status of the P48 source is displayed both in STcontroller and by way of an orange LED that is located on the Model 236's back panel adjacent to the analog microphone input connector. Select *On* or *Off* to meet the needs of the connected microphone.

No problems will occur if an external source of P12 or P48 phantom power is present on the connection made to the analog microphone output connector. In this case, simply turn off the Model 236's P48 phantom power source. The external source of microphone power will "pass thru" from the analog microphone output connector to the device connected to the analog microphone input connector.

Microphone Input – Analog Mic Input Limiter: This section has two configurable choices: *Off* and *On*.

A level limiting function can be applied to the analog microphone input. This can be useful in helping to prevent overload of the signal associated with the analog input when it is utilized by the main and talkback output channels. When *Off* is selected no limiter action will take place. When *On* is selected the limiter function will be active.

Microphone Input – Source: This section has two configurable choices: *Analog Mic In* and *Dante Mic In*.

Using STcontroller the Model 236's microphone source can be selected. When *Analog Mic In* is selected the audio source will be the device connected to the analog microphone input connector. As expected, a signal connected to the analog microphone input will first pass through the microphone preamplifier circuitry. An audio signal that arrives by way of the Model 236's *Dante Mic In* receiver (input) channel can also be utilized. Any audio signal that is present on this Dante receiver (input) channel can serve as the Model 236's microphone source. This could be useful, for example, where the output of a wireless microphone receiver or a console microphone channel could effectively be utilized as the Model 236's microphone source.

Microphone Input – Analog Gain: When configured for analog, this section can be configured from among the following range: 20 to 65 dB in 1-dB steps.

When the Model 236's microphone input source is selected for Analog Mic In the gain of the microphone preamplifier can be adjusted over a range of 20 to 65 dB in 1-dB steps. There's no problem changing the gain setting while the unit is operating although small audio clicks or pops may occur during gain transitions. Selecting the correct amount of gain for an application might require some experimentation. The goal is to bring the analog microphone's signal up to the Dante reference level which Studio Technologies considers to be –20 dBFS. (This is 20 dB below digital maximum.) Operating at this "reference" signal level will help ensure the delivery of "clean" audio to the destination devices.

There's no "perfect" preamplifier gain setting that this guide can recommend. The two issues that impact the setting are the output sensitivity of the connected microphone and the acoustical output level of the microphone's user. With some microphones or headsets, such as the popular Sennheiser HMD 26 or HMD 27, selecting an initial setting of 40 or 45 dB would be appropriate. Users who speak loudly might need to have their gain reduced to 35 or even 30 dB. "Quiet" users might need a gain setting of 50 or 55 dB.

STcontroller includes a "virtual" level meter which is located to the right of the gain graphic "slider." It also includes an indication of when the limiter function is actively controlling the signal level. The level meter will typically be used as a guide when adjusting the preamplifier gain setting. The level meter function can be enabled or disabled as desired. However, it should be disabled when not required in order to reduce the amount of network data traffic being transported to and from the Model 236. When a voice signal at a normal level is present on the connected microphone or headset the level meter's green area should light. The orange section should light on signal peaks. The red zone and the limiter active indicator should never light.

The multi-purpose display, located on the front panel of the Model 236, can serve as an 8-segment audio level meter which can be used when setting the microphone preamplifier gain. When a voice signal at a normal level is present on the connected microphone or headset the five green LEDs should light. The two orange LEDs should light on signal peaks; the red LED will only light when the signal is near to or actually "clipping."

The signal present/peak LED, located on the unit's back panel below the analog microphone input connector, can also be used as a guide when setting the microphone preamplifier gain. This LED can light orange during signal peaks, but should never light solid red as that would indicate an undesirably high-level condition.

As a "reality check," it's recommended that a level meter associated with a device that's digitally connected to the Model 236's Dante Main transmitter (output) channel be observed. That will be an excellent means of checking the actual signal level within the signal "chain." If necessary, adjust the gain of the Model 236's analog microphone preamplifier to achieve the desired result.

Microphone Input – Dante Trim: When configured for Dante, this section can be configured from among the following range: –20 to 20 dB in 1-dB steps.

A separate input gain setting is provided in STcontroller if the microphone source has been selected to be the Dante Mic In receiver (input) channel. A configuration choice allows adjustment over a range of –20 to 20 dB in 1-dB steps. The information provided in the previous section, Analog Gain, is also appropriate when adjusting the Dante Trim value. The usual goal is to achieve a nominal Dante transmitter (output) level of –20 dBFS when a typical audio signal is being provided on the Dante Mic In receiver (input) channel.

System – Mic Level Display: This section has three configurable choices: *Off*, *Always On*, and *On When Main Active*.

An eight-LED bi-color multi-purpose display is provided on the front panel of the Model 236. It can display the level of the selected microphone input source. (The source is configured, in a separate STcontroller menu choice, to be either the output of the analog microphone preamplifier or the Dante Mic In receiver (input) channel.) This configuration choice is used to enable or disable the microphone level display function as is appropriate for an application. It's possible that the display will only be enabled during setup and testing of an application. After performance confirmation, including setting of the preamplifier gain or Dante trim level, a user may request that the microphone level display function be turned off.

System – Signal Present Display: This section has four configurable choices: *Off*, *Main Inputs*, *Talkback Inputs*, and *All Inputs*.

The top surface of the eight rotary encoders can light as an indication of the level of the audio signal associated with a specific rotary encoder and related input function. RGB (red-green-blue) LEDs are located within each rotary encoder and create the colors that are displayed. A rotary encoder knob will light green when the signal level is within the normal range, yellow when the signal level is greater than normal, and red when the signal level is high enough to risk degrading the audio performance.

The ability of the rotary encoders to provide a signal present indication can be enabled or disabled as desired. The eight rotary encoders are organized into two groups. One group consists of the two rotary encoders associated with the Main inputs. The other group consists of the six rotary encoders associated with the Talkback inputs. A configuration setting in STcontroller allows which group, or groups, will provide a signal present display. Which exact configuration is selected will depend on the requirements of an application and the personal tastes of Model 236 users. Note that even if the Signal Present Display function is configured for *Off* each rotary encoder will still display their operating states using LED illumination: blue for normal and purple for mute.

System – LED Intensity and Action: This section has four configurable choices: *High*, *Low*, *High – Main Off When Inactive*, and *Low – Main Off When Inactive*.

The pushbutton switches, rotary encoders, and multi-purpose display have LED indicators associated with them. The LED Intensity and Action configuration choices allows the overall intensity of these LEDs to be selected. In addition, the choices allow the LED action in the main output pushbutton switch to be selected. Typically, deciding which intensity choice is appropriate will depend on the amount of ambient light present where the Model 236 is located. One of the high settings would usually be used where there is significant ambient light present.

In most announcer's console applications an LED would be lit in the main pushbutton switch to indicate that the function is in its off, inactive, or standby state. In the case of the Model 236 by default the main pushbutton will light red when the function is off and green when the function is on. To assist users who have trouble with color discrimination ("color blindness") the main pushbutton switch can be configured to not light when the function is off (not active). When selecting either of these configuration choices the green LED associated with the main pushbutton switch will light green when the function is active; the pushbutton's red LED will never light.

System – Dante Aux Output: This section has five configurable choices: *Off*, *Hot Mic*, *Tally Tone*, *Main with 18 kHz Tone*, and *Any Talkback Active*.

The Dante Aux transmitter (output) channel is available for special applications and is capable of providing a variety of audio signals. Each configuration choice has been carefully selected and may prove useful in sophisticated applications.

When *Hot Mic* is selected audio associated with the output of the Model 236's microphone preamplifier will always be present on the Dante Aux transmitter (output) channel. (The input source for the preamplifier will always be the analog microphone input.) The *Hot Mic* function can be very useful but has the potential for abuse. By the very nature of a hot mic function, no button press by the user will cause the audio signal on the Dante Aux transmitter (output) channel to become inactive. Users must be aware that audio from the microphone preamplifier will always be present on this Dante transmitter (output) channel. A level anomaly can also be present when the Dante Aux transmitter (output) is configured for *Hot Mic*. Refer to the Technical Notes section for details.

When *Tally Tone* is selected a 20 kHz sine wave signal at a level of –20 dBFS will be present on the Dante Aux transmitter (output) channel whenever the Main output function is active. This is provided as a control signal for use in REMI/At-Home or other specialized applications. A 20 kHz tone being present on the Dante Aux transmitter (output) will serve as the tally signal indicating that audio may be present on the Dante Main transmitter (output) channel.

Main with 18 kHz Tone is an interesting configuration choice. Whenever the Dante Main transmitter (output) channel is active the signal on the Dante Aux transmitter (output) channel will consist of a combination of audio from the selected microphone input source (either the output of the microphone preamplifier or the Dante Mic In receiver (input) channel) and an 18 kHz sine wave tone. In this way, a single Dante transmitter (output) channel will contain both on-air audio and a tally indicator that is specifically provided for REMI/At-Home applications. Other products from Studio Technologies, including the Model 5422A Dante Intercom Audio Engine and the Model 391 Dante Alerting Unit, can directly utilize this signal.

If *Any Talkback Active* is selected then audio associated with the output of the Model 236's microphone preamplifier will be present on the Dante Aux Output transmitter channel whenever one or more of the six talkback functions is active.

System – Analog Mic Output: This section has two configurable choices: *Switched* and *Always On*.

On the Model 236's back panel is a 3-pin male XLR connector that is labeled Mic Out. The way in which this output acts is determined by the Analog Mic Output configuration choice. If *Switched* is selected then audio from the analog microphone input connector will be routed to the analog microphone output connector whenever audio is present on the Dante Main transmitter (output) channel. The analog microphone output connector will be muted whenever the Dante Main transmitter (output) channel is not active. If *Always On* is selected then audio from the analog microphone input connector will always be routed to the analog microphone output connector.

No matter which configuration choice is selected phantom power that is present on the analog microphone output connector will pass through to the analog microphone input connector. This is intentional and will not cause damage to the Model 236. However, for correct operation only one source of phantom power should be enabled at any one time. In practical terms this means that if phantom power is being provided by an external connection to the analog microphone output connector (such as from a microphone input of an audio console or remote I/O interface) then the Model 236's analog microphone input P48 phantom power source should be disabled.

System – Encoder Minimum Level: This section has three configurable choices: *–30 dB*, *–40 dB*, and *Full Mute*.

