Model 372A Intercom Beltpack

User Guide

Issue 1, April 2019

This User Guide is applicable for serial numbers
M372A-00151 and later with application firmware 1.1 and later

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Revision History

Issue 1, April 2019:

- Initial release.

Introduction

The Model 372A Intercom Beltpack is a highly compact user-worn device that combines a single channel of talk and two channels of listen. The unit begins with the features offered by traditional analog party-line (PL) intercom user devices and adds 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 372A and other compatible Studio Technologies' beltpack units can be used to create party-line intercom applications with help from an external Dante-enabled audio matrix such as the Studio Technologies' Model 5422 Dante Intercom Audio Engine. Alternately, Model 372A units can be used "point-to-point" or interfaced with Dante-compatible matrix intercom systems.

Having one talk and two listen channels may seem unconventional. But it can be ideal for many "real-world" applications. Often an intercom user is primarily listening and non-verbally responding to requests made by producers, directors, or stage managers. Typically, the Model 372A will be configured to be part of one talk-and-listen party-line intercom channel. During the time that an event is taking place the listen function will serve a much more important role; the talk function will rarely be utilized. However, the second listen channel will often be important. Typically, it will be designated as a program-listen or "show audio" channel. The two listen channels, along with the ability to receive and display call signals, allow the Model 372A to very effectively support production personnel in a compact and cost-effective manner.

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, talk button operation, and headphone channel assignment. Features include integrated sidetone, call signal receive display, and remote mic kill ("talk off"). The range of capabilities, along with the excellent audio quality provided by the digital audio signal path, offers a unique and powerful user experience.

Setting up and configuring a Model 372A 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. The Model 372A is compatible with both broadcast and "gaming" headsets.

Figure 1. Model 372A Intercom Beltpack top and bottom views

A broadcast or intercom-style headset with a dynamic or electret (DC-powered) microphone can be interfaced with the Model 372A using a 5-pin XLR connector. The Model 372A also directly supports connection of earbuds or gaming headsets that utilize a 3.5 mm 4-conductor TRRS plug. These moderately priced devices, commonly associated with mobile phones or personal computers, are often of high-quality and may be preferred for some applications. With the Model 372A's moderate price and ability to support a broad range of headset devices the overall cost of deploying an intercom system can often meet budget goals.

The STcontroller software application is used to select the unit's operating parameters. The talk pushbutton switch can be configured for optimal operation. Two "push-in/push-out" ("pop-out") rotary controls make it easy to set and maintain the desired headphone output level. The Model 372A's compact 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 372A 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 education and commercial theater, sports and entertainment TV and radio events, streaming broadcasts, corporate and government AV, post production, and aerospace.

Dante Audio-over-Ethernet

Audio data is sent to and received from the Model 372A using the Dante audio-over-Ethernet media networking technology. As a Dante-compliant device, the Model 372A's one audio output (Dante transmitter) channel 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 372A is AES67 compatible and compliant with the Dante Domain Manager™ software application.

Two bi-color LEDs provide status indications of the Dante interface. The Dante Identify command takes on a unique role with the Model 372A. Not only will it cause the talk pushbutton's orange LED to light in a highly visible sequence, it will also turn off ("kill") the talk function if it is active.

Audio Quality

The Model 372A offers "pro" audio performance that is not found in typical party-line (PL) intercom beltpacks. 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 as required. 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 372A's processor where channel routing, headphone level control, and sidetone creation are performed within the digital domain. This provides flexibility, allowing precise control of the audio signals and eliminating the need for the two rotary level controls from having to directly handle analog audio signals. The audio signals destined for the 2-channel headphone output are sent to a high-performance digital-to-analog converter and then on to robust driver circuitry. High signal levels can be provided to a variety of headsets.

Call Function Receive

A call receive function allows Model 372A users to be provided with a visual indication that a call signal is active. The orange LED within the pushbutton switch will first flash then light solid whenever a call signal is detected on either of the Dante receiver (input) channels. Using 20 kHz tones, the call signals are sent within the Dante audio channels ("in band") allowing interoperability between multiple Studio Technologies' beltpack units as well as being compatible with legacy party-line intercom systems. Call signals 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 372A 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 372A by way of an Ethernet network connection. Configurable parameters include microphone power and preamplifier gain, headphone monitoring, sidetone audio operation, and talk button operation.

