Model 371 Intercom Beltpack

User Guide

Issue 1, January 2017

This User Guide is applicable for serial numbers
M371-00151 and later with application firmware 1.1 and later
and Dante firmware 2.0.0 (Ultimo 3.10.1.12) and later

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Revision History

Issue 1, January 2017:

1. Initial release.

Introduction

The Model 371 Intercom Beltpack combines the features offered by broadcast-style 2-channel party-line (PL) intercom user devices with a single-channel listen output and a 4-pin male XLR headset interface connector. This output arrangement allows deployment in PL intercom applications where support for single-ear headsets is desired. These headsets, which utilize dynamic microphones, are widely used in theatrical, corporate, and industrial applications where it's important to have one ear remain open to ambient audio signals.

The Model 371 uses the advanced capabilities that Dante audio-over-Ethernet can provide. Over a standard IP network, multiple Model 371 units can be used in PL intercom applications with help from an external Dante-enabled audio matrix. Or, units can be used "point-to-point" or directly interfaced with ports on compatible matrix intercom systems. Only a single Power-over-Ethernet (PoE) connection is required for operation. Key user features can be easily configured including microphone preamplifier gain and talk button operation. Additional user features include integrated sidetone, remote microphone off, and call alerting. This capability, along with the great audio quality provided by the digital audio signal path, offers a superior user experience.

Set up and configuration of the 2-channel Model 371 is simple. An etherCON® RJ45 jack 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 microphone interfaces to the unit using a cable-mounted 4-pin female XLR connector. The connected headset can be either single- or dual-ear models but the Model 371 has only a single channel headphone audio output; the two listen audio channels are always combined and sent to the one audio output. Three DIP switches are used to establish the unit's operating parameters. Two "push-in/push-out" rotary level controls make it easy to set the headphone output volume as well as maintaining the desired settings. The Model 371'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 371's two audio channels is excellent, with low distortion, low noise, and high headroom. Careful

Figure 1. Model 371 Intercom Beltpack top and bottom views

circuit design and rugged components ensure long, reliable operation. A wide range of applications can be supported, including TV, radio, and streaming broadcast events, corporate and government AV installations, and industrial facilities.

Dante Audio-over-Ethernet

Audio data is sent to and received from the Model 371 using the Dante audio-over-Ethernet media networking technology. As a Dante-compliant device, the Model 371's two output (Dante transmitter) and two input (Dante receiver) audio channels can be assigned to other devices (routed) 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 24 with a sampling rate of 48 kHz. 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 371. Not only will it cause the talk button LEDs to light in a unique highly visible sequence, it will also turn off any active talk channels.

Audio Quality

The Model 371's completely "pro" performance really bucks the reputation of "intercom audio." A low-noise, wide dynamic-range microphone preamplifier and associated voltage-controller-amplifier (VCA) dynamics controller (compressor) ensures that input audio quality is preserved while minimizing the chance of signal overload. The output of the microphone preamp and compressor is routed to an analog-to-digital conversion (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, routes 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 the Dante receiver channels and pass into the Model 371's processor. The sampling rate is 48 kHz with a bit depth of up to 24. Headphone level control, channel mixing (combining), and sidetone creation are performed in the digital domain. This provides flexibility, allows precise control, and keeps the three user level potentiometers (channel 1, channel 2, and sidetone) from having to directly handle analog audio signals. The two audio channels destined for the headphone output are sent to a high-performance 2-channel digital-to-analog converter and then on to a robust driver circuit. High signal levels can be provided to a variety of headsets, headphones, and earpieces.

Call Function

A call function allows Model 371 users to send and receive channel-specific visual alert signals. Pressing the dedicated call button on the top of a 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 371 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 371 is its ability to be easily configured to the meet the needs of specific users and applications. Three DIP switches allow control of the microphone preamplifier gain and talk button operation. The gain of the microphone preamplifier can be selected from nominally 40 or 46 dB. This allows compatibility with dynamic microphones that are part of many industry-standard intercom and broadcast headsets. The two pushbutton switches that control the talk audio on/off status can be individually configured for push to talk or push to talk/tap to latch operation.

Ethernet Data and PoE

The Model 371 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 connector. 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 371'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 371's PoE interface reports to the power sourcing equipment (PSE) that it's a class 1 (very low power) device. If a PoE-enabled Ethernet port can't be provided by the associated Ethernet switch a low-cost PoE midspan power injector can be utilized.

Future Capabilities and Firmware Updating

The Model 371 was designed so 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.

