Model 371A Intercom Beltpack

User Guide

Issue 1, June 2018

This User Guide is applicable for serial numbers
M371A-02501 and later with application firmware 1.1 and later
and Dante® firmware 1.0.0 (UltimoX2 4.1.2.1) and later

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Revision History

Issue 1, June 2018:

1. Initial release.

Introduction

The Model 371A Intercom Beltpack begins with the features offered by 2-channel broadcast-style party-line intercom user devices then adds support for traditional single-channel headsets. Applications that have traditionally deployed single-channel analog party-line intercom systems from firms such as Clear-Com® can now easily upgrade to a high-performance dual-channel system. To maintain compatibility and user familiarity, legacy headsets that use a 4-pin female XLR connector are directly compatible.

The Model 371A includes a range of new capabilities, along with the advanced performance and flexibility that Dante® audio-over-Ethernet provides. Over a standard IP network, multiple Model 371A units can be used in party-line (PL) intercom applications with help from an external Dante-enabled audio matrix such as the Studio Technologies Model 5422 Dante Intercom Audio Engine. Alternately, units can be used "point-to-point" or interfaced with Dante-compatible matrix intercom systems.

Only a single Power-over-Ethernet (PoE) connection is required for operation. Key user features can be easily configured using the STcontroller software application. Configurable parameters include electret microphone powering, microphone preamplifier gain, sidetone level, and talk button operation. User features include microphone dynamic range control, call function, and remote mic kill ("talk off"). The wide range of capabilities, along with the excellent audio quality provided by the digital audio signal path, offers a unique and powerful user experience.

Set up and configuration of the Model 371A is simple. An etherCON® RJ45 receptacle is used to interconnect with a standard twisted-pair Ethernet port associated with a local-area network (LAN). This connection provides both power and bidirectional digital audio. A broadcast or intercom-style headset with a dynamic or electret (low-voltage DC-powered) microphone interfaces with unit using a 4-pin XLR connector. The Model 371A's enclosure is made from an aluminum alloy which offers both light weight and ruggedness. A stainless steel "belt clip," located on the back of the unit, allows direct attachment to a user's clothing.

The audio quality of the Model 371A's audio channels is excellent, with low distortion, low noise, and high headroom. Careful circuit design and rugged components ensure long, reliable operation. A wide range of applications can be supported, including theater facilities, over-the-air and streaming broadcasting,

Figure 1. Model 371A Intercom Beltpack top and bottom views

corporate and government AV installations, aerospace command and control, and post-production facilities.

Dante Audio-over-Ethernet

Audio data is sent to and received from the Model 371A using the Dante audio-over-Ethernet media networking technology. As a Dante-compliant device, the Model 371A's two audio output (Dante transmitter) and two audio input (Dante receiver) channels can be interconnected (routed) with other devices using the Dante Controller software application. The Dante transmitter and receiver channels are limited to supporting four Dante flows, two in each direction. The digital audio's bit depth is up to 24 with a sampling rate of 48 kHz. The Model 371A is AES67 compatible and compliant with the Dante Domain Manager™ software application.

Two bi-color LEDs provide an indication of the Dante connection status. The Dante Controller's *Identify* command takes on a unique role with the Model 371A. Not only will it cause the talk and call button LEDs to light in a unique highly visible sequence, it can also be configured to turn off any active talk channels.

Audio Quality

The Model 371A's completely "pro" performance is counter to the less-than-stellar reputation of typical party-line (PL) intercom audio. A low-noise, wide dynamic-range microphone preamplifier and associated voltage-controlled-amplifier (VCA) dynamics controller (compressor) ensures that microphone audio quality is preserved while minimizing the chance of signal overload. DC power to support electret microphones can be enabled if necessary. The output of the microphone preamp and compressor is routed to an analog-to-digital converter (ADC) section that supports a sampling rate of 48 kHz with a bit depth of up to 24. The audio signal, now in the digital domain, travels through the processor and on to the Dante interface section where it is packetized and prepared for transport over Ethernet.

Audio input signals arrive via two Dante receiver channels. The supported sampling rate is 48 kHz with a bit depth of up to 24. The audio signals pass into the Model 371A's processor where channel routing, headphone level control, and sidetone creation are performed within the digital domain. Two push-in/push-out ("pop out") rotary controls make it easy to set and maintain the desired headphone output level. The audio signal destined for the headphone output is sent to a high-performance digital-to-analog converter and then on to a robust driver circuit. A high signal level can be provided to a variety of headsets, headphones, and earpieces.

Call Function

A call function allows Model 371A users to send and receive channel-specific visual alert signals. Pressing the call pushbutton on the top of the unit is all that's required to signal other users that attention is requested. Using 20 kHz tones, the call signals are sent within the audio channels ("in band") allowing interoperability between multiple Model 371A units as well as being compatible with legacy party-line intercom systems. The call function can be useful to indicate to users that they are needed "on headset" or should be actively listening to an intercom channel. The call function can also be used to provide real-time cues to production personnel during the running of live events.

Configuration Flexibility

A highlight of the Model 371A is its ability to be easily configured to meet the needs of specific users and applications. All configuration choices are made using the STcontroller software application that communicates with the Model 371A by way of an Ethernet network connection. Configurable parameters include microphone power and preamplifier gain, sidetone audio operation, and talk button operation.