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

To support optimum user performance, the Model 372A's talk pushbutton switch can be configured from among three choices: Push to Talk, Latching, or Push to Talk/Tap to Latch. Two audio channels arrive via Dante receivers (inputs) and are destined for the 2-channel headphone output. Each input source can be independently routed to the left headphone channel, right headphone channel, or both the left and right headphone channels. This flexibility allows a variety of listening environments to be created, including stereo, single-channel monaural, and dual-channel monaural.

Ethernet Data and PoE

The Model 372A 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 372A'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 372A'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 372A 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 372A 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 372A 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 may be made using the three connectors located on the bottom of the Model 372A'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 dual-channel or single-channel (dual- or single-ear) headset will be connected using a cable-mounted 5-pin male XLR connector. Alternately, a headset with a 3.5 mm 4-conductor TRRS plug can be used. (The headset's plug needs to follow the CTIA™/AHJ configuration standard.)

Ethernet Connection with PoE

A 100BASE-TX Ethernet connection that supports Power-over-Ethernet (PoE) is required for Model 372A operation. This one connection will provide both the Ethernet data interface and power for the Model 372A'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 372A 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 372A's enclosure. This allows connection by way of a cable-mounted etherCON connector or a standard RJ45 plug. The Model 372A's Ethernet interface supports auto MDI/MDI-X so that a crossover cable is not required.

Headset Connections

The Model 372A allows two different types of headsets to be connected. A 5-pin female XLR connector is provided to support connection of a standard broadcast- or intercom-style communications headset. A 3.5 mm 4-conductor TRRS jack allows a computer gaming headset to be directly connected. The Model 372A's two headset connectors are electrically wired in parallel. As such, only one type of headset should be connected at one time.

Headset A

The Model 372A provides a 5-pin female XLR connector that interfaces with the microphone and headphone connections of a single- or dual-ear intercom- or broadcast-style headset. The connector is labeled Headset A. 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 372A's mic in –/shield connection. No support is provided for microphones that require P12 or P48 phantom power.

To allow users of stereo (dual-earpiece or "double muff") headsets to hear a monaural version of the two headphone output channels does not require special wiring of the 5-pin male XLR mating connector. The headset's left headphone channel should always be wired to pin 4 and the right headphone channel to pin 5. Configuration choices, discussed later in this guide, can then be used to create the desired monaural output. It's important not to connect together (short) pins 4 and 5 of the headset's connector as damage to the Model 372A's output circuitry could result.

A monaural (single-earpiece or "single muff") headset should be wired such that its headphone is wired only to pin 4; pin 5 should be remain unused. Configuration choices, discussed later in this guide, can be used to create a monaural output.

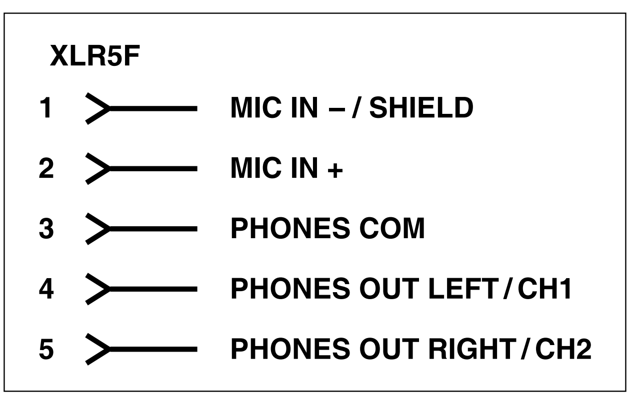

Figure 2. Headset A connection pinout chart

It's possible that some Beyerdynamic headset interconnecting cable assemblies terminate the earpiece's left and right connections opposite from what the Model 372A and other broadcast equipment require. These cables may terminate the left earpiece to pin 5 of the 5-pin male XLR connector and the right earpiece to pin 4. If this condition is present it will require reversing or "flipping" the two wires in a headset's connector such that the left earpiece connects to pin 4 and the right earpiece to pin 5.

If a separate microphone and pair of headphones need to connected an adapter cable assembly should be fabricated. It would consist of a 5-pin male XLR connector wired to both a 3-pin female XLR connector for the microphone and a ¼-inch or 3.5 mm TRS jack for the headphones. In this scenario the microphone would in most cases be a dynamic type as the Model 372A provides only low-voltage DC "electret" power. Phantom-powered (P12 or P48) microphones would not be compatible. Most hand-held or "stick" microphones are dynamic and should function correctly.