The Model 236 has eight rotary encoders, two associated with the main section and six associated with the talkback section. This configuration determines the minimum level of signals associated with the rotary encoders as they are turned in their counterclockwise (CCW) direction. The *–30 dB* and *–40 dB* settings ensure that some audio signal will always pass through to the headphone output. This can be useful when it's critical for audio to always be present on the headphone output; a user can't inadvertently fully mute audio input signals when turning a rotary encoder. The *Full Mute* setting is quite different, ensuring that when a rotary encoder is moved counterclockwise (CCW) to the end of its level steps the associated audio input signal(s) will fully mute.

System – Sidetone Mode: This section has four configurable choices: *Off*, *Main Button*, *Talkback Buttons*, and *Main and Talkback Buttons*.

Sidetone is audio from the selected microphone input source that is sent to the analog and Dante headphone outputs. This can be important, allowing a user to "hear" themselves for performance confirmation and comfort. Making a specific selection from among the four available modes will depend on the needs of the application. If a "full mix" is being supplied to the Model 236 then locally provided sidetone won't be needed and the *Off* configuration choice should be selected.

If "mix-minus" audio is being supplied to the Model 236 then selecting a mode which enables sidetone can be an important means of establishing user confidence. The specific sidetone mode that is selected will establish exactly when sidetone audio will be sent to the headphone outputs. If the *Main Button* choice is selected then sidetone audio will be sent to the headphone outputs whenever the audio signal associated with the selected microphone source is present on the Dante Main transmitter (output) channel. If the *Talkback Buttons* mode is selected then the sidetone function will be active whenever one or more of the talkback functions are active. Selecting the *Main and Talkback Buttons* mode will cause the sidetone function to be active whenever the selected microphone audio signal is present on the Dante Main transmitter (output) channel or any of the Dante Talkback transmitter (output) channels.

System – Sidetone Routing: This section has three configurable choices: *Left*, *Right*, and *Left and Right*.

Sidetone is audio from the selected microphone input source. A configuration choice allows the sidetone audio signal to be routed to the headphone outputs' left channel, right channel, or both the left and right channels. The sidetone signal will be routed to the selected channels of both the analog and Dante headphone outputs.

Remote Inputs / Tally Output – Overview

The Model 236 provides four contact closure inputs that can be utilized in installer-implemented remote-control applications. They are physically located inside the Model 236's enclosure on the main printed circuit board and accessible using two 3-pin male "header" connectors. The same electrical connection used for Remote Control In 2 can also be configured to provide a low-voltage, low-current DC output tally signal. A technician will make connections to the headers and will then install one or two connectors in the Model 236's back panel. Refer to the Technical Notes section for details.

System – Remote Control In 1: This section has ten configurable choices: *Off*, *Main Button*, *Talkback 1 Button*, *Talkback 2 Button*, *Talkback 3 Button*, *Talkback 4 Button, Talkback 5 Button,* and *Talkback 6 Button*.

The action performed when Remote Control In 1 is enabled will follow the configuration selected from among six choices. When *Off* is selected no action will be associated with Remote Control In 1. When *Main Button* is selected activating Remote Control In 1 will provide the same function as pressing the main button. (A separate configuration choice allows the action of the main button to be selected.) When any of the talkback button choices are selected then activating Remote Control In 1 will cause the selected talkback function to change state. The precise action of the remote talkback function will be identical to how the associated talkback button is configured. (Separate configuration choices allow the action of the talkback buttons to be selected.)

System – Remote Control In 2 / Tally Out: This section has nine configurable choices: *Off*, *Main Button*, *Talkback 1 Button*, *Talkback 2 Button*, *Talkback 3 Button*, *Talkback 4 Button*, *Talkback 5 Button*, *Talkback 6 Button*, and *Main Out Tally*.

The configuration choices for Remote Control In 2 are the same as those provided for Remote Control In 1 with the addition of a choice of *Main Out Tally*. When this configuration choice is selected then the physical pin on this "header" connector becomes an output. It provides a main output active (tally) signal, changing from 0 to 3.3 volts DC whenever microphone audio is present on the Dante Main transmitter (output) channel.

System – Remote Control In 3: The configuration choices for Remote Control In 3 are the same as those provided for Remote Control In 1.

System – Remote Control In 4: The configuration choices for Remote Control In 4 are the same as those provided for Remote Control In 1.

Main – Button Mode: This section has four configurable choices: *Push to Mute*, *Push to Talk*, *Latching*, and *Hybrid*.

The manner in which the main button functions can be configured from among four choices. If *Push to Mute* is selected the main button's function will normally be active and the green LED associated with the button will be lit. Whenever the main button is pressed the associated function will become inactive and its red LED will light. In a broadcast setting this would typically be considered to be a "cough" function.

If *Push to Talk* is selected the button's function will normally be inactive and its red LED will be lit. Whenever the main button is pressed the associated function will become active and the button's green LED will light.

When the *Latching* mode is selected and the main button is pressed, the main button's function will alternate between its active and inactive state. The button's green LED will be lit whenever the associated function is active, red when it's inactive. Upon power up the main button will be in its inactive state and its red LED will be lit.

The *Hybrid* mode is a combination of the *Push to Talk* and *Latching* modes. It's similar to the way in which pushbutton switches often function in user stations associated with broadcast or production intercom systems. If the main button is pressed and held the associated function will be active. It will stay active until the main button is released. If the main button is momentarily "tapped" the status of the main function will change, either from inactive-to-active or from active-to-inactive. The main button's green LED will be lit whenever the associated function is active, red when it's inactive. Upon power up the main button will be in its inactive state and its red LED will be lit.

Main – Encoder 1 Tap to Mute: This section has two configurable choices: *Disabled* and *Enabled*.

This configuration determines the manner in which the Main 1 rotary encoder functions in relation to the mute action.

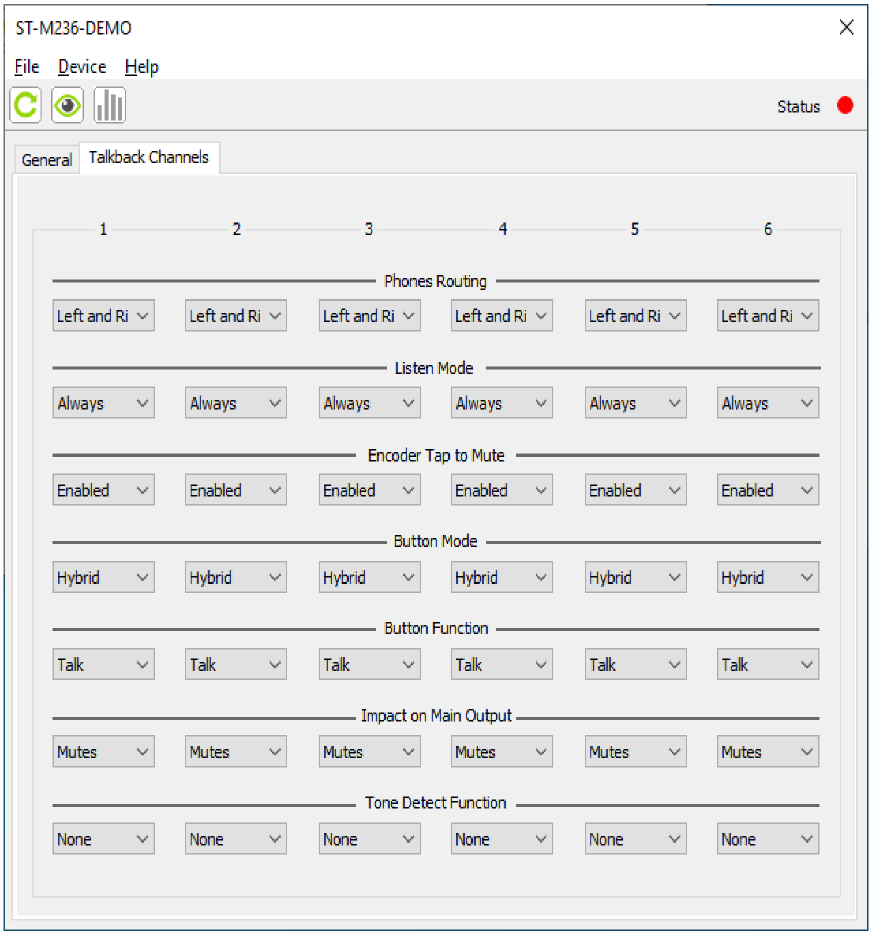
If *Disabled* is selected then momentarily pressing (tapping) the knob associated with the rotary encoder for Main 1 will not cause the associated audio input channels to mute. When pressed, the knob will flash purple five times to indicate that the encoder press was recognized but no action is taking place.

If *Enabled* is selected then momentarily pressing (tapping) the knob associated with the rotary encoder for Main 1 will cause the associated audio input channels to mute. The knob will light purple when mute is active.

Main – Encoder 2 Tap to Mute: The configuration choices for Main – Encoder 2 Tap to Mute are the same as those provided for Main – Encoder 1 Tap to Mute.

Talkback Channels Menu Page

The following configuration selections are available in STcontroller's Talkback Channels menu selections tab. The configuration selections apply individually to Talkback Channels 1-6.



Phones Routing: This section has three configurable choices: *Left*, *Right*, and *Left and Right*.

A configuration choice allows the signal that arrives on the Model 236's Dante Talkback receiver (input) channel to be routed to the left channel, right channel, or both the left and right channels of the analog and Dante headphone outputs. A rotary encoder is associated with the talkback function and is used to control the on/off status and level of the Dante Talkback receiver (input) source.

Listen Mode: This section has four configurable choices: *Always*, *Main Out Active*, *Main Out Inactive*, and *Talkback 1 Active*.

This configuration choice selects under which operating mode(s) the audio source connected to the Dante Talkback receiver (input) channel will be routed to the analog and Dante headphone outputs. This is a somewhat esoteric configuration but can prove useful in specialized applications. If *Always* is selected an audio source connected to the Dante Talkback receiver (input) channel can always be monitored using the headphone outputs. No main or talkback button press will impact the ability to monitor the talkback source.

If *Main Out Active* is selected then monitoring of the source connected to the Dante Talkback receiver (input) channel can only take place when the main output function is active. If *Main Out Inactive* is selected then the opposite action will take place. Only when the main output function is inactive can the source connected to the Dante Talkback receiver (input) channel be monitored. When *Talkback 1 Active* is selected then the source connected to the Dante Talkback 1 receiver (input) channel can only be monitored when the Talkback 1 function is active. Note that this function will only apply to the specific talkback function that is being configured, i.e., Talkback 1, Talkback 2, etc.

Encoder Tap to Mute: This section has two configurable choices: *Disabled* and *Enabled*.

This configuration determines the manner in which the Model 236's six talkback rotary encoders function in relation to the mute action.