The microphone input can be selected for compatibility with dynamic or elected (DC-powered) microphones. The gain of the microphone preamplifier can be selected from among four choices, two for supporting dynamic microphones and two for supporting electret microphones. These choices allow compatibility with the variety of microphones that are part of broadcast and intercom headsets.

A unique Model 371A feature is the ability to individually configure the way in which the two talk pushbutton switches function; four choices are available. For standard intercom beltpack operation it's typical to select *Push to Talk*, *Latching*, or *Push to Talk/Tap to Latch*. For situations where only monitoring of an intercom channel is desired a *Disabled* mode is available.

Ethernet Data and PoE

The Model 371A connects to an Ethernet data network using a standard 100 Mb/s twisted-pair Ethernet interface. The physical interconnection is made by way of a Neutrik® etherCON RJ45 receptacle. While compatible with standard RJ45 plugs, etherCON allows a ruggedized and locking interconnection for harsh or high-reliability environments. An LED displays the status of the network connection.

The Model 371A's operating power is provided by way of the Ethernet interface using the 802.3af Power-over-Ethernet (PoE) standard. This allows fast and efficient interconnection with the associated data network. To support PoE power management, the Model 371A's PoE interface reports to the power sourcing equipment (PSE) that it's a class 1 (very low power) device.

Future Capabilities and Firmware Updating

The Model 371A was designed such that its capabilities and performance can be enhanced in the future. A USB connector, located on the unit's main circuit board (underneath the unit's cover), allows the application firmware (embedded software) to be updated using a USB flash drive.

The Model 371A uses Audinate's Ultimo™ integrated circuit to implement the Dante interface. The firmware in this integrated circuit can be updated via the 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 371A Intercom Beltpack and a printed copy of this guide. As a device that is Power-over-Ethernet (PoE) powered, no external power source is provided. In most applications an Ethernet switch with the required PoE capability will be utilized.

Connections

In this section signal interconnections will be made using the two connectors located on the bottom of the Model 371A's enclosure. An Ethernet data connection with Power-over-Ethernet (PoE) capability will be made using either a standard RJ45 patch cable or an etherCON protected RJ45 plug. A headset that has a cable-mounted 4-pin female XLR connector will be plugged into the Model 371A's 4-pin male headset connector.

Ethernet Connection with PoE

A 100BASE-TX Ethernet connection that supports Power-over-Ethernet (PoE) is required for Model 371A operation. This one connection will provide both the Ethernet data interface and power for the Model 371A's circuitry. A 10BASE-T connection is not sufficient and a 1000BASE-T ("GigE") connection is not supported unless it can automatically "fall back" to 100BASE-TX operation. The Model 371A supports Ethernet switch power management, enumerating itself as a PoE class 1 device. Any source that is compliant with the IEEE® 802.3af standard will function correctly. Should the selected Ethernet switch port support Energy-Efficient Ethernet (EEE) it must be disabled to help ensure reliable Dante operation.

The Ethernet connection is made by way of a Neutrik etherCON protected RJ45 receptacle that is located on the bottom of the Model 371A's enclosure. This allows connection by way of a cable-mounted etherCON connector or a standard RJ45 plug. The Model 371A's Ethernet interface supports auto MDI/MDI-X so that a crossover cable will never be required.

Headset Connection

The Model 371A provides a 4-pin male XLR connector that interfaces with the microphone and headphone connections of an intercom or broadcast-style headset. Refer to Figure 2 for connection details. The microphone input connections are compatible with most unbalanced dynamic or electret (low-voltage DC-powered) microphones. A balanced dynamic microphone should, in most cases, also function correctly if its signal – (low) is connected to Model 371A's mic in –/shield connection. No support is provided for microphones that require P12 or P48 phantom power.

In most cases headsets associated with single-channel party-line intercom systems will be directly compatible. A monaural (single-ear or "single-muff") headset should be wired such that its –/shield/screen lead is connected to pin 3 and its + lead is connected to pin 4. To allow users of stereo (dual-ear or "double-muff") headsets to hear the Model 371A's headphone output in both ears requires that both its –/shield/screen leads be connected to pin 3 and both its + leads be connected to pin 4.

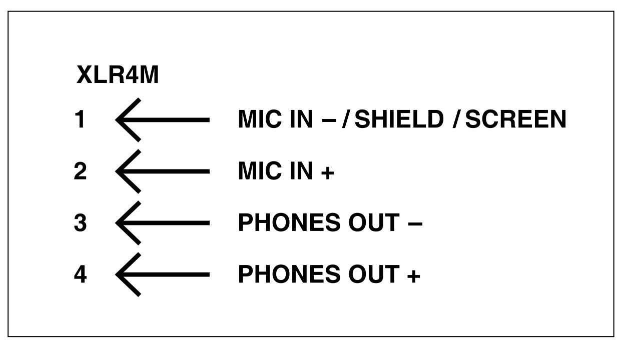

Figure 2. Headset connection pinout chart

Dante Configuration

For audio to pass to and from the Model A requires that several Dante-related parameters be configured. These configuration settings will be stored in non-volatile memory within the Model 371A's circuitry. Configuration will typically be done with the Dante Controller software application which is available for download free of charge at www.audinate.com. Versions of Dante Controller are available to support Windows® and OS X® operating systems. The Model 371A uses the Ultimo 2-input/ 2-output integrated circuit to implement the Dante architecture.