Headset B

The Model 372A also allows direct connection of gaming headsets that are ubiquitous in the personal computer world. The 3.5 mm 4-conductor TRRS jack, labeled Headset B, is compatible with the CTIA™/AHJ configuration standard which has the headphone left channel on the tip connection, the headphone right channel on the ring 1 connection, common on the ring 2 connection, and the microphone on the sleeve connection. Compatible headsets are readily available, typically described as personal computer or gaming headsets. The microphones used in these headsets are electret type which require a low-voltage DC source for operation. The Model 372A is able to provide this power and requires only that the appropriate setting in STcontroller be selected. Refer to Figure 3 for a detailed description of the compatible TRRS plug.

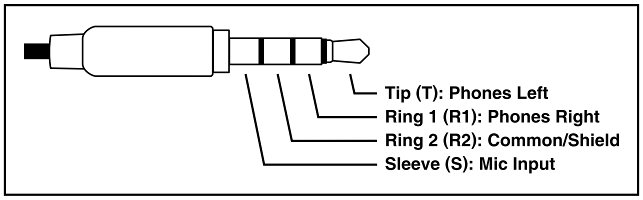

Figure 3. Headset B connection pinout chart

Dante Configuration

For audio to pass to and from the Model 372A requires that several Dante-related parameters be configured. These configuration settings will be stored in non-volatile memory within the Model 372A'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 372A uses the Ultimo 2-input/2-output integrated circuit to implement the Dante architecture. One transmitter (output) channel and both receiver (input) channels are utilized.

The Dante transmitter (output) channel associated with the Model 372A's Dante interface must be assigned to the desired Dante receiver (input) channel. This achieves routing the Model 372A's one talk output audio channel to the device (or devices) that will be "listening" to it. Within Dante Controller a "subscription" is the term used for routing a transmitter (Tx) flow (a group of output channels) to a receiver (Rx) 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 372A's transmitter channel needs 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 (input) channels associated with the Model 372A's audio inputs also need to be routed to the Dante transmitter (output) channels provided by the desired source device. These two audio signals can be sent to the Model 372A's 2-channel headphone output.

The Model 372A supports an audio sample rate of 48 kHz with no pull-up/pull-down values available. The Model 372A 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 372A has a default Dante device name of **ST-M372A-** followed by a unique suffix. The suffix identifies the specific Model 372A 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 one Dante transmitter (Tx) channel has a default name of **Ch1**. 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 372A Configuration

Many of the Model 372A'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-application/) and is compatible with personal computers running Windows operating systems that are version 7 and later. STcontroller version 2.01.00 or later is required to support the Model 372A. 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 372A unit or units that are to be configured.

Parameters

STcontroller allows observation and selection of functions that include:

- Microphone input electret power on/off
- Microphone input gain
- Headphone input-to-output channel routing
- Sidetone level
- Talk button operation

Changes made using STcontroller will be immediately reflected in the unit's operation; no Model 372A "reboot" is required. Each time a change is made the talk button, located on the top panel, will momentarily flash orange 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. All Model 372A units that are present on the network will be recognized and display on the device list. Use the Identify command to allow easy recognition of a specific Model 372A 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 – Electret Power

Choices are *Enabled* or *Disabled*.

If the headset has an electret microphone that requires a source of low-voltage DC power for operation enable the Electret Power check box. In virtually all cases a gaming or computer headset that uses a 3.5 mm TRRS plug will require microphone power. If the associated headset has a dynamic (non-powered) microphone do not enable the Electret Power check box. Most broadcast headsets that terminate on 5-pin male XLR connectors will not require microphone power. The on/off status is displayed by way of a red LED, labeled MIC POWER, that is located adjacent to the Headset A connector.

Note that the Model 372A 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 Input – Gain

Choices are *24 dB*, *30 dB*, *36 dB*, *42 dB*, and *48 dB*.

Dynamic microphones have an output level that is typically lower than that provided by electret microphones. As such, the 36, 42, or 48 dB gain settings will probably be appropriate for them. Electret microphones often have internal circuitry which provides a relatively high output level. The 24 and 30 dB gain settings will typically be appropriate for use with this type of microphone.

When a dynamic microphone is connected to the Model 372A the 42 dB gain setting should be appropriate for many applications. The 36 dB choice may be correct 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 48 dB may be helpful in some applications, such as with headsets that have a low microphone output level. The 48 dB gain setting may also be useful when the Model 372A 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 where “whispering” may be necessary.