If *Disabled* is selected then momentarily pressing (tapping) the knob associated with a rotary encoder will not cause the associated audio input channel to mute. When pressed, the knob will flash purple five times to indicate that the encoder press was recognized but no action is taking place.

If *Enabled* is selected then momentarily pressing (tapping) the knob associated with a rotary encoder will cause the associated audio input channel to mute. The knob will light purple when mute is active.

Button Mode: This section has three configurable choices: *Push to Talk*, *Latching*, and *Hybrid*.

The manner in which each talkback button functions can be configured from among three choices. If *Push to Talk* is selected the button's function will normally be inactive and the LED associated with the button will not be lit. Whenever the talkback button is pressed its associated function will become active and the button's green LED will light.

When the *Latching* mode is selected and the associated talkback button is pressed, the button's function will alternate between its active and inactive state. The button's green LED will be lit whenever the function is active. Upon Model 236 power up the function will be in its inactive state and its LED will not be lit.

The *Hybrid* mode is a combination of the Push to Talk and Latching modes. It's similar to the way in which pushbutton switches often function in user stations associated with broadcast or production intercom systems. If the talkback button is pressed and held the associated function will be active. It will stay active until the button is released. If the talkback button is momentarily tapped the status of the function will change, either from inactive-to-active or from active-to-inactive. The talkback button's green LED will be lit whenever the function is active. Upon Model 236 power up the talkback button will be in its inactive state and its LED will not be lit.

Button Function: This section has five configurable choices: *Talk*, *Talk with 18 kHz Tone*, *Call*, *20 kHz Tone*, and *Disabled*.

The action caused by a talkback button being active can be configured from among five choices including being disabled. Each choice is unique and is worth contemplating how it can serve a specific application. If configured for *Talk*, whenever the associated talkback function is active then microphone audio will be sent out the associated Dante Talkback transmitter (output) channel. This is the typical talkback action where audio from an announcer or commentator is sent to broadcast support personnel including directors, producers, "spotters," or statisticians. The Dante Talkback transmitter (output) channel might be routed directly to an amplified loudspeaker, audio monitor panel, or input on a matrix intercom system.

The *Talk with 18 kHz Tone* function is similar to the Talk mode except that an 18 kHz sine wave tone at a level of –20 dBFS is added (summed) with the microphone audio. It is intended for use in REMI/At-Home applications where a "pilot tone" needs to be sent along with microphone audio. This is specifically provided for use with the tone-activated ("TOX") talent cue (IFB) function supported in the Studio Technologies' Model 5422A Dante Intercom Audio Engine product. The 18 kHz tone will serve as the "trigger," enabling microphone audio coming from a Model 236 to pass through a Model 5422A and on to a talent cue listener.

The *Call* function is provided to emulate the call button function provided on certain intercom equipment. If a talkback button is configured for *Call*, whenever it is enabled a 20 kHz sine wave tone at a level of –20 dBFS will be sent out an active Model 236 talkback channel that is configured for either the *Talk or the Talk with 18 kHz Tone* modes. The 20 kHz sine wave will be combined with microphone audio and, if configured, the 18 kHz tone. This function can allow a Model 236 to serve as both an announcer's console and an intercom user station.

When *20 kHz Tone* is selected and its associated talkback button is pressed, only 20 kHz sine wave audio at a level of –20 dBFS will be sent out the associated Dante transmitter (output) channel. This function can be useful for triggering other functions associated with connected devices such as the Model 5422A Dante Intercom Audio Engine and the Model 391 Dante Alerting Unit.

The *Disabled* selection allows a talkback button to not impact the performance of the other Model 236 resources. This can simplify the user experience when an application isn't utilizing all of the talkback buttons. When a talkback function has be disabled tapping the associated button will cause its LED to flash red four times to indicate that it's doing nothing. If the button is pressed and held its red LED will flash continuously.

Impact on Main Output: This section has two configurable choices: *None* and *Mutes*.

A configuration choice selects how enabling a talkback function will interact with the main output function. For on-air use the choice of *Mutes* would typically be selected. In this way, a user pressing a talkback button will cause microphone audio present on the Dante Main transmitter (output) channel to automatically mute. This would be the normal operating condition in an announcer's console application; the microphone audio content that's being sent to a talkback channel isn't intended to be heard by broadcast listeners!

For special Model 236 applications the *None* configuration choice may be desired. When selected for this mode microphone audio remains present on the Dante Main transmitter (output) channel, whether or not the talkback function becomes active. This might be appropriate if a Model 236 is being used as an intercom station or in a stage manager console application.

Tone Detect Function: This section has three configurable choices: *None*, *Call Light*, and *IFB (Dims Main Phones Audio)*.

The Model 236 has the ability to detect a high-frequency tone that's present on a Dante Talkback receiver (input) channel. This detection capability can be used for two unique actions – one is to create a call light function and the other to create a talent cue (IFB) function. If the tone detect function is not needed the configuration choice of *None* would be appropriate.

A valid tone is defined as a signal that's continuously present in the Dante Talkback receiver (input) channel with a frequency of approximately 15 kHz or greater. (There is no means to allow a momentary tone to create a latching function.) The talkback audio source can consist of just a high-frequency tone, but more typically will be a combination of a high-frequency tone and a voice or broadcast program audio signal.

The *Call Light* function is provided such that an intercom call tone signal, when present on a Dante Talkback receiver (input) channel, will result in the associated talkback rotary encoder knob flashing yellow. It will also cause the multi-purpose display to flash orange. By design the multi-purpose display will flash a maximum of five times while the rotary encoder knob will continue to flash as long as a call tone is detected. The call light function allows the Model 236 to be integrated into an intercom application that uses high-frequency call signals. For example, intercom stations, such as the Studio Technologies' Model 348, can generate call signals which can be detected by the Model 236's talkback input channels. In addition, voice or program audio that's present on this Dante Talkback receiver (input) channel can be routed to either or both the headphone output channels. Before the signal is routed to the headphone outputs it will pass through a low-pass filter (LPF) which removes the high-frequency call tone content.

Selecting *IFB (Dims Main Phones Audio)* allows a broadcast type talent cue (IFB) function to be created directly inside the Model 236. This can be very useful in REMI/At-Home applications where there may not be a central location that supports generation of talent cues. By using the Model 236's tone detect function in the *IFB (Dims Main Phones Audio)* mode, simply providing interrupt audio along with a high-frequency "pilot" or activation tone will allow a broadcast-standard IFB signal to be created.

The actual IFB function's action is very simple. If a high-frequency tone is detected on the Dante Talkback receiver (input) channel the Model 236 will respond by attenuating (dimming or reducing in level) the audio associated with the Dante Main receiver (input) channels to which the talkback audio has been assigned to. For example, if the talkback audio source has been assigned to the left headphone channel then audio coming from a Dante Main receiver (input) channel that is routed to the left phones channel will be attenuated ("dimmed") whenever the IFB function is active. The audio level associated with the Dante Main receiver (input) signals will return to normal when the high-frequency tone is no longer present. The attenuation (dim) level is fixed at 15 dB which should work well for virtually all IFB applications. As previously mentioned, before the talkback signal is routed to the headphone output it will pass through a low-pass filter (LPF) which removes the high-frequency call tone content. Additional details on this function are provided in the Technical Notes section.

Operation

At this point all connections and configuration steps should have been completed and Model 236 operation should be ready to commence. In most cases an Ethernet connection with Power-over-Ethernet (PoE) capability will have been made to the Primary-PoE Ethernet connector. Alternately, a source of 12 volts DC may have been connected to provide power for the Model 236. A second Ethernet connection may be present if Redundant Dante is being utilized by the application. A microphone and a pair of headphones or earbuds should have been connected. Alternately, a broadcast-style headset may have been connected. If desired, a connection to the analog microphone output will have been made.

Using the Dante Controller software application the Model 236's network interface and audio channel routing configuration should have been made. In this way, the operational characteristics of the unit's two Ethernet interfaces will have been selected. In addition, some or all of the ten Dante transmitter (output) channels and eleven Dante receiver (input) channels will have been routed, by way of Dante "subscriptions," to the receiver and transmitter channels on associated Dante-enabled equipment.

Seven of the Model 236's Dante transmitter (output) channels are associated with the Main and Talkback functions. Another Dante transmitter (output) channel is associated with the Model 236's Aux output. Two additional Dante transmitter (output) channels are digital duplicates of the 2-channel (stereo) analog headphone output. Ten of the Dante receiver (input) channels are associated with the main and talkback headphone monitor sections. These inputs have associated rotary encoders to adjust the signal levels as they are sent to the headphone output "mix." A eleventh Dante receiver (input) channel can be used to provide an alternate microphone input signal.

Using the Studio Technologies' STcontroller software application, the Model 236's operating configuration should have been selected to meet the needs of the application. An extensive range of configurable parameters is available, allowing the way in which a specific Model 236 operates to be quite unique.

Initial Operation

The Model 236 will start to function as soon as a Power-over-Ethernet (PoE) or a 12 volts DC power source is connected. However, it may take up to 30 or 40 seconds for full operation to commence. Upon initial power up the various status LEDs, located on the back panel, along with the rotary encoders, multi-purpose display, and pushbutton switch LEDs on the front panel, will light as the internal processors, Ethernet network, and Dante interface begin to function. The various LEDs will then become operational, displaying the status of their associated functions.

Power Status LEDs

An LED labeled PoE is located below the Primary-PoE Ethernet connector on the Model 236's back panel. It will light green if an Ethernet signal that is providing PoE power is connected. An LED labeled DC is located below the 4-pin XLR connector that is provided for connection to a DC power source. It will light green if a source of 10 to 18 volts DC is connected. If both the PoE and DC LEDs are lit then the PoE Ethernet connection will be providing Model 236 operating power. The DC power supply will be ready as a "hot standby" should the PoE source fail.

Ethernet Status LEDs

Four Ethernet status LEDs are located below the two etherCON RJ45 connectors on the Model 236's back panel. The LINK ACT LED will light green whenever an active connection to an Ethernet network has been established. It will flash in response to Ethernet data packet activity on that interface. The GbE LED will light orange when an active Gigabit Ethernet connection is present on that interface, something that is required for optimal operation.

Dante Status LEDs

The SYS and SYNC LEDs, also located on the back panel, display the operating status of the Dante interface and its associated network activity. (The Model 236 has one Dante interface although single and redundant local area networks (Redundant Dante) can be connected.) After lighting red and green as part of the power-up sequence the SYS LED will light red to indicate that the Dante interface is not ready. After a short interval it should light green to indicate that it is ready to pass data with other Dante devices. The SYNC LED will also be part of the power-up sequence and will light red when the Model 236 is not synchronized with a Dante network. It will light solid green when the Model 236 is synchronized with a Dante network and an external clock source (timing reference) is being received. It will slowly flash green when this specific Model 236 is part of a Dante network and is serving as the Leader clock. It's possible that up to 30 or 40 seconds may be required for the SYNC LED to reach its final state.