The two Dante transmitter (Tx) channels associated with the Model 371A's Dante interface must be assigned to the desired receiver (Rx) channels. This achieves routing the Model 371A's two talk output audio channels to the device (or devices) that will be "listening" to them. Within Dante Controller a "subscription" is the term used for routing a transmitter flow (a group of output channels) to a receiver flow (a group of input channels). The number of transmitter flows associated with an Ultimo integrated circuit is limited to two. These can either be unicast, multicast, or a combination of the two. If the Model 371A's transmitter channels need to be routed using more than two flows it's possible that an intermediary device, such as the Studio Technologies Model 5422 Dante Intercom Audio Engine, can be used to "repeat" the signals. (Use the Model 5422's pass-thru group configuration mode to provide this resource.)

The two Dante receiver (Rx) channels associated with the Model 371A's audio inputs also need to be routed to the Dante transmitter (Tx) channels provided by the desired source device. These two audio signals can be sent to the Model 371A's headphone output.

The Model 371A supports an audio sample rate of 48 kHz with no pull-up/pull-down values available. The Model 371A can serve as the clock master for a Dante network but in most cases it will be configured to "sync" to another device that is serving as the "grand master" for the Dante implementation.

The Model 371A has a default Dante device name of **ST-M371A-** followed by a unique suffix. The suffix identifies the specific Model 371A that is being configured. The suffix's actual alpha and/or numeric characters relate to the MAC address of the unit's Ultimo integrated circuit. The two Dante transmitter (Tx) channels have default names of **Ch1** and **Ch2**. The two Dante receiver (Rx) channels have default names of **Ch1** and **Ch2**. Using the Dante Controller application the default device name and channel names can be revised as appropriate for the specific application.

Model 371A Configuration

Many of the Model 371A's operating parameters can be configured to match the needs of specific applications. The STcontroller software application is used to observe a unit's current configuration and perform any changes that are required. No DIP switch settings or other local actions are used to configure the unit. This makes it imperative that the STcontroller software application be available for use in a personal computer that's connected to the related LAN.

STcontroller is available free of charge on the Studio Technologies website (www. studio-tech.com/stcontroller) and is compatible with personal computers running Windows operating systems that are version 7 and later. STcontroller versions 1.06.00 and later is required to support the Model 371A. If necessary, download and install STcontroller onto a designated personal computer. This personal computer must be on the same local area network (LAN) and subnet as the Model 371A unit or units that are to be configured.

STcontroller allows observation and selection of functions that include:

- Microphone input type (dynamic or electret)
- Microphone preamplifier gain
- Sidetone level
- Mic kill support
- Talk button operation

Changes made using STcontroller will be immediately reflected in the unit's operation; no Model 371A "reboot" is required. Each time a change is made the three buttons, talk 1, talk 2, and call, on the top panel will momentarily flash orange in a distinctive pattern to indicate that a command from STcontroller has been received.

Using STcontroller

Immediately after starting STcontroller the application will locate and display the devices that it can control. The one or more Model 371A units that are recognized will appear in the device list. Use the *Identify* command to allow easy recognition of a specific Model 371A unit. Double clicking on a device name will cause the associated configuration menu to appear. Review the current configuration and make changes as required.

Microphone Input Type

Choices are *Dynamic* and *Electret*.

If the associated headset has a dynamic (non-powered) microphone select the *Dynamic* input type. If the headset has an electret microphone that requires a source of low-voltage DC power for operation select *Electret*. Note that the Model 371A cannot supply P12 or P48 phantom power that may be required for balanced condenser (capacitor) microphones. This should not pose an issue as this type of microphone is essentially never associated with a headset that would be used for intercom applications.

Microphone Preamplifier Gain

Choices are *Low (42 dB)* and *High (48 dB)* for dynamic input type and *Low (22 dB)* and *High (30 dB)* for electret input type.

A total of four microphone preamplifier gain settings are provided. Which are available depends on the configuration of the microphone input type. Dynamic microphones have an output level that is typically lower than that provided by electret microphones. As such, the gain associated with the low and high settings when the input type is *Dynamic* are significantly greater than the gain choices available when the input type is *Electret*.

When the input type is *Dynamic* the *Low (42 dB)* gain setting will be appropriate for most applications. (Technically this gain is approximately equal to the gain provided within an RTS® BP-325 analog party-line beltpack.) The *Low (42 dB)* gain setting could also be valuable should the connected microphone have a high sensitivity (high output level for a given acoustical input) or an enthusiastic user routinely talks loudly into the microphone. Setting the gain for *High (48 dB)* may be helpful in some applications, such as with headsets that have a low microphone output level. The higher gain setting may also be useful when the Model 371A is going to be deployed at events where users are not able to speak at normal levels, e.g., sporting events such as golf tournaments.