To implement the Dante interface the Model 371 uses Audinate's Ultimo™ integrated circuit. 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 371 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. Should a PoE midspan power injector be required it must be purchased separately.

Connections

In this section signal interconnections will be made using the two connectors located on the bottom of the Model 371. 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 will be connected using a cable-mounted 4-pin female XLR connector.

Ethernet Connection with PoE

A 100BASE-TX Ethernet connection that supports Power-over-Ethernet (PoE) is required for Model 371 operation. This one connection will provide both the Ethernet data interface and power for the Model 371'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 371 supports Ethernet switch power management, enumerating itself as a PoE class 1 device.

The Ethernet connection is made by way of a Neutrik etherCON protected RJ45 connector that is located on the bottom panel of the Model 371. This allows connection by way of a cable-mounted etherCON connector or a standard RJ45 plug. The Model 371's Ethernet interface supports auto MDI/MDI-X so that a "cross-over" or "reversing" cable should not be required.

Ethernet Connection without PoE

As previously discussed in this guide, the Model 371 was designed such that the Ethernet connection will provide both data and Power-over-Ethernet (PoE) power. There may be situations where the associated Ethernet switch does not provide PoE power. In such cases an external PoE midspan power injector can be used. If the selected midspan power injector is 802.3af-compatible it should function correctly. Midspan units are available from a variety of sources, including many online retailers.

Headset Connection

The Model 371 provides a 4-pin male XLR connector that can interface with the microphone and headphone connections of many intercom or broadcast-style headsets. In most cases headsets associated with single-channel party-line intercom beltpacks will be directly compatible. Headset models with dynamic microphones from vendors such as Clear-Com®, Sennheiser, Shure®, and beyerdynamic should perform very well. Refer to Figure 2 for connection details.

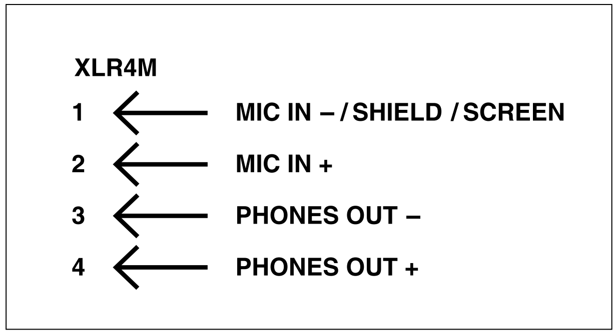

Figure 2. Headset connection pinout chart

The microphone input connections are intended primarily for use with unbalanced dynamic microphones. On the selected headset, connect microphone common, shield, or screen to pin 1 and microphone + to pin 2. Headsets with a balanced microphone should also function correctly. Connect both microphone – and common, shield or screen to pin 1 and microphone + to pin 2. No support is provided for unbalanced or balanced microphones that require a power source including low-voltage "eletret," P12 phantom, or P48 phantom.

A monaural (single-ear, single-sided, or "single-muff") headset should be wired such that its –, shield, or screen lead is connected to pin 3 and its + lead is connected to pin 4. To allow users of stereo (double-ear, double- sided, or "double-muff") headsets to hear the Model 371's audio output in both ears requires that both its -, shield, or screen leads be connected to pin 3 and both its + leads be connected to pin 4.

Dante Configuration

For audio to pass to and from the Model 371 requires that several Dante-related parameters be configured. These configuration settings will be stored in non-volatile memory within the Model 371'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 371 uses the Ultimo 2x2 channels (2-input/2-output) integrated circuit to implement the Dante architecture.

The two Dante transmitter (Tx) channels associated with the Model 371's Dante interface must be assigned to the desired receiver channels. This achieves routing the Model 371'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 371's transmitter channels need to be routed to more than two flows it's possible that an intermediary device, such as a Studio Technologies' Model 5422 Dante Intercom Audio Engine, can be used to "repeat" the signals.

The two Dante receiver (Rx) channels associated with the Model 371's audio inputs also need to be routed to the desired Dante transmitter channels. These two audio signals will be sent to the Model 371's headphone output.

The Model 371 supports an audio sample rate of 48 kHz with a limited selection of pull-up/pull-down values available. In most cases the default will be used and a pull-up or pull-down rate will not be selected. The Model 371 can serve as the clock master for a Dante network but in most cases it will be configured to "sync" to another device.

The Model 371 has a default Dante device name of **ST-M371** and a unique suffix. The suffix identifies the specific Model 371 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 Dante Controller the default device name and channel names can be revised as appropriate for the specific application.