P48 Power Status LED

An LED indicator is located on the Model 236's back panel adjacent to the analog microphone input connector. It is labeled P48 and will light red whenever the analog microphone input's P48 phantom power source is active and capable of providing DC power to the connected microphone.

Microphone Input Signal Active LED and Mic Preamp Gain

An LED indicator is located on the back panel adjacent to the analog microphone input connector. It is labeled SIG PEAK and displays the output level of the Model 236's microphone preamplifier. The LED will light green when a signal at a normal level is present on the output of the microphone preamplifier. The LED will light orange when a signal is above the normal level but audio quality is not being compromised. This is a "warning" indication that should be noted as a potential issue. The LED lighting orange in response to an occasional signal peak is not a problem. If the LED lights red then the signal from the microphone preamplifier is close to, or has reached, its maximum level. A signal reaching "clipping" must be avoided to ensure that microphone audio quality has not been compromised. The Model 236's limiter function, when enabled, will minimize the chance that clipping will occur. The limiter function being active on more than an infrequent occasion will normally indicate that the gain of the microphone preamplifier should be reduced. Refer to the Microphone Input – Analog Gain section for details on how to best to configure the gain of the microphone preamplifier.

LED Intensity and Action

On the Model 236's front panel there are LED indicators associated with the pushbutton switches, rotary encoders, and multi-purpose display. Using the STcontroller software application the intensity of these LEDs, as a group, can be configured from among four choices, two that specify high and two low. If during operation the LED intensity isn't perceived to be optimal then a change in the configuration choice might be warranted. In most cases, the default intensity, high, will be applicable. However, one of the low LED intensity configuration choices may be superior in applications where the ambient light in the Model 236's location is also low.

The green LED associated with the main pushbutton switch will always light when the function is active. Two configuration choices cause the red LED associated with the main pushbutton switch to be lit when the function is not active. For applications where the user may have trouble identifying the difference between the colors green and red two configuration choices allow the red LED to be disabled. When one of these configuration choices is selected the main pushbutton switch will not light when the function is not active. It will light green when the function is active.

How to Identify a Specific Model 236

Both the Dante Controller and STcontroller software applications provide a function to allow a specific Model 236 unit to be identified. Both applications include an "eyeball" icon that when clicked will activate the Model 236's Identify function. On that unit the eight LEDs associated with the front-panel multi-purpose display will flash red in a unique sequence four times. In addition, the LINK ACT and GbE status LEDs, located directly below the two RJ45 connectors on the back panel, will slowly flash. After a few seconds the LED identification patterns will cease and normal Model 236 multi-purpose display and network status LED operation will resume.

General Operating Notes

As has been previously covered, many Model 236 operating features can be configured using the STcontroller software application. For details refer to the Model 236 Configuration section. The following paragraphs will highlight some of the Model 236's capabilities and introduce them to new users.

Microphone Source Selection

The Model 236's microphone input source can be selected, using STcontroller, to meet the specific needs of an application. In most cases the microphone source will be configured to be the output of the preamplifier circuitry associated with the analog microphone input. This is the usual arrangement where a microphone is connected to the 3-pin female XLR connector located on the Model 236's back panel. Alternately, the microphone source can be selected to be the Dante Mic In receiver (input) channel. This is provided so that devices with Dante transmitter (output) channels, such as a wireless microphone receiver, can be directly utilized as the Model 236's microphone source. Each microphone source, analog and Dante, has an independent gain configuration parameter which is accessible by way of STcontroller. The applicable parameter should be adjusted if a level issue prevents excellent Model 236 audio performance from being achieved.

Multi-Purpose Display

A display consisting of eight bi-color LEDs is provided on the Model 236's front panel. The display, which allows each LED to light green, orange, or red, is used for several functions. It will flash red in a unique pattern to indicate that the Dante Identify command has been received. It will flash orange in a different pattern when a configuration change has been received from the STcontroller software application. The multi-purpose display's LEDs will also light orange to indicate the approximate setting of the rotary encoders. A call tone detected on a talkback input channel will cause all the display's LEDs to momentarily flash orange. Details about these actions are covered in later sections.

The multi-purpose display can also be used to indicate the level of the selected microphone source. Starting from the left, the first five LEDs (1-5) will light green to indicate a microphone signal level that's within the normal range. The next two LEDs (6 and 7) will light orange when a microphone signal level is greater than normal. This can be considered to be a "warning" range. The LED on the far right (8) will light red when the microphone source's signal level is nearing, or has reached, its maximum. A microphone signal at an acceptable level may occasionally light the orange LEDs. But the red LED should rarely, if ever, light. A configuration choice in STcontroller allows the multi-purpose display's level meter mode to be enabled or disabled as desired.

Dante Aux Output

Although a user probably won't be aware of it, the Model 236 provides a Dante transmitter (output) channel designated as Aux. How this output functions will depend on how it has been configured in STcontroller. By default, the Dante Aux output is disabled. Four other choices allow some interesting functions to be created. When configured for *Hot Mic* the signal associated with the output of the microphone preamplifier will always be present on the Dante Aux output. (The preamplifier supports the analog microphone input.) The user should be aware that when this configuration is selected there is no means to mute the Dante Aux transmitter (output) channel. They must assume that other personnel will be continually monitoring audio associated with the analog microphone source. When configured for the *Tally Tone* mode the Dante Aux channel will have an 18 kHz sine wave tone present whenever the main output function is active. A third configuration mode, *Main with 18 kHz Tone* will have both the microphone audio source and an 18 kHz sine wave tone present on the Dante Aux output whenever the main output function is active. These latter two modes are provided for use in supporting

REMI/At-Home applications. When *Any Talkback Active* is selected the microphone audio source will be present whenever one or more of the six talkback functions is active.

Main and Talkback Buttons

The action of the main and talkback pushbutton switches can be independently configured using STcontroller. These are called the button modes and there are four choices for the main button and three choices for the talkback buttons.

Main Button Modes

The main button can be configured from among four choices: *Push to Mute*, *Push to Talk*, *Latching*, and *Hybrid*.

Push to Mute: When the main button is configured for this mode the function is enabled when the button is not pressed and disabled when the button is pressed. This is a typical "cough" function that's frequently utilized in on-air broadcast applications. The main button will light green when the main function is active and will either light red or not light when it is not active.

Push to Talk: In this mode the function becomes active whenever the main button is pressed and held.

Latching: In this mode the function will change state whenever the main button is momentarily pressed (tapped), either off-to-on or on-to-off.

Hybrid: This mode is a combination of the two previous modes. If the main button is pressed and held the function will become active. It will become inactive when released. If the button is tapped then the function will change states.

Talkback Button Modes

The six talkback buttons can be independently configured to function in the *Push to Talk*, *Latching*, or *Hybrid* modes. These take on the same actions as were discussed in the previous section for the main button.

Talkback Button Functions

The six talkback channel functions can be independently configured. The channels can function from among four choices (*Talk*, *Talk with 18 kHz Tone*, *Call*, and *20 kHz Tone*) as well as being able to be disabled. A careful study of each choice will highlight the large variety of applications a Model 236 can support. While the product's title includes the text "Announcer's Console," that doesn't cover all the applications in which a Model 236 might prove useful.

Talk: When a talkback button function has been configured for this mode and its button is active then audio from the selected microphone source will be connected to the associated Dante transmitter (output) channel. The button's LED will light green whenever the function is active. When the function is not active its LED will not be lit and no audio will be present on the associated Dante transmitter (output) channel.

Talk with 18 kHz Tone: This function is similar to the Talk function except that an 18 kHz sine wave tone is added (summed) with audio from the selected microphone source. This function is specifically provided for use in REMI/At-Home applications when an on-air or tally indication in the audio path is desired.

Call: When a talkback button's function has been configured for this mode it will emulate the call button function provided on intercom equipment. Whenever a talkback button that is configured for *Call* is enabled a 20 kHz sine wave tone will be added (summed) with the audio present on an active Model 236 talkback channel that is configured to either the Talk or the Talk with 18 kHz Tone modes.

20 kHz Tone: When a talkback button function has been configured for this mode the action is very simple. When the function is active a 20 kHz sine wave tone is connected to the associated Dante transmitter (output) channel. No other audio signal will be present on that channel.

Disabled: When a talkback button function has been configured for this mode no audio action will take place. When a talkback function has be disabled tapping the associated button will cause its LED to flash red four times to indicate that it's doing nothing. If the button is pressed and held its red LED will flash continuously. No voice or tone audio will be present on the associated Dante transmitter (output) channel.

Audio Monitoring

The Model 236's audio monitor section allows a user to create their desired audio mix from eleven sources. The sources consist of ten Dante receiver (input) channels, four main and six talkback, as well as the selected microphone input source (sidetone). Mixing (summing) of the selected sources is done in the digital domain to create the 2-channel signal "bus." This 2-channel (stereo) mix is available as both an analog headphone output, provided on the Model 236's back panel by way of a 3-conductor ¼-inch jack, and in the form of two Dante transmitter (output) channels.

Headphones, headsets, or in-ear monitors can be directly connected to the analog headphone output. The Dante Phones L and Phones R transmitter (output) channels can be routed to devices that can utilize the networked digital audio sources. Examples include wireless headphone or talent cue (IFB) systems, matrix intercom frames, or audio consoles.

Using eight rotary encoders and one rotary potentiometer, a user can directly create the desired combination of audio sources (the "mix") they want. The rotary encoders allow adjustment of the main and talkback input levels. A rotary potentiometer, located on the left side of the Model 236's front panel, is used to adjust the sidetone level. Another rotary potentiometer, located on the right side of the Model 236's front panel and labeled Master, is used to adjust the overall level of the headphone output. Setting this control to its 2 o'clock position is the recommended starting point for effective listening. Both the Sidetone and Master potentiometers are push-in/push-out types which allow their respective levels to be easily set and then "pushed-in" to provide protection from being accidentally changed.

Headphone Routing and Level Control

Main Input 1 and Main Input 2 each have two Dante receiver (input) channels associated with them. How the Dante source signals are subscribed (routed) to the left and right channels of these inputs (using Dante Controller) will dictate what is routed to the headphone output channels. Stereo or dual-channel sources can be routed to the left and right channels of a main input. Or a monaural signal can be subscribed (routed) to either or both of a main input's left and right channels. One rotary encoder is associated with Main Input 1 and will control both the left and right channels as they are sent to the analog and Dante headphone outputs. A second rotary encoder impacts the left and right channels associated with Main Input 2.