Electret microphones typically have a higher output level due to their internal preamplifier. As such less preamplifier gain is required. Select either the *Low (22 dB)* or *High (30 dB)* setting as appropriate for the application.

The compressor active LED, visible on the bottom of the Model 371A's enclosure adjacent to the headset connector, can act as a guide when setting the microphone preamp gain. During normal talk operation the compressor active LED should light intermittently. If, for example, with a dynamic microphone the LED rarely lights and the preamp gain is set to *Low (42 dB)* it might be a good idea to change it to *High (48 dB)*. If the LED is lit fully during normal talking in a situation where the headset has an electret microphone and the gain is set for *High (30 dB)*, changing it to *Low (22 dB)* might be a good idea.

Sidetone Level

Choices are *Off*, *Low*, *Medium Low*, *Medium*, *Medium High*, and *High*.

The Model 371A includes a sidetone function that allows microphone audio coming from the microphone preamplifier to be sent to the headphone output. This serves as a user confirmation that they are actively sending audio to one or more of the talk output channels. Sidetone audio will only be routed to the headphone output when either or both of the talk channels is active. The level of the sidetone audio can be selected from among five values. The correct value is simply the one that makes the user most comfortable. Sidetone audio can also be disabled by selecting *Off*. Selecting *Off* would be useful only in special applications such as during troubleshooting or where user talk audio is being returned as part of their listen audio signals. The two rotary level controls will not impact the sidetone level.

Mic Kill Support

Choices are *Off*, *Device ID*, *Global Kill Command*, and *Device ID/Global Kill Command*.

The mic kill function allows an enabled talk channel to be disabled by remote control. Specifically, it's possible to remotely change all active (on) talk channels to their off state. This function is sometimes referred to as a "talk off" function. There are two means available to remotely disable all active talk channels. The *Device ID* method uses the identification command generated by the Dante Controller or STcontroller software applications as a "trigger" to disable any active talk channels. The I can be generated by applications that incorporate the Studio Technologies' proprietary method. This method utilizes sending a data command over the Ethernet connection. Select the desired mic kill configuration from the four choices. *Off* simply means that no remote talk control is supported.

Talk Button Operation

Choices are *Push to Talk*, *Latching*, *Push to Talk/Tap to Latch*, and *Disabled*.

This configuration section allows independent selection of how the two talk buttons will function. When the *Push to Talk* mode is selected the operation is self-explanatory. Only when a talk button is pressed and held will talk audio be sent out its associated Dante transmitter channel.

When the *Latching* mode is selected tapping (momentarily pressing) the button will cause the function to "latch" into the talk active mode. Tapping the button again will cause the function to “unlatch” and talk will no longer be active.

When the *Push to Talk/Tap to Latch* mode is selected a “hybrid” function of sorts is enabled. Pressing and holding the talk button will enable audio to be sent out its associated Dante transmitter channel. When the button is released audio will stop being sent out its associated transmitter channel. Tapping (momentarily pressing) the button will cause the function to “latch” into the talk active mode. Tapping the button again will cause the function to “unlatch” and talk will no longer be active.

Many applications are best served when the buttons are configured in the *Push to Talk* mode. This ensures that a channel can’t accidentally be left in its talk active mode. But there are certainly valid situations when the *Latching* or *Push to Talk/Tap to Latch* modes setting will prove to be very useful.

When the *Disabled* mode is selected microphone audio will never be sent out the associated Dante transmitter channel. This can be useful when the audio input associated with this channel is going to be monitored but talk audio is not allowed to “join” this specific intercom channel.

Operation

At this point everything should be ready and Model 371A operation can commence. An Ethernet connection with Power-over-Ethernet (PoE) capability should have been made. A headset terminated with a cable-mounted 4-pin female XLR connector should be plugged into the headset connector.

Using the Studio Technologies STcontroller software application the unit’s configuration should have been selected to meet the needs of the specific application. The Model 371A’s Dante configuration settings should have been selected using the Dante Controller software application. In this way the Model 371A’s two audio output channels (Dante transmitter channels) and two audio input channels (Dante receiver channels) should have been routed, by way of Dante “subscriptions,” to the receiver and transmitter channels on associated Dante-enabled equipment.

Initial Operation

The Model 371A will start to function as soon as a Power-over-Ethernet (PoE) power source is connected. However, it may take 20 to 30 seconds for full operation to commence. Upon initial power up the three status LEDs located on the bottom panel below the etherCON RJ45 receptacle will begin to light as network and Dante connections are established. The green and orange LEDs that illuminate the three pushbutton switches on the top panel may initially light randomly and then will light in a test sequence to indicate that the application firmware (embedded software) has started. Once the entire sequence has completed and the Dante connection has been established full operation will begin.

Ethernet and Dante Status LEDs

Three status LEDs are located below the etherCON RJ45 receptacle on the Model 371A’s bottom panel. The LINK ACT LED will light green whenever an active connection to a 100 Mb/s Ethernet network has been established. It will flash in response to Ethernet data packet activity. The SYS and SYNC LEDs display the operating status of the Dante interface and associated network. The SYS LED will light red upon Model 371A power up to indicate that the Dante interface is not ready. After a short interval it will light green to indicate that it is ready to pass data with another Dante device. The SYNC LED will light red when the Model 371A is not synchronized with a Dante network. It will light solid green when the Model 371A 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 371A is serving as the clock master for the network. It's possible that up to 30 seconds may be required for the SYNC LED to reach its final state.