Electret microphones typically have a higher output level due to their internal preamplifier circuitry. As such, less preamplifier gain may be required. Selecting the 24 or 30 dB gain setting will probably be appropriate in these cases.

The compressor active LED, labeled COMP and visible adjacent to the HEADSET A connector, can act as a guide when setting the microphone preamplifier 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 42 dB it might be a good idea to change it to 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 30 dB, changing it to 24 dB might be warranted.

Headphone Output Channel Routing

Choices are *Left*, *Right*, and *Left and Right*.

Two Dante transmitter (output) channels can be assigned to the Model 372A’s two Dante receiver (input) channels using the Dante Controller application. How these two audio signals are routed to the Model 372A’s headphone output channels can be independently configured. For flexibility, each input source has three routing options. They can be independently configured for sending only to the left channel, only to the right channel, or sending to both the left and right headphone output channels.

In some applications split-mono or stereo operation will be desired. In this case input channel 1 might be routed to the left headphone output while input channel 2 might be routed to the right headphone output. When using a stereo or dual-ear headset sending both inputs to both headphone output channels is often referred to as a dual-channel mono output. If a monaural (“single muff”) headset is used the Left configuration option allows the two input channels to be combined to monaural and sent to just the left headphone output channel.

Sidetone – Level

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

The Model 372A includes a sidetone function that allows microphone audio coming from the microphone input section to be sent to the headphone output. This serves as a user confirmation that they are actively sending audio to the talk output channel. Sidetone audio will only be routed to the headphone output when the talk channel 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.

Button Operation – Talk

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

This configuration section allows selection of how the talk button will function. When the Push to Talk mode is selected the operation is self-explanatory. Only when the talk button is pressed and held will talk audio be sent out the 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 the Dante transmitter channel. When the button is released audio will stop being sent out the 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 mode or Push to Talk/Tap to Latch mode settings will prove to be very useful.

Operation

At this point everything should be ready and Model 372A operation can commence. An Ethernet connection with Power-over-Ethernet (PoE) capability should have been made. A headset terminated with a cable-mounted 5-pin male XLR connector may be plugged into the Headset A connector. Alternately, a gaming or computer-type headset will be connected to Headset B using a 3.5 mm 4-conductor TRRS plug.

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 372A's Dante configuration settings should have been selected using the Dante Controller software application. In this way, the Model 372A's audio output channel (Dante transmitter channel) 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 372A 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 compressor active LED might somewhat randomly flash for a brief period. The green and orange LEDs that illuminate the talk pushbutton switch will light as part of a short test sequence to indicate that the application firmware (embedded software) has started. The mic power LED will also light for a short interval to indicate that it is functioning. Once the test 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 372A'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 372A 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 372A is not synchronized with a Dante network. It will light solid green when the Model 372A 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 372A 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 A connector. Labeled COMP, the LED displays the status of the microphone preamplifier 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 connected 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 the talk channel is active.

Headphone Output

Two rotary potentiometers ("pots"), located on the Model 372A's top panel, allow individual adjustment of the level of the two audio input signals as they are sent to the 2-channel headphone output. Depending on the configuration of the unit, each audio input can be sent to the left channel, the right channel, or both the left and right channels of the headphone output. 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 372A'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 talk channel does not impact the headphone output.

Button Operation

A pushbutton switch is associated with the Model 372A's talk channel. How the switch functions will depend on the configuration of the unit.

Push to Talk

When the 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 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 the 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 the 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 pushbutton will light green. The button will not be lit when the function is not active.

Sidetone Function

The Model 372A includes a sidetone function that sends microphone audio to the headphone output whenever the talk function is active. The quality of the sidetone audio should be excellent and will provide Model 372A users with a confidence signal that they are actively talking to other intercom users. Sidetone audio will always be sent to both the left and right headphone output channels. This is because the function is trying to simulate what a user would hear if they didn't have a headset covering both their ears. An exception is if both Dante input channels are routed to only the left headphone output channel. In this case sidetone audio will only be sent to the left headphone output channel.

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 372A 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 typically 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 Detection Display

A Model 372A function allows a call signal that's present on either Dante receiver (input) channel to cause the pushbutton switch's LED to light. Whenever a call signal is detected the button's orange LED will first flash and then light continually. Technically, a call signal is achieved by sending a 20 kHz audio tone on the desired Dante audio channel. This signal is summed (mixed) with normal talk audio. The Model 372A's two audio input channels continually monitor the audio input for the presence of 20 kHz.