STcontroller provides independent configuration choices for each of the Dante talkback receiver (input) channels. This allows these signals to be routed to the left, right, or both the left and right headphone channels. Turning one of the talkback rotary encoders will impact its associated talkback audio source as it is sent to the left, right, or both the left and right headphone output channels.

Sidetone

The sidetone function is provided for assisting users in hearing audio from the Model 236's selected microphone input source. Exactly when the sidetone function will operate will depend on a STcontroller configuration setting. Three choices are available, along with the ability to disable the sidetone function. When configured for *Main Button* the sidetone function will be active only when the main button function is active. When set to the *Talkback Buttons* mode sidetone audio will be sent to the headphone outputs whenever one or more of the talkback functions are active. The third choice, *Main and Talkback Buttons*, will cause the sidetone function to be active whenever any function is active. A separate STcontroller configuration choice allows sidetone audio to be routed to the left, right, or both the left and right headphone output channels.

Listen Mode

A unique STcontroller configuration mode allows each talkback input function to be individually configured for when its audio will be routed to the headphone outputs. There are four choices available, but in most cases the default, *Always*, will be appropriate. This mode will allow an audio source entering the Model 236 by way of a Dante Talkback receiver (input) channel to always be able to be monitored.

For special applications the other three choices will restrict when the audio source can be monitored. When configured for the *Main Out Active* mode the associated talkback audio input source can only be monitored when the main output function is active. The *Main Out Inactive* mode provides the opposite action; only when the main output function is not active can the associated talkback audio source be monitored. The third choice allows the six talkback functions to have a unique mode that relates to its specific channel. For example, the *Talkback 1 Active* mode will allow the audio source associated with the talkback 1 audio source to be monitored only when the talkback 1 function is active.

Rotary Encoders

Eight rotary encoders are provided on the front panel of the Model 236. Two of the rotary encoders are located above the Main pushbutton switch. They are associated with the four Dante Main receiver (input) channels. Each will control the level of one or two audio sources, allowing monaural or stereo operation. A rotary encoder is associated with each of the six Dante Talkback receiver (input) channels. Each rotary encoder has 24 steps ("clicks" or "detents") for each 360 degrees of rotation. Turning a rotary encoder knob clockwise by 1 and 1/3 rotations (32 steps or 480 degrees) is required to change from audio minimum to audio maximum. (As expected, the same action occurs in reverse: when turning a rotary encoder counterclockwise by 32 steps the audio level will change from maximum to minimum.)

The top surface of each rotary encoder knob is translucent. RGB (red-green-blue) LEDs within each rotary encoder can display a wide range of colors. A rotary encoder knob that is dimly lit with a dark blue color indicates that its associated function is active. This provides a physical identification of each rotary encoder's location when a Model 236 is being used in an environment where the ambient light is very low or nonexistent. It's important to note that there could be a signal present on the associated Dante receiver (input) channel but unless the Signal Present Display configuration has been enabled for that input channel its knob will remain dark blue.

The RGB (red-green-blue) LEDs within each rotary encoder can shine through the end of its associated knob to provide a visual indication of the level of the signal or signals associated with the rotary encoder's function. As previously mentioned, for the signal level to be displayed on the rotary encoder knobs requires that the Signal Present Display configuration in STcontroller be selected as desired. The default condition is for the signal present display to be active for both the Dante Main receiver (input) channels and the Dante Talkback receiver (input) channels. This setting would be appropriate for most applications. The other configuration choices allow just the rotary encoders associated with the main inputs or just the rotary encoders associated with the talkback inputs to have their signal present display function active.

If enabled for signal present display a rotary encoder knob will light in response to the signal level on its associated Dante receiver (input) channel or channels. This function is pre-encoder (also known as "pre-fader") which means that the encoder's LEDs will respond to an input signal level prior to any gain or loss that may be selected by adjusting the associated rotary encoder. A rotary encoder knob will light green to indicate that a signal with a level of –40 to –16 dBFS is present on its associated Dante receiver (input) channel or channels. The knob will light yellow when the signal has a level of –15 to –5 dBFS. The knob will light red when the signal has a level that meets or exceeds –5 dBFS.

Main Section Rotary Encoders

A rotary encoder is associated with the left and right input channels of Main Input 1. A second rotary encoder is associated with the left and right input channels of Main Input 2. A number of configuration choices in STcontroller determine how these rotary encoders will function. If enabled to display the level of the associated input signals, the rotary encoder knobs will light in response to the higher of the signal levels that are present on their respective left and right input channels; independent display of the signal levels present on the left and right inputs is not possible.

Two configuration choice determines if pressing an encoder's knob will mute or unmute the associated input signals or if the knob press will be ignored. An STcontroller choice also determines if the level of the main receiver channels will either be fully muted or attenuated (reduced in level) when an encoder is rotated the maximum number of counterclockwise (CCW) steps.

If either or both of the two main rotary encoders have been configured to allow muting, momentarily pressing (tapping) a rotary encoder knob will cause the associated audio channels to mute. An encoder's knob will light purple to indicate that its associated function has been muted. To unmute the signals simply requires that the associated rotary encoder knob again be pressed and released (tapped). If the main input rotary encoders have been configured to not allow muting, momentarily pressing (tapping) a rotary encoder knob will not impact the audio signals. The encoder's knob will momentarily flash purple to indicate that the button press was recognized but no action was taken.

The minimum encoder level can also be configured. This directly impacts the functionality of the two rotary encoders that support the main 1 and main 2 listen functions. If the configuration is selected for *Full Mute* then turning either of the rotary encoders in their counterclockwise (CCW) direction will eventually cause the associated input signals to fully mute. If the configuration is selected for one of the attenuation values (choices are *–30 dB* or *–40 dB*) then turning a rotary encoder CCW will eventually attenuate the associated audio input channels by the selected value. But the inputs won't fully mute, ensuring that audio signals can always be sent to the headphone output channels. This is provided for broadcast applications where users, typically on-air talent, must be able to hear program, mix-minus, or other support audio at all times. This will help to ensure that users can't accidentally mute their headphone output channels and cut off their audio "link" to production personnel.

Each encoder provides 24 steps per one full rotation. (24 steps per 360 degrees of rotation.) As created in software, 32 encoder steps (1 and 1/3 rotations of a rotary encoder knob) are required to move between the minimum and maximum levels. For user assistance, the eight LEDs associated with the multi-purpose display on the front of the Model 236 can indicate the approximate position of each rotary encoder. Each LED on the multi-purpose display will light or flash orange in response to a rotary encoder being within a group of four steps. Encoder step 1 is considered to be full mute and causes the display's LED 1, located on the far left, to flash orange. LED 1 will light orange when a rotary encoder has been set to steps 2-4. The display's LEDs 1 and 2 will light when a rotary encoder has been set to steps 5-8, etc. Step 32 is the maximum level and causes the display's LEDs 1 through 7 to light orange and LED 8 to flash orange.

There are two ways to initiate display of the approximate position of a rotary encoder. One is to press and hold a rotary encoder knob. While the knob is pressed LEDs in the multi-purpose display will light orange. If the main rotary encoders are configured to allow full mute then LED 1 (on the display's far left) can flash orange. If the main rotary encoders are configured for *–30 dB* or *–40 dB* minimum then LED 1 will never flash. (This is because the rotary encoders associated with the main inputs are not allowed to be set to full mute.) If a rotary encoder is moved fully clockwise (CW) to reach its full (maximum) level then the multi-purpose display's LEDs 1 through 7 will light orange and LED 8 (on the far right) will flash orange. The indication of the rotary encoder's level will continue until its knob is no longer pressed.

The second method of displaying a rotary encoder level setting is to rotate the encoder knob at least one step in either the clockwise or counterclockwise direction.

The multi-purpose display will light orange to indicate the approximate level, maintaining that indication for approximately one second after the last rotary encoder change has been made.

Talkback Receiver Rotary Encoders

Each talkback channel has a Dante receiver (input) channel associated with it along with a related rotary encoder. Depending on the unit's configuration the encoder's knob can light in response to the level present on its associated input channel. Another configuration choice allows pressing the rotary encoder's knob to mute or unmute the associated audio input signal. If a talkback channel has been configured to allow mute operation, to enable or disable the mute function on a talkback input simply requires that its associated rotary encoder knob be pressed and released (tapped). A knob that is lit purple indicates that its associated input has been muted. Whenever mute is active any changes to a rotary encoder's position will not be recognized. If the encoder has been configured to not allow mute to be enabled, pressing the knob will result in a set of purple flashes that indicate that the button press has been recognized but no change is being made to the audio performance.

The minimum encoder level can also be configured. This directly impacts the functionality of the six rotary encoders that support the talkback listen functions. If the configuration is selected for *Full Mute* then turning one of the rotary encoders in their counterclockwise (CCW) direction will eventually cause its associated input signal to fully mute. If the configuration is selected for one of the attenuation values (choices are *–30 dB* or *–40 dB*) then turning a rotary encoder CCW will eventually attenuate its associated audio input channel by the selected value. But an input won't fully mute, ensuring that audio signals can always be sent to the headphone output channels. This is provided for broadcast applications where users, typically on-air talent, must be able to hear support audio at all times. This will help to ensure that users can't accidentally mute their headphone output channels and cut off their audio "link" to production personnel.

As previously mentioned, 32 encoder steps (1 and 1/3 rotation of a rotary encoder knob) are required to move between the function's minimum and maximum levels. For user assistance, the eight LEDs associated with the multi-purpose display on the front of the Model 236 can indicate the approximate position of the rotary encoder. To display the rotary encoder's level setting simply requires that the encoder knob be rotated at least one step in either the clockwise or counterclockwise direction. The multi-purpose display will light orange to indicate the approximate level, maintaining that indication for approximately one second after the last rotary encoder change has been made.

Each LED on the multi-purpose display will light or flash orange in response to a rotary encoder being within a group of four steps. Encoder step 1 is considered to be the full-mute state and will cause the display's LED 1, located on the far left, to flash orange. LED 1 will light orange when a rotary encoder has been set to steps 2-4. The display's LEDs 1 and 2 will light when a rotary encoder has been set to steps 5-8, etc. Step 32 is the maximum level and causes the display's LEDs 1 through 7 to light orange and LED 8 to flash orange.