Compressor Active LED

A yellow LED indicator is located on the bottom panel adjacent to the headset connector. Labeled COMP, the LED displays the status of the microphone audio compressor function. It will light whenever the input level from the microphone, along with the configured preamplifier gain, is such that the dynamic range of the talk signal is being controlled. It's perfectly acceptable for this LED to light intermittently whenever a user is talking at a normal voice level into the associated microphone. But if the COMP LED lights solid while a user is talking at a normal voice level this will typically indicate that the mic preamp gain setting should be reduced. Conversely, if the COMP LED almost never lights when normal talking is taking place, it's possible that changing the mic preamp gain to the higher setting would be beneficial. Note that due to the design of the circuitry the compressor active LED will function whether or not either or both of the talk channels are active.

Headphone Output

Two rotary potentiometers ("pots"), located on the Model 371A's top panel, allow individual adjustment of the level of the two audio input signals as they are sent to the headphone output. The audio signal that is sent to the headphone output will be the sum (mix) of the two input signals. The pots are "push in/push out" type which allow their associated knobs to be in their "out" position to be adjusted and then moved to their "in" position when protection from an unwanted change is desired

Users should find the headphone output audio quality to be excellent, with high maximum output level and low distortion. Audio signals do not pass directly through the level pots. The position of the pots is recognized by the Model 371A's processor which then adjusts the signal level within the digital domain. When a pot is in its fully counterclockwise position the associated audio signal is fully muted. The on/off status of the two talk channels, or the status of the call function, does not impact the headphone output.

Button Operation

Two pushbutton switches are associated with the Model 371A's two talk channels. How they function will depend on the configuration of the unit. Each button can be configured independently.

Push to Talk

When a button has been configured for the *Push to Talk* mode how it functions is pretty self-explanatory. Press and hold the button when headset microphone audio is to be sent out the associated Dante audio output (transmitter) channel. The button's green LED will light to indicate that the output is active. The button will not be lit when the function is not active.

Latching

If a button has been configured for the *Latching* mode operation momentarily pressing ("tapping") the button will cause the function to change states; off-to-on or on-to-off. Whenever the talk function is active the green LED will light. The button will not be lit when the function is not active.

Push to Talk/Tap to Latch

If a button has been configured for the *Push to Talk/Tap to Latch* mode operation is a bit different and certainly more flexible. Press and hold the button to activate the talk function. When released the talk function will turn off. Momentarily pressing ("tapping") the button will cause the function to change states; off-to-on or on-to-off. Whenever the talk function is active the green LED will light. The button will not be lit when the function is not active.

Disabled

A button that is configured to the *Disabled* mode will never allow talk audio to be sent to the associated Dante audio output channel. But to provide "feedback" that the button has been pressed the button's green LED will flash rapidly three times. This says, in effect, "Yes, I recognize that you pressed the button but I'm not going to do anything in response!"

Sidetone Function

The Model 371A includes a sidetone function that sends microphone audio to the headphone output whenever one or both of the talk buttons are active. The quality of the sidetone audio should be excellent and will provide Model 371A users with a confidence signal that they are actively talking to other intercom users. The exact sidetone level is configured from among five values within the STcontroller application. There is also a selection choice for turning off sidetone. There is no means of adjusting the sidetone level using a physical button or control on the Model 371A unit. Also, the two headphone level controls on the top panel of the unit do not impact the sidetone level.

In most cases the exact sidetone level setting is not critical and typical users will not be concerned about revising it. But setting the sidetone level to something reasonable is important. Setting the level too low will encourage users to speak too loudly; setting it too high and users will be tempted to speak hesitantly. And while users aren't provided with a sidetone level control or other adjustment means, should the need arise STcontroller allows rapid adjustment of the sidetone level. In most cases the default sidetone level, *Medium*, should provide an appropriate level.

Call Function

An integrated call function allows Model 371A users, and users of compatible equipment, to signal to each other visually. Individual call functions are provided for each of the two talk channels. Either or both can have its call function active at any one time. Technically call is achieved by sending a 20 kHz audio tone on the desired audio output channel. This signal is summed (mixed) with normal talk audio. The Model 371A's two audio input channels continually monitor for the presence of 20 kHz. A call signal recognized when a continuous 20 kHz tone is detected on the audio input for that channel.

Normal talk audio signals will not be confused with a call signal. Digital filters within the Model 371A's processor integrated circuit limits the microphone input's high frequency response to about 10 kHz. This helps to ensure that false call detection won't take place. Digital filtering is also performed on the audio input (Dante receiver) signals before they are sent to the headphone output. This prevents users with extended high-frequency sensitivity, such as hosts and guests associated with dog shows, from receiving undesirable audio content.

To send a call signal is simple: just enable either or both of the talk buttons and simultaneously press and hold the call button. When call sending is active the orange LED associated with the call button will light. At the same time the channel or channels that are active in their talk mode will have their associated orange LED flash. Release the talk button and/or the call button and the sending of a call signal will cease. As expected, pressing the call button when neither talk channel is active will result in nothing occurring.