The detection circuitry will not confuse normal talk audio signals with a call signal.

Digital filters within the Model 372A's processor integrated circuit help to ensure that false call detection won't take place. Digital filtering is also performed on the microphone audio output signals as well as the audio input (Dante receiver) signals before they are sent to the headphone outputs. This prevents users with extended high-frequency sensitivity, such as hosts and guests associated with dog shows, from receiving undesirable audio content.

By using a 20 kHz tone for call signaling the Model 372A is compatible with legacy intercom equipment, including the venerable RTS BP-325 beltpack. When interconnecting Model 372A 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 372A. 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.

How to Identify a Specific Model 372A

Both the Dante Controller and STcontroller software applications offer a command that can be used to help locate a specific Model 372A. When the Identify command is selected it will send a message to a single Model 372A unit. On that specific unit the pushbutton switch's orange LED will flash rapidly. 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 372A button LED and Dante status LED operation will resume.

Mic Kill Support

The Model 372A includes a mic kill function, allowing the talk button, if it has been placed in its enabled (on) state, to be remotely forced to its disabled (off) state. This function is sometimes referred to as a "talk off" function. Two actions can enable the mic kill function. Any time a Model 372A receives a Dante Identify command it will cause both an LED pattern to start as well as enabling the mic kill function. The second "trigger" method uses the Studio Technologies' Global Mic Kill command. This command can be activated from a menu choice in the STcontroller software application.

The reason for the Model 372A to offer a mic kill function is simple. It's common in intercom applications for users to enable ("latch on") 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. This makes having the ability to disable a talk channel very useful. The Dante Identify command allows a "latched on" talk channel on a specific Model 372A to be remotely disabled. Alternately, by using the Global Mic Kill command a large group of units can simultaneously have their talk channels disabled. This would include any Studio Technologies' compatible device that follows this protocol.

Technical Notes

IP Address Assignment

By default, the Model 372A'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 372A uses Ultimo and a design limitation in it prevents a one-to-one interconnection with another Model 372A (or any other Ultimo-based product). An Ethernet switch linking the two Ultimo-based units is required to successfully interconnect them. The technical reason that a switch is required relates to the need for a slight latency (delay) in the data flow; an Ethernet switch will provide this. While this is certainly an anomaly, since PoE power is required for Model 372A operation it's highly unlikely that an application would use two Model 372A units without a PoE-enabled Ethernet switch being present.

Using the Dante Controller software application, the Model 372A'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 IP address, if fixed IP 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 372A'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 372A. 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 Dante device is not present and prevent proper operation.

Application Firmware Version Display

There are two ways in which the version number of the Model 372A's application firmware (embedded software) can be identified. One requires only the Model 372A unit and involves a button-press sequence performed upon power up. The other method utilizes the Model 372A 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 372A'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 talk pushbutton. Then connect the Ethernet cable. Upon application of PoE power the Model 372A will go through its normal power-up sequences followed by a display of the firmware version. The LED associated with the talk button will "flash" green to display the major version number. Then, after a short pause, the talk button's LED will "flash" orange to display the minor version number. Once the version number has been displayed the talk button can be released and normal operation will begin. As an example of what would be a typical application firmware display, if the LED in the talk button first "flashes" green once then "flashes" orange twice this would indicate that application firmware version 1.2 was present in the Model 372A.

A selection in the STcontroller software application allows the Model 372A's application firmware version to be identified. Connect the Model 372A 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 372A that interests you. Then select **Version** under the **Device** tab. A page will then display that will provide a lot 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 372A'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 its MCU's non-volatile flash memory by way of a USB interface. The Model 372A implements a USB host function that directly supports connection of a USB flash drive. The Model 372A's MCU updates its application firmware using a file named **M372A.bin**. (Note for geeks: the suffix .bin indicates that it's a binary file.)

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 **M372A.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 372A, the name of the zip file itself will include the file's version number. For example, a file named **M372Av1r2MCU.zip** would indicate that version 1.2 of the application firmware (**M372A.bin**) is contained within this zip file.

Once a correctly prepared USB flash drive is inserted into the USB interface, located under the cover of the Model 372A's main circuit board, 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 following paragraphs.