An alternate method is available to cause the level of a rotary encoder to be displayed. This involves pressing and holding the rotary encoder's knob. While the knob is pressed the LEDs in the multi-purpose display will light orange as described previously. The indication of the rotary encoder's level will continue until its knob is no longer pressed.

Talkback Impact on Main Output

An STcontroller configuration setting allows the relationship between the six talkback functions and the main output function to be selected. Each talkback function can be configured from two choices. In most cases the Model 236 will be used on-air and the default mode, *Mutes*, will be appropriate. When this mode is selected, activating a talkback channel will cause the main output function, if active, to become inactive and mute the Dante Main transmitter (output) channel. (It will also cause the analog microphone output to mute if it's configured to follow the status of the Dante Main transmitter (output) channel.) This setting would be correct in traditional broadcast applications where a user's microphone audio signal needs to be off-air when communicating ("talking back") to a producer, director, or other production support personnel. For other applications the *None* configuration setting could prove useful. It allows a talkback function to become active while having no impact on the main output function. In this way, the main and talkback functions can operate independently.

Tone Detect Function

The Model 236 can be configured to respond to a high-frequency tone signal that is present in a Dante Talkback receiver (input) channel. A continuous signal of approximately 15 kHz or greater that's in-band (part of the audio content in a channel) will be detected as an active tone signal. An STcontroller configuration choice allows the tone detect function for each talkback channel to be independently selected from two quite different actions as well as being disabled. When set for *Call Light* the associated talkback channel's rotary encoder knob will flash orange when a tone is detected. The multi-purpose display will also flash orange. By design the multi-purpose display will flash a maximum of five times; the rotary encoder knob will flash as long as a call tone is detected. This is provided so that the Model 236 can be utilized in intercom environments where call light signals are present. For example, many products from Studio Technologies utilize high-frequency tones for call light signaling.

The *IFB (Dims Main Phones Audio)* mode allows talent cue (IFB) functionality to be created directly within a Model 236. This can be extremely useful in REMI/At-Home applications. When this mode is selected and a high-frequency tone is detected the level of the audio signals coming from the Dante Main receiver (input) channels and routed to the headphone output channels will be attenuated (dimmed) by 15 dB. In this way, audio arriving on a Dante Talkback input channel will be sent to the headphone outputs at a normal level while signals from the Dante Main input channels will be momentarily dimmed. Details on this function are provided in the Technical Notes section.

Additional Capabilities

An installer may have implemented additional Model 236 capabilities. A competent technician can easily add several functions, such as supporting remote control inputs, providing a tally output, or adding a multi-pin XLR connector for headset use. From the factory the Model 236 includes two unused connector locations on the enclosure's back panel. Internally, the Model 236 includes a number of 3-pin "header" connectors which provide access to various points in the unit's circuitry. These resources allow the simple installation of application-specific connectors and associated wiring.

Connections can be made to the four remote control inputs which are typically interfaced using a 5-pin XLR connector. The remote control inputs are voltage, current, and ESD protected, allowing direct connection to external devices such foot switches or contact closures. Configuration selections in STcontroller allow these inputs to mimic the actions of the main and talkback pushbutton switches.

One remote control input can also be configured to provide a low-voltage, low-current DC output that is active whenever the Model 236's main function is active. This is intended to serve as an on-air tally output to directly light an LED or activate an input on another piece of equipment.

A 6- or 7-conductor XLR connector may have been added to allow the direct connection of a broadcast-type headset that integrates both stereo headphones as well as a high-performance microphone.

Technical Notes

IP Address Assignment

By default, the Model 236's Ethernet interface will attempt to automatically obtain an IP address and associated settings using DHCP (Dynamic Host Configuration Protocol). If a DHCP server is not detected an IP address will automatically be assigned using the link-local protocol. This protocol is known in the Microsoft® world as Automatic Private IP Addressing (APIPA). It is also sometimes referred to as auto-IP (PIPPA). Link-local will randomly assign a unique IP address in the IPv4 range of 169.254.0.1 to 169.254.255.254. In this way, multiple Dante-enabled devices can be connected together and automatically function, whether or not a DHCP server is active on the LAN. Even two Dante-enabled devices that are directly interconnected using an RJ45 patch cord and receiving power should correctly acquire IP addresses and be able to communicate with each other.

As previously discussed, the Model 236's Dante IP address and related network parameters can be manually set for a fixed (static) configuration using the Dante Controller software application. In the unfortunate event that a specific Model 236's IP address is "lost" there are several techniques that may prove useful. It's possible that the Address Resolution Protocol (ARP) networking command can be used to "probe" devices on a network for this information. For example, in Windows OS the **arp –a** command can be used to display a list of LAN information that includes MAC addresses and corresponding IP addresses. (Within this list should be the Model 236's statically assigned Dante IP address.) This method is most effective with a LAN that uses unmanaged Ethernet switches. Contemporary LANs that utilize managed switches may limit the ARP command's effectiveness.

In general, the simplest means of identifying an unknown IP address is to create a "mini" LAN with an unmanaged Ethernet switch. Then by using the appropriate ARP command the required "clues" can be obtained. It's possible to make a direct connection between a personal computer and a Model 236. The only additional piece of equipment necessary would be a 12 volt DC power supply for providing power to the Model 236.

Optimizing Network Performance

For best Dante audio-over-Ethernet performance a network that supports VoIP QoS (voice-over-Internet-protocol quality of service) capability is recommended. This can typically be implemented on virtually all contemporary managed Ethernet switches. There are even specialized switches that are optimized for entertainment-associated applications. Refer to the Audinate website (audinate.com) for details on how to optimize a network for Dante applications.

Dante Aux and Talkback Output Levels

A level anomaly can occur with the Model 236's Dante Aux and Talkback transmitter (output) channels. This condition will only occur if two configuration conditions are met. The first is that the analog microphone input must be selected as the microphone source. The second is that the analog microphone output must be configured to mute whenever the main output function is also muted. If these two configurations are in effect then in some conditions a 400-ohm load will be applied to the analog microphone input. (This 400-ohm load is part of the analog microphone output's hardware muting circuit.) Adding 400 ohms across the connected analog microphone will typically reduce its output level by 1 to 3 dB. Not surprisingly, this reduction in signal level will result in the post-analog preamplifier level also being 1 to 3 dB lower. The exact amount of attenuation will depend on the specific microphone that is connected to the unit's analog microphone input. The impact of an additional load of 400 ohms will have will depend on the characteristics of the microphone. Typically, the lower its source impedance the less an impact the additional load of 400 ohms will have on it.

The level on the Dante Main transmitter (output) channel will never be impacted because the additional 400-ohm load will never be placed on the analog microphone input when the main function is active. This can be considered as the Model 236's Dante reference output level. However, if the Dante Aux output function has been configured for *Hot Mic* operation an odd condition can occur. In this case, whenever the main output is active the signal on the Dante Aux transmitter (output) channel will be at the same level as the Dante reference level. (The main and hot mic outputs will be at the same level.) But whenever the main output function is not active, the analog microphone output will be placed in its muted state and the level on the Dante Aux transmitter (output) channel will drop due to the additional 400-ohm load being placed on the analog microphone. The hot mic output's level will change by 1 to 3 dB in real-time as the analog muting circuit changes from inactive to active or vice-versa. This could prevent effective use of the hot mic function.

The level on the Dante Talkback transmitter (output) channels can also be impacted by the analog microphone muting circuit. If the analog microphone output has been configured for *Switched* then the analog microphone signal level will always be impacted by the additional 400 ohms being applied whenever the main output function is not active. This will typically happen whenever a talkback function is active. In this case, the output level on a Dante Talkback transmitter (output) channel would always be 1 to 3 dB less than the Model 236's Dante reference level. This wouldn't typically be an issue but could cause some confusion should precise system level testing be performed. (The main output will be 1 to 3 dB "hotter" than the talkback output channels.)

Is there a way to have this not become an issue? Yes. Unless absolutely necessary, ensure that the analog microphone output is configured for its *Always On* mode. This will prevent the analog muting circuit from ever becoming active and ensure that the additional 400-ohm load won't be placed on the connected analog microphone. If the analog microphone output does need to be configured for *Switched* then ensure that its ramifications are understood. In this case, the hot mic function shouldn't be utilized. And a slight level reduction in the talkback outputs should be expected.

Button Labeling

The Model 236 provides pushbutton switches that are associated with the main and talkback functions. The button covers are clear with a white translucent lens underneath. As of the time of writing this guide the factory doesn't have a recommended method as to how to install custom labels underneath the button caps. We have observed units where this has been successfully implemented but exactly how it was executed is not known. It's possible that a button cap can be removed using a gentle prying force on its bottom edge using a straight-blade screwdriver. An Xcelite® "greenie" screwdriver, ubiquitous to the audio and broadcast industry, might be appropriate. Once a cap is removed a custom label could be inserted between it and the translucent lens. Then the cap could be "snapped" back onto the body of the switch.

Additional Connector Locations

Two spare connector locations, labeled A and B, are provided on the Model 236's back panel. From the factory each contains a blank plate that can be removed and replaced with a variety of XLR connectors. The spare connector locations are included so that a Model 236 can be customized to meet the many specific needs that arise in broadcast and related audio applications. Expected uses for these locations include adding support for the remote control inputs, the tally output, or incorporating a 6- or 7-pin XLR connector to allow direct connection of a broadcast headset. A number of connector and interface cable assemblies, along with some special function kits, are available from Studio Technologies. Please refer to the website for details on what is available.

The spare connector locations are compatible with the Neutrik DL-series of connectors. For flexibility XLR versions are available from Neutrik that provide from three to seven contacts. For example, a compatible 3-pin female connector would be Neutrik part number NC3FD-L-1. Another female connector with the unique Switchcraft® 6-pin arrangement, the NC6FDS-L-1, is often used to support headsets. Other connectors, such as the etherCON protected RJ45 and 3-conductor ¼-inch jack, can be also be installed. The 4-40 thread pitch hardware that secures the blank plates to the Model 236's back panel are also intended to be used to secure the selected connectors.

The Model 236's enclosure must be disassembled prior to installing a connector in a spare location. Ensure that power is removed from the Model 236 prior to disassembling the enclosure. Six 4-40 flathead machine screws, one on each side of the enclosure, two on the bottom front of the enclosure, and two on the back panel, must be removed. (The four machine screws associated with the microphone input and microphone output connectors should not be removed.) These 4-40 screws are best removed using a #1 Phillips-head screwdriver. Once the six screws are removed the cover can then be carefully separated from the chassis. The cover will remain tethered to the chassis by means of a flexible cable assembly. This "flex-cable" links the motherboard (main printed circuit board assembly) with the button and rotary encoder board assembly. Ensure that the flex cable is not damaged while the Model 236 is being customized.