Whenever a Model 371A audio input channel receives a call signal (20 kHz audio tone) the orange LED on its companion talk button will first flash and then light continually. If a call signal is present on both audio input channels then the orange LEDs on both talk buttons will first flash then remain lit. When receiving a call signal on either or both audio input channels the orange LED associated with the call button will not light.

By using a 20 kHz tone for call signaling the Model 371A is compatible with legacy intercom equipment, including the venerable RTS BP-325. When interconnecting Model 371A and BP-325 units using an appropriate Dante-enabled interface, such as the Studio Technologies' Model 45DR Intercom Interface, call signaling is fully compatible. Compatibility with the Clear-Com method of call functionality is also possible by using the Model 45DC Intercom Interface. (It converts the DC call signal associated with pin 3 of a Clear-Com party-line circuit to a 20 kHz tone.) Also, devices such as the Studio Technologies Model 44D Audio Interface will send and receive 20 kHz signals that are be compatible with the Model 371A. The Model 44D converts its GPI (general-purpose input) signals to 20 kHz tones and sums them with the audio signals which are then transported "in band" via the Dante audio paths. Audio signals received by the Model 44D will be monitored for the presence of 20 kHz call signals. If call signals are detected the GPO (general-purpose output) signals will activate.

How to Identify a Specific Model 371A

The Dante Controller software application offers an Identify command that can be used to help locate a specific Model 371A. When *Identify* is selected it will send a command to a single Model 371A unit. On that specific unit the three pushbutton LEDs (talk 1, talk 2, and call) will light in a unique pattern. In addition, the SYS and SYNC status LEDs, located directly below the etherCON RJ45 receptacle on the bottom panel, will slowly flash green. After a few seconds the LED identification pattern will cease and normal Model 371A button LED and Dante status LED operation will resume.

If configured, the *Identify* command can also cause the mic kill function to activate. For details please refer to the next paragraph of this guide.

Mic Kill Function

The Model 371A includes a mic kill function, allowing talk buttons that have been placed in their enabled (on) state to be remotely changed to their disabled (off) state. This allows one or both talk channels on a specific Model 371A that have been enabled to be remotely disabled. The reason for this function is simple. It's common in intercom applications for users to enable a talk channel and then go "off headset," forgetting that they've left that talk channel enabled. While they take a break or go to lunch, all other users are forced to listen to that channel which may include unwanted audio. Having the ability to disable this talk channel can be very useful.

A configurable choice in the STcontroller software application is used to select which of the mic kill methods, *Device ID* or *Global Kill Command*, will be recognized. Either or both can be enabled as desired. It's also possible to disable the mic kill function.

If the *Device ID* method of activating mic kill is enabled, whenever a specific Model 371A is identified using the Identify command within Dante Controller or STcontroller any enabled talk channel will be forced to its disabled state. If the *Global Kill Command* method is enabled the Model 371A will respond to any such request and force all talk channels to their disabled state.

Technical Notes

IP Address Assignment

By default the Model 371A's Ethernet interface will attempt to automatically obtain an IP address and associated settings using the 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. 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 cable will, in most cases, correctly acquire IP addresses and be able to communicate with each other.

An exception does arise when trying to directly interconnect two Dante-enabled devices that use Ultimo integrated circuits to implement Dante. The Model 371A uses the Ultimo "chip" and, as such, a direct one-to-one interconnection to another Model 371A (or other Ultimo-based product) is not supported. An Ethernet switch linking the two units is required to successfully interconnect two Ultimo-based devices. The technical reason that a switch is required relates to the need for a slight latency (delay) in the data flow. And an Ethernet switch will provide this. While this is certainly an anomaly, since PoE power is required for Model 371A operation it's highly unlikely that an application would use two Model 371A units without a PoE-enabled Ethernet switch being present.

Using the Dante Controller software application the Model 371A's IP address and related network parameters can be manually set for a fixed (static) configuration. While this is a more-involved process than simply letting DHCP or link-local automatically assign an address, if fixed addressing is necessary then this capability is available. But in this case it's highly recommended that each unit be physically marked, e.g., directly using a permanent marker or "console tape," with its specific static IP address. If knowledge of a Model 371A's IP address has been misplaced there is no reset button or other method to easily restore the unit to a default IP setting.

In the unfortunate event that a device's IP address is "lost," 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. The simplest means of identifying an unknown IP address is to create a "mini" LAN with a small PoE-enabled Ethernet switch connecting a personal computer to the Model 371A. Then by using the appropriate ARP command the required "clues" can be obtained.

Optimizing Network Performance

For best Dante audio-over-Ethernet performance a network that supports VoIP Quality-of-service (QoS) capability is recommended. This can 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 (www.audinate.com) for details on optimizing networks for Dante applications. Also, be certain to disable Energy-Efficient Ethernet (EEE) support on all Ethernet switch ports that are associated with Dante devices. Some implementations of EEE can incorrectly interpret that a connected device is not present and prevent proper Dante operation.

Application Firmware Version Display

There are two ways in which the version number of the Model 371A's application firmware (embedded software) can be identified. One requires only the Model 371A unit and involves a button-press sequence performed upon power up. The other method utilizes the Model 371A and the STcontroller software application. Either method may prove to be useful when working with factory personnel on application support and troubleshooting.