To update the application firmware file follow these steps:

1. Disconnect power from the Model 372A. This will entail removing the Ethernet connection that is providing PoE power.
2. Remove the cover from the Model 372A. 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 pushbutton and then lift it off.
3. Locate the USB connector on the main circuit board. It's near the front, adjacent to the pushbutton and level controls. Insert the prepared USB flash drive into the USB connector.
4. Apply power to the Model 372A by connecting to a Power-over-Ethernet (PoE) Ethernet source.
5. After a few seconds the Model 372A will run a "boot loader" program that will automatically load and save the new application firmware file (**M372A.bin**). This will take only a few seconds. During this time period the talk pushbutton's LED will flash slowly in alternate colors. Once the entire process is over, taking approximately 10 seconds, the Model 372A will restart using the newly saved application firmware.
6. At this time the Model 372A 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 talk pushbutton, applying PoE power to the Model 372A, and then "reading" the application firmware version number by observing the talk pushbutton'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 372A, if a connected USB flash drive doesn't have the correct file (**M372A.bin**) in the root folder no harm will occur. If the correct file is not present upon power up the talk pushbutton's LED will flash rapidly, alternating green and orange, for a few seconds to indicate this error condition after which normal operation using the unit's existing application firmware will begin.

Ultimo Firmware Update

As previously discussed in this guide, the Model 372A 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 Ultimo's firmware version. (Use the **Version** selection under the **Device** tab.) The Ultimo firmware can be updated by way of the Model 372A's Ethernet connection. The latest Ultimo firmware file is available on the Studio Technologies' website. The Dante Firmware Update Manager (FUM) application program has traditionally been used to install the Ultimo firmware. The Dante Controller software application also includes an automated method of updating Ultimo firmware. Both applications are available, free of charge, on the Audinate website (www.audiante.com).

Restoring Factory Defaults

A command in the STcontroller software application allows the Model 372A's configuration to be reset to the factory default values. From STcontroller select the Model 372A 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 Model 372A's factory default values.

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: 1

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)

Compatibility – Headset A: single- or dual-ear broadcast-style with dynamic or electret (low-voltage DC-powered) microphone: pin 1 mic common; pin 2 mic; pin 3 phones common; pin 4 phones left; pin 5 phones right

Compatibility – Headset B: CTIA™/AHJ configuration (typically uses electret powered mic): tip phones left; ring 1 phones right; ring 2 common; sleeve mic

Audio Channels: 1 talk, 2 listen

Microphone Input:

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

Type: unbalanced

Electret Microphone Power: 3.3 volts DC via 2.00 k resistor, selectable on/off

Impedance: 1 k ohms, nominal, microphone power off; 690 ohms, nominal, microphone power on

Gain: 24, 30, 36, 42, or 48 dB, selectable, ref. –60 dBu input to Dante output (–20 dBFS nominal)

Frequency Response: 40 Hz to 20 kHz, –3 dB

Distortion (THD+N): <0.02% (at minimum gain)

Dynamic Range: 91 dB of dynamic range

Compressor:

Application: applies to Dante transmitter (output) channel and sidetone audio

Threshold: 2 dB above nominal level (–19 dBFS)

Slope: 2:1

Status LED: compressor active

Headphone Output:

Type: 2-channel

Compatibility: intended for connection to stereo (dual-channel) or monaural (single-channel) headsets with nominal impedance of 50 ohms or greater

Maximum Output Voltage: 2.8 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 Receive Function:

Implementation: monitors both Dante receiver (input) channels for presence of call signals

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

Call Receive Level: –27 dBFS minimum

Connectors:

Headset A: 5-pin female XLR

Headset B: 4-conductor (TRRS) 3.5 mm jack, per Japanese standard JEITA/EIAJ RC-5325A

Ethernet: Neutrik NE8FBH etherCON RJ45 receptacle

USB: type A receptacle (located inside Model 372A's enclosure and used only for application firmware updates)

Configuration: requires Studio Technologies' STcontroller software application, version 2.01.00 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):

3.1 inches wide (7.9 cm)

1.5 inches high (4.0 cm) without belt clip;
1.8 inches (4.6 cm) with belt clip

4.9 inches deep (12.5 cm)

Deployment: intended for portable applications; contains integral belt clip

Weight: 0.5 pounds (0.23 kg)

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

Appendix A

STcontroller default Model 372A configuration values:

Microphone Input – Electret Power: Off

Microphone Input – Gain: 36 dB

Headphone Output – Channel 1 Input Routes to: Left

Headphone Output – Channel 2 Input Routes to: Right

Headphone Output – Sidetone Level: Medium

Button Operation – Talk: Push to Talk/Tap to Latch