If one or two connectors are installed in the Model 236's spare connector locations labeling them can be helpful. For a good look it is recommended that a Brother® P-Touch ¼-inch (6 mm) label be created. Tape material that prints white text on a black background would be effective for use with the Model 236. The Brother label cassette number TX-3151, white on black, is appropriate for use with many of their printers.

3-Position Headers

As previously discussed, provision has been made to allow competent technicians to implement additional Model 236 functionality. This can be accomplished by interfacing with male "header" connectors located on the Model 236's main and analog circuit boards. These 3-position, 0.1-inch center, headers allow "no solder" solutions which makes customizing Model 236 units a relatively simple process. The headers are Molex® part number 22-23-2031. They mate with Molex housing number 22-01-3037. To make the interconnection, separate crimp terminals are attached to three loose wires and then "snapped" into the housing. Molex part number 08-50-0114 specifies crimp terminals that are appropriate for 22 to 30 gauge wires. These terminals are available worldwide from sources such as Digi-Key (digikey.com).

To make the process of connecting to the Model 236's headers an easy task an interface cable kit, Studio Technologies' part number 31087, is available. Each kit includes five cable assemblies and a length of heat-shrinkable tubing. Each cable assembly consists of a mating connector with three color-coded wires attached. These wires, approximately 12 inches in length, allow convenient soldering to a connector selected to be installed in a spare location on the Model 236's back panel. For reference, the wire color for pin 1 is gray, pin 2 is yellow, and pin 3 is blue. The heat-shrinkable tubing is provided so that the connector terminals or "solder cups" can be insulated from each other. It will also provide some strain relief to the solder joints. Be certain to slip the desired length of tubing over the wire prior to soldering a connection!

For details on each header's reference designator and functionality please refer to Appendix B.

Creating IFB within a Model 236

In STcontroller selecting IFB (Dims Main Phones Audio) as the configuration choice for a talkback channel's Tone Detect Function allows a broadcast type talent cue (IFB) function to be created inside a Model 236. IFB "feeds" are typically created in a fixed or mobile broadcast facility, most often in a sophisticated matrix intercom system. IFB signals can also be created using stand-alone IFB controllers or devices such as the Studio Technologies' Model 5422A Dante Intercom Audio Engine. These IFB signals would then be routed, either locally or via a long-distance transmission path, to inputs on an announcer's console. The user of the announcer's console would then listen to the IFB signals by way of headphones or a broadcast-type headset. This works well but can be challenging to implement in REMI/At-Home applications where there may not be a central location with IFB generation capability. But resources in the Model 236 were specifically included to make IFB creation directly at the on-air talent's location a simple matter. Using the Model 236's tone detect function in the IFB (Dims Main Phones Audio) mode, a production control or separate producer or director location that provides interrupt audio along with an in-band high-frequency "pilot" or activation tone will allow a broadcast-standard IFB signal to be created. And the audio quality will be excellent with no clicks, pops, or other audio artifacts.

The actual action that creates an IFB "feed" within the Model 236 is very simple. If a tone is detected on the audio signal arriving on a Dante Talkback receiver (input) the tone detect function will respond by attenuating (dimming or reducing in level) the audio associated with the Dante Main inputs in the headphone channel or channels that the interrupt audio has been assigned to. The interrupt audio with high-frequency tone present will first pass through a low-pass filter to remove the high-frequency tone and then be routed to the selected headphone output channel(s) at normal level. In this way, a program-with-interrupt IFB channel is created.

An example of using the Model 236's IFB capabilities for an on-air sports application will be detailed in this paragraph. A 2-channel IFB "feed" will typically have program audio with interrupt audio in its left channel and program-only audio in its right channel. Program audio would enter the Model 236 by way of two of the Dante receiver (input) channels: Main 1 L and Main 2 R. This would route the program audio to both the left and right channels of the headphone output, each with independent level control using the Main 1 and Main 2 rotary encoders. The interrupt audio source would enter the Model 236 by way of the designated Dante Talkback receiver (input) channel. This interrupt audio source would include both voice (the actual interrupt content destined for the on-air talent) and a high-frequency tone, both of which should only be present when IFB (interrupt) action is desired. Using STcontroller the interrupt audio source from the Dante talkback receiver (input) would be assigned to the left headphone channel. And finally, again in STcontroller, the Tone Detect Function for the applicable talkback channel would be configured for *IFB (Dims Main Phones Audio)*.

Now whenever the interrupt signal (voice and high-frequency tone) is present on the talkback input an IFB (interrupt) condition will be detected and the following actions will take place: the program audio source being routed to the left headphone channel will attenuate ("dim") by 15 dB, and interrupt audio will be routed to the left headphone channel. The user would hear the interrupt audio (at a level controlled by its associated talkback rotary encoder) along with program audio that has been attenuated by 15 dB. Once the high-frequency tone is no longer present in the interrupt source the IFB (interrupt) condition would end; interrupt audio would no longer be sent to the left headphone channel and the program audio being sent to the left headphone channel would return to its normal (non-dimmed) level. Whatever the status of the IFB function, the level of the program audio being sent to the right channel of the headphone output would remain constant.

Limiter Function

A limiter function can be enabled to control the maximum level of the output of the microphone preamplifier. This applies only to a signal that has been connected to the analog microphone input connector on the back panel. The design intent of the limiter function is to prevent digital "clipping" of the signal as it is presented on a Dante output channel. It's very possible that an audio professional, or even a "listener", will be able to detect when the limiter function is active. That shouldn't prove to be an operational issue, the limiter should prevent the horrible sound that a clipped audio signal would make. The remedy that will minimize the chance that a Model 236 will have its limiter function frequently active is to revise the microphone preamplifier gain configuration using the STcontroller software application. Reducing the amount of gain should reduce the need for the limiter function to become active. Shaping the "dynamics" characteristics of the audio signal associated with the Model 236's microphone input should be performed by a device further on in the audio signal chain, not within the Model 236. That's an artistic choice that the Model 236 is not intended to be involved with.

The limiter function is created using digital control of the microphone preamplifier integrated circuit. When enabled, the limiter function will become active when the signal level reaches a threshold of –8 dBFS, which equates to being within 8 dB of the 0 dBFS maximum as measured on a Dante transmitter (output) channel. (This is the same as describing the threshold as 12 dB above the –20 dBFS nominal). When the signal level is detected as reaching the threshold the gain of the microphone preamplifier will be reduced in one or two dB steps. The maximum amount of gain reduction is related to the microphone preamplifier's gain configuration. When the gain has been configured to be 36 dB or greater the maximum gain reduction will be 16 dB. With the preamplifier gain is configured for 20 dB there will be no gain reduction. Between 21 and 35 dB of microphone preamplifier gain the maximum reduction will correspond to 1 to 15 dB. (So, for example, gain that is configured for 28 dB will allow a maximum limiter gain reduction of 8 dB.)

The performance of the limiter has been optimized for voice signals. The parameters are not adjustable. The speed at which the limiter becomes active (the "attack" time) has been made as fast as the circuitry allows. The speed at which the limiter is no longer active (the "release time") is much longer. The gain reduction will be maintained for a minimum of 30 milliseconds. The only indication that the limiter function is active is by observing the icon in the Model 236 configuration page within the STcontroller software application.

Application Firmware and Version Display

The Model 236 has three firmware (embedded software) components. These are the main firmware, the secondary firmware, and the FPGA firmware. The main firmware is used by the main microcontroller integrated circuit (located on the motherboard) to implement most of the Model 236's general operating actions. The secondary firmware is used by a second, less-sophisticated microcontroller integrated circuit that controls and reports on the actions of the rotary encoders, the pushbutton switches, and the LEDs associated with them. The FPGA firmware is used by the field-programmable-gate-array integrated circuit (also located on the motherboard) in which all audio processing is performed.

To observe the firmware version numbers, begin by connecting the Model 236 unit to local area network (LAN). A source of 12 volts DC is required if a PoE Ethernet connection has not been made. Once power is applied the unit will go through a boot-up (start-up) sequence and then begin to function. Using STcontroller review the list of identified devices and select the specific Model 236 from which you want to determine its application firmware versions. Then select **Version and Information** under the **Device** tab. A dialog will then display that will provide a host of useful information. This will include the firmware versions as well as details on the Dante interface firmware.

Application Firmware Update Procedure

It's possible that updated versions of the three firmware (embedded software) files that are utilized by the Model 236's two microcontroller and one FPGA (field-programmable-gate-array) integrated circuit will be released to add features or correct issues. Refer to the Studio Technologies' website for the latest firmware files. The unit has the ability to load revised files into non-volatile memory by way of a standard USB flash drive. The Model 236 implements a USB host function and provides access by way of a type A receptacle located on the back panel. The Model 236 updates its firmware using files named **M236.bin** for the main firmware, **M23Xsec.bin** for the secondary firmware, and **BWY.bit** for the FPGA firmware.

The update process begins by preparing a USB flash drive. The flash drive doesn't have to be empty (blank) but must be in the personal-computer-standard FAT32 format. One, two, or all three of the firmware files can be automatically loaded into a Model 236 at the same time. In the flash drive's root folder, save the desired new firmware file(s) ensuring that the required name or names are specified. For the main firmware the file name must be **M236.bin**. For the secondary firmware the file name must be **M23Xsec.bin**. And for the FPGA firmware the file name must be **BWY.bit**. Studio Technologies will supply each firmware file inside a .zip archive file. While the firmware file inside of the zip file will adhere to the naming convention required by the Model 236, the name of the zip file itself will include the file's version number. For example, a file named **M236v3r05MCU.zip** would indicate that version 3.05 of the main MCU firmware (**M236.bin**) is contained within this zip file.

Once the USB flash drive is inserted into the USB interface, located on the Model 236's back panel, the unit must be powered off and again powered on. At this point the file(s) will automatically load. The precise steps required will be highlighted in the next paragraphs.

To install one, two, or all three firmware files follow these steps:

1. Remove power from the Model 236. This will entail either removing the Ethernet connection from the RJ45 jack labeled Primary-PoE or removing the external source of 12 volts DC. (Both must be disconnected if for some reason dual-powering has been implemented.)
2. Locate the USB connector on the Model 236's back panel labeled Firmware. Directly below the USB connector is a small hole that provides visual access to a green LED indicator.
3. Insert the prepared USB flash drive into the USB connector.
4. Apply power to the Model 236. Power can be provided by either Power-over-Ethernet (PoE) associated with an Ethernet signal connected to Primary-PoE jack or from a 12 volts DC source.
5. After a few seconds the Model 236 will run a "boot loader" program that will automatically load and save the new firmware file(s) that are present on the USB flash drive. The time required by the update process can range from approximately 15 seconds to approximately 45 seconds, depending on which combination of the three files are going to be updated. While files are being loaded the green LED, located below the USB connector, will flash slowly. It's also possible that the knobs associated with the rotary encoders will light red to indicate that normal activity is not taking place. Once the entire loading process has completed the Model 236 will restart using the newly saved firmware.
6. At this time the Model 236 will be functioning with the newly loaded firmware and the USB flash drive can be removed. But to be conservative, remove the power source first and then remove the USB flash drive.
7. Confirm that the desired firmware has been loaded and that the version numbers of the Model 236's three firmware files are as expected. This is easily accomplished using STcontroller. (Use the **Version and Information** selection under the **Device** tab.)

Note that upon power being applied to the Model 236 if the USB flash drive doesn't have the correct file(s) in its root folder no harm will occur. Upon power up the green LED, located below the USB connector, will flash on and off rapidly for a few seconds to indicate that a valid file has not been found. Then normal operation will commence using the unit's existing firmware files.

Broadway Firmware Update

As previously discussed, the Model 236 implements Dante connectivity using the Broadway integrated circuit from Audinate. The Dante Controller software application can be used to determine the version of the firmware (embedded software) that resides in Broadway. STcontroller can also be used to identify the Dante firmware version. (Use the **Version** selection under the **Device** tab.) The Dante Firmware Update Manager (FUM) application program has traditionally been used to update the Broadway firmware. Recent versions of the Dante Controller software application also include an automated method of updating Broadway firmware. Both applications are available, free of charge, on the Audinate website (audinate.com).

Restoring Factory Defaults

A command in STcontroller allows the Model 236's configuration selections to be reset to their factory default values. From within STcontroller select the specific Model 236 unit for which you want to restore its defaults. Select the **Device** tab and then select the **Factory Defaults** feature. Then click on the **OK** box. Refer to Appendix A for a list of the factory default values.

Specifications

Network Audio Technology:

Type: Dante audio-over-Ethernet

AES67-2018 Support: yes, selectable on/off

Dante Domain Manager (DDM) Support: yes

Ethernet Interface Configuration: Switched or Redundant

Bit Depth: 16, 24, or 32

Sample Rate: 48 or 96 kHz

Pull-Up/Down Support: no

Dante Transmitter (Output) Channels: 10

Dante Receiver (Input) Channels: 11

Dante Audio Flows: 32; 16 transmitter, 16 receiver

Internal Digital Audio Processing: 32-bit, fixed

Input-to-Output Audio Processing Latency: <200 uSec

Nominal Dante Receiver (Input) and Transmitter (Output) Level: –20 dBFS

Network Interfaces: 2, Primary-PoE and Secondary

Type: 1000BASE-T Gigabit Ethernet (GigE) per IEEE® 802.3ab (10 and 100 Mb/s not supported)

Power-over-Ethernet (PoE): per IEEE 802.3af (applicable to the Primary-PoE network connection only)

Analog Microphone Input:

Compatibility: dynamic or phantom-powered microphones

Type: balanced, capacitive coupled

Impedance: 3.6 k ohms, nominal

Gain: 20 to 65 dB in 1-dB steps

Frequency Response: 20 Hz to 22 kHz, +0/–1 dB

Distortion (THD+N): <0.0004%, 1 kHz, –4 dBFS output, 20 dB gain, 22 kHz BW

Dynamic Range: 120 dB, ref 0 dBFS output, 20 dB gain, A-weighted

Phantom Power: P48 per IEC 61938 standard, on/off selectable with status LED

Limiter Function:

Type: digitally controlled analog

Application: applies only to analog microphone input, selectable on/off

Threshold: 12 dB above nominal level (–8 dBFS)

Minimum On Time: 30 milliseconds

Maximum Attenuation: 16 dB at 36 dB or greater microphone preamplifier gain

Analog Microphone Output:

Type: passive via two 200 ohm resistors from microphone input (will pass phantom power in either direction)

Muting: solid-state relay contacts, 60 dB attenuation @ 1 kHz

Headphone Output:

Type: analog 2-channel (stereo)

Compatibility: stereo (dual-channel) or monaural (single-channel) headphones, headsets, or earpieces with nominal impedance of 50 ohms or greater

Maximum Output Voltage: 3.0 Vrms, 1 kHz, 150 ohm load

Frequency Response: 20 Hz to 20 kHz, +0/–1 dB

Dynamic Range: >102 dB, A-weighted

Distortion (THD+N): <0.025%, measured at 1 kHz, 0 dBu output

Talkback Input Low-Pass Filter (LPF):

Function: active only when specific talkback channel has tone detect function enabled

Type: digital, 8-pole Butterworth (48 dB/octave)

Cutoff: –3 dB at 12 kHz

18 kHz and 20 kHz Tone Outputs:

Type: sine wave

Level: –20 dBFS

Frequency Accuracy: <10 ppm

Distortion: <0.0001%

Tone Detect Function:

Receive Frequency: ≥15 kHz nominal to upper end of frequency response, within audio channel

Receive Level: –27 dBFS minimum

Send Frequency: 20 kHz

Send Level: –20 dBFS

IFB (Dims Main Phones Audio) Function:

Activation: in-band tone detection per Tone Detect function

Attenuation (Dim): 15 dB, fixed

Call Light Function:

Activation: in-band tone detection per Tone Detect function

Remote Control Inputs: 4

Function: configurable, can mimic the action of the main and talkback button functions

Type: active low, 1 mA maximum, input pulled up to 3.3 volts DC via 3.4 k ohm resistors

Limitation: Remote Control Input 2 shares hardware pin with Tally Output (only one function can be configured at a time)

Tally Output:

Functions: follows the status of the main output

Type: active high, 3.3 volts DC in series with 130 ohm resistor (25 mA maximum)

Limitation: tally output shares hardware pin with Remote Control Input 2 (only one function can be configured at a time)

Connectors:

Analog Microphone Input: 3-pin female XLR

Analog Microphone Output: 3-pin male XLR

Headphone Output: 3-conductor ¼-inch jack

Ethernet: 2, Neutrik NE8FBH etherCON RJ45 receptacles

DC Input: 4-pin male XLR (pin 1 negative, pin 4 positive)

USB: type A receptacle (used only for updating firmware)

Remote Control Inputs / Tally Output: 2, 3-pin headers located on the main circuit board

Headphone Output: 3-pin header located on the main circuit board

DC Output: 3-pin header located on the main circuit board

Analog Microphone Input: 3-pin header located on the analog circuit board

Spare Connector Locations: 2

Allows a Studio Technologies' cable assembly or option module to be installed. Also compatible with Neutrik NC*D-L-1 connectors (*=3F, 3M, 5M, 6F, 6FS, etc.).

Configuration: requires Studio Technologies' STcontroller software application

Power Sources:

Power-over-Ethernet (PoE): class 3 (mid power, ≤12.95 watts) per IEEE 802.3af

DC Input: 10 to 18 volts DC, 0.55 A maximum at 12 volts DC (can be powered by optional PS-DC-02)

Environmental:

Operating Temperature: 0 to 50 degrees C (32 to 122 degrees F)

Storage Temperature: –30 to 70 degrees C (–22 to 158 degrees F)

Humidity: 25 to 85%, non-condensing

Altitude: not characterized

Dimensions (Overall):

6.5 inches wide (16.5 cm)
2.9 inches high (7.4 cm)
4.9 inches deep (12.5 cm)

Deployment: intended for tabletop applications

Weight: 2.1 pounds (0.95 kg)

Specifications and information contained in this User Guide subject to change without notice.

Appendix A: STcontroller Default Configuration Values

General Menu Page:

Microphone Input – Analog Mic In P48: Off

Microphone Input – Analog Mic In Limiter: Off

Microphone Input – Source: Analog Mic In

Microphone Input – Analog Gain: 50 dB

Microphone Input – Dante Trim: 0 dB

System – Mic Level Display: Always On

System – Signal Present Display: All Inputs

System – LED Intensity and Action: High

System – Dante Aux Output: Off

System – Analog Mic Output: Always On

System – Encoder Minimum Level: Full Mute

System – Sidetone Mode: Talkback Buttons

System – Sidetone Routing: Left and Right

System – Remote Control In 1: Off

System – Remote Control In 2 / Tally Out: Off

System – Remote Control In 3: Off

System – Remote Control In 4: Off

Main – Button Mode: Push to Mute

Main – Encoder 1 Tap to Mute: Enabled

Main – Encoder 2 Tap to Mute: Enabled

Talkback Channels Menu Page (parameters identical for all Talkback channels):

Phones Routing: Left and Right

Listen Mode: Always

Encoder Tap to Mute: Enabled

Button Mode: Hybrid

Button Function: Talk

Impact on Main Output: Mutes

Tone Detect Function: None

Appendix B: 3-Pin Header Connector Details

The following list provides details on the 3-pin header connectors located on the Model 236's printed circuit boards. Shown are both reference numbers and associated functions.

I. Header on the Analog Board:

P4: Analog Microphone Input
Pin 1: Common / Shield
Pin 2: High (+)
Pin 3: Low (–)
Note: These pins are electrically in parallel with pins 1, 2, and 3 respectively of the back panel 3-pin female XLR analog mic in connector.

II. Headers on the Main ("Broadway") Board:

P4: Headphone Output
Pin 1: Common (–)
Pin 2: Left Channel (Tip)
Pin 3: Right Channel (Ring)
Note: These pins are electrically in parallel with the Sleeve, Tip, and Ring connections respectively of the back-panel 3-conductor ¼-inch connector.

P7: Remote In 1, Remote In 2 / Main Tally Out
Pin 1: Common (–)
Pin 2: Remote In 1 (+)
Pin 3: Remote In 2 / Main Tally Out (+)
Note: Inputs pulled up to 3.3 volts DC via 3.4 k ohm resistors. Output active high, 3.3 volts DC in series with 130 ohm resistor (25 mA maximum).

P9: Remote In 3, Remote In 4
Pin 1: Common (–)
Pin 2: Remote In 3 (+)
Pin 3: Remote In 4 (+)
Note: Inputs pulled up to 3.3 volts DC Inputs via 3.4 k ohm resistors.

P8: DC Output
Pin 1: Common (–)
Pin 2: +12 volts DC Nominal; range 10-18 volts DC
Pin 3: Not Used
Note: Output not current limited – be careful!

Appendix C: Block Diagram

The following block diagram shows a simplified version of the microphone input and microphone output circuitry.