As part of the Model 371A's power-up sequence the unit's application firmware (embedded software) version number can be displayed. Before connecting the PoE-enabled Ethernet cable, press and hold the call button. Then connect the Ethernet cable. Upon application of PoE power the Model 371A will go through its normal power-up sequences followed by a display of the firmware version. The LED associated with the channel 1 talk button will "flash" to display the major version number. Then the LED associated with the channel 2 talk button will "flash" to display the minor version number. Once the version number has been displayed the call button can be released and normal operation will begin. As an example of what would be a typical application firmware display, if the channel 1 talk button "flashes" once followed by the channel 2 talk button "flashing" two times this would indicate that application firmware version 1.2 was present in the Model 371A.

A selection in the STcontroller software application allows the Model 371A's application firmware version to be identified. Connect the Model 371A unit to the network and let it connect and start to function. Then, after starting STcontroller, review the list of identified devices and select the specific Model 371A that interests you. Then select **Version** under the **Device** tab. A page will then display that will provide lots of useful information. This includes the application firmware version and well as details on the firmware present in the Ultimo integrated circuit.

Application Firmware Update Procedure

It's possible that updated versions of the application firmware (embedded software) that is utilized by the Model 371A's processor (microcontroller or MCU) integrated circuit will be released to add features or to correct issues. Refer to the Studio Technologies website for the latest application firmware file. The unit has the ability to load a revised file into the MCU's non-volatile flash memory by way of a USB interface. The Model 371A implements a USB host function that directly supports connection of a USB flash drive. The Model 371A's MCU updates its application firmware using a file named **M371A.bin**.

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. Save the new application firmware file in the root directory with a name of **M371A.bin**. Studio Technologies will supply the application firmware file inside a .zip archive file. While the application firmware file inside of the zip file will adhere to the naming convention required by the Model 371A, the name of the zip file itself will include the file's version number. For example, a file named **M371Av1r2MCU.zip** would indicate that version 1.2 of the application firmware (**M371A.bin**) is contained within this zip file.

Once the USB flash drive is inserted into the USB interface, located on the main circuit board under the cover, the unit must be powered off and again powered on. At this point the file will automatically be loaded into the processor's flash memory. The precise steps required will be highlighted in the next paragraphs of this guide.

To update the application firmware file follow these steps:

1. Disconnect power from the Model 371A. This will entail removing the Ethernet connection that is providing PoE power.
2. Remove the cover from the Model 371A. Begin by removing the four Phillips-head machine screws (#1 screw driver tip), two per side. Be certain to save the screws so that re-assembly will be fast and painless. Then carefully slide the cover forward to separate it from the level controls and buttons and then lift it off.

3. Locate the USB connector on the main circuit board. It's near the call button. Insert the prepared USB flash drive into it.
4. Apply power to the Model 371A by connecting to a Power-over-Ethernet (PoE) Ethernet source.
5. After a few seconds the Model 371A will run a "boot loader" program that will automatically load and save the new application firmware file (**M371A.bin**). This will take only a few seconds. During this time period the call button's LED will flash slowly in alternate colors. Once the entire process is over, taking approximately 10 seconds, the Model 371A will restart using the newly-saved application firmware.
6. At this time the Model 371A is operating under the newly-saved application firmware and the USB flash drive can be removed. But to be conservative, remove PoE power first and then remove the USB flash drive.
7. Confirm that the desired application firmware version has been correctly saved. This can be done by pressing and holding the call button, applying PoE power to the Model 371A, and then "reading" the application firmware version number by first observing the channel 1 talk button's LED and then observing the channel 2 talk button's LED. Alternately, the STcontroller software application can be used to identify the application firmware version number. Whatever method you use, ensure that the desired version is present.
8. Once the update process has been completed reverse the steps and reattached the cover using the four machine screws.

Note that upon power being applied to the Model 371A if a connected USB flash drive doesn't have the correct file (**M371A.bin**) in the root folder no harm will occur. If the correct file is not present upon power up the call button's LED will flash on and off rapidly for a few seconds to indicate this error condition and then normal operation using the unit's existing application firmware will begin.

Ultimo Firmware Update

As previously discussed in this guide, the Model 371A implements Dante connectivity using the 2-input/2-output Ultimo integrated circuit from Audinate. The Dante Controller software application can be used to determine the version of the firmware (embedded software) that resides in the Ultimo "chip." The STcontroller software application can also be used to identify the Ultimo's firmware version. (Use the **Version** selection under the **Device** tab.) The Ultimo firmware can be updated by way of the Model 371A's Ethernet connection. The latest Ultimo firmware file is available on the Studio Technologies website. The Dante Firmware Update Manager (FUM) application program is used to install the Ultimo firmware. This program is also available for download on the Studio Technologies website. It's anticipated that an updated version of the Dante Controller software application will include an automated method of updating Ultimo firmware. It should offer a much simpler method of keeping the Ultimo firmware current.