Model 371 Configuration

Three DIP switches are used to select the microphone preamplifier gain and how the two talk buttons will function. These configuration choices allow the performance of the Model 371 to be optimized for specific applications and user preferences. The switches are accessible from the back of the unit's enclosure, through a small opening that is located under the top of the belt clip. To access the switches requires that the belt clip be rotated. Normally the belt clip is secured to the back using one rivet (non-removable) and one machine screw that has a thread pitch of 6-32. To allow the belt clip to rotate, remove the machine screw using a #1 Phillips-head screwdriver. The belt clip should now be able to be easily rotated in either direction. Save the screw so that it can be re-installed once the desired configuration choices have been made. Note that the threaded fastener within the back of the Model 371's chassis has an internal locking mechanism (a plastic bushing) that prevents the belt clip's 6-32 machine screw from vibrating loose. So no additional method of thread locking, such as a lock washer or chemical compound, is necessary.

The three switches provide a simple set of configuration choices. One switch allows the gain of the microphone preamplifier to be adjusted. The other two switches are used to select the manner in which the talk buttons will function. The switches are connected to the Model 371's logic circuitry which responds to changes by way of the application firmware; no audio passes directly through the switches. Changes made to any of the switches will immediately be reflected in the unit's operation. A power cycle is not required for configuration changes to be recognized and implemented. Two additional switches are not utilized as of the writing of this guide but may find application at a later date.

Microphone Preamp Gain

Switch SW1 allows the gain of the microphone preamplifier to be selected. When SW1 is in the down (off) position 40 dB of gain is selected. This 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.) To select 46 dB of gain place SW1 to the up (on) position. This additional 6 dB of gain 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 371 is going to be deployed at events where users are not

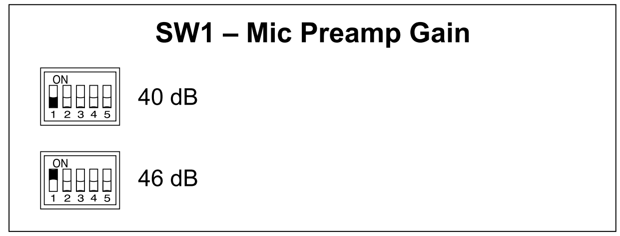

Figure 3. Microphone preamp gain configuration switch

able to speak at normal levels, e.g., sporting events such as golf tournaments.

The compressor active LED, visible on the bottom of the Model 371 adjacent to the headset connector, can act as a guide when setting the preamp gain. During normal talk operation the compressor active LED should light intermittently. If it rarely lights and the gain is set to 40 dB, it might be a good idea to change to setting to 46 dB. If the LED is lit fully during normal talking and the gain is set for 46 dB, changing it to the 40 dB setting might be warranted. There's no "hard and fast" rule about what gain setting is appropriate. But unless otherwise indicated, 40 dB is typically a good choice.

Talk Button Modes

On the Model 371's top panel are two pushbutton switches that allow the user to control the on/off status of the headset microphone audio that's sent to the two audio output (Dante transmitter) channels. Configuration switches SW2 and SW3 allow the buttons' operating modes to be independently configured. The two operating choices are push to talk and push to talk/tap to latch. SW2 is used to select the operating mode for the talk button associated with channel 1. SW3 is used to select the operating mode for the talk button associated with channel 2. Many applications are best served when the buttons are configured for

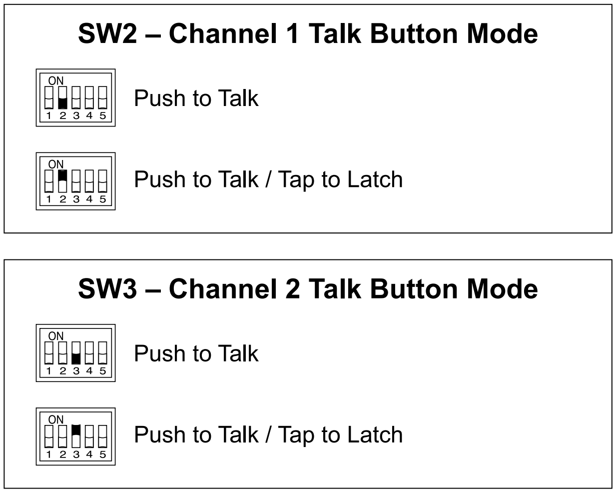

Figure 4. Talk button mode configuration switches

push to talk operation. This ensures that a channel won't accidentally be left in the talk mode. But there will certainly be valid situations where the push to talk/tap to latch settings will prove to be very useful.

Operation

At this point Model 371 operation should be taking place. An Ethernet connection with Power-over-Ethernet