Model 371A versus Model 371

Some users may be interested in the differences between the original Model 371 Intercom Beltpack and the newer Model 371A that is described in this guide. A comparison chart is available on the Studio Technologies website. In addition, the following text provides a summary of the changes:

- All configuration selections for the Model 371A are performed using the STcontroller software application. STcontroller also allows the version numbers of the Model 371A's application firmware (embedded software) and Ultimo firmware to be remotely identified.
- Support for electret microphones was added to the Model 371A. This involved adding a 5 volt DC power source to the microphone input circuitry. This power source can be enabled or disabled as required using STcontroller.
- The configurable microphone input gain settings were modified to be compatible with the increased output level supplied by electret microphones.
- The DIP switch assembly on the Model 371 is not present on the Model 371A. As such the belt clip no longer needs to be rotated and its mounting arrangement has been changed.
- The user-accessible sidetone trim potentiometer found on the Model 371 is not present on the Model 371A. In the Model 371A the level of the sidetone function is configured using STcontroller.
- The Model 371A offers a second method of activating the mic kill function. This allows a data signal, in additional to the *Device ID* command, to activate mic kill.
- The STcontroller software application now offers the ability to configure the mic kill function, including allowing it to be disabled.

Belt Clip

The belt clip is attached to the back of the Model 371A's enclosure by way of two 6-32 thread, Philips-head machine screws. Unlike other the original Model 371 Intercom Beltpack units, the belt clip on the Model 371A will never have to be rotated as part of the configuration process. (There are no configuration DIP switches under the belt clip.) But if the belt clip does need to be replaced a #1 Philips-head screw driver would be the correct tool to remove the machine screws.

Restoring Factory Defaults

A command in the STcontroller software application allows the Model 371A's configuration to be reset to the factory default values. From STcontroller select the Model 371A 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. The values shown on the screen should reflect the default values. They are:

Microphone Input Type: Dynamic

Microphone Preamplifier Gain:
Low (42 dB)

Sidetone Level: Medium

System Mic Kill: Device ID/Global Kill Command

Button Operation – Channels 1 and 2:
Push to Talk/Tap to Latch

Specifications

Power Source:

Power-over-Ethernet (PoE): class 1 (very low power, ≤3.84 watts) per IEEE® 802.3af

Network Audio Technology:

Type: Dante audio-over-Ethernet

AES67-2013 Support: yes

Dante Domain Manager (DDM) Support: yes

Bit Depth: up to 24

Sample Rate: 48 kHz

Number of Transmitter (Output) Channels: 2

Number of Receiver (Input) Channels: 2

Dante Audio Flows: 4; 2 transmitter, 2 receiver

Network Interface:

Type: 100BASE-TX, twisted-pair Ethernet, Power-over-Ethernet (PoE) supported

Data Rate: 100 Mb/s (10 Mb/s and 1000 Mb/s "GigE" Ethernet not supported)

Audio Channels: 2 talk, 2 listen

Microphone Input:

Compatibility: dynamic or electret (low-voltage DC-powered) microphones

Type: unbalanced

Electret Microphone Power: 5 volts DC via 2.21 k resistor, selectable on/off

Gain, configured for dynamic microphones: 42 or 48 dB, selectable, ref. –60 dBu input to Dante output (–20 dBFS nominal)

Gain, configured for electret microphones: 22 or 30 dB, selectable, ref. –60 dBu input to Dante output (–20 dBFS nominal)

Frequency Response: 50 Hz to 10 kHz, –3 dB

Distortion (THD+N): <0.02%

Dynamic Range: 75 dB

Compressor:

Threshold: 1 dB above nominal level (–19 dBFS)

Slope: 2:1

Status LED: compressor active

Headphone Output:

Type: single-channel (sum (combination) of audio input 1 and audio input 2)

Compatibility: intended for connection headsets with nominal impedance of 50 ohms or greater

Maximum Output Voltage: 3.8 Vrms, 1 kHz, 150 ohm load

Frequency Response: 20 Hz to 10 kHz, –3 dB

Distortion (THD+N): <0.002%

Dynamic Range: >100 dB

Call Function:

Signaling Method: 20 kHz, ±800 Hz, within audio channels

Call Send Level: –20 dBFS

Call Receive Level: –27 dBFS minimum

Connectors:

Headset: 4-pin male XLR (Associated headset requires cable-mounted 4-pin female XLR connector wired with pin 1 mic –/shield/screen; pin 2 mic +; pin 3 headphone –; and pin 4 headphone +)

Ethernet: Neutrik NE8FBH etherCON RJ45 receptacle

USB: type A receptacle (located inside l 371A's enclosure and used only for application firmware updates)

Configuration: requires Studio Technologies STcontroller software application, version 1.06.00 and later (STcontroller is compatible with Windows version 7 and later)

Environmental:

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

Storage Temperature: –40 to 70 degrees C (–40 to 158 degrees F)

Humidity: 0 to 95%, non-condensing

Altitude: not characterized

Dimensions (Overall):

3.6 inches wide (9.2 cm)
1.6 inches high (4.0 cm)
4.8 inches deep (12.6 cm)

Mounting: intended for portable applications; contains integral belt clip; optional MBK-01 Mounting Bracket Kit allows Model 371A to be permanently mounted

Weight: 0.6 pounds (0.3 kg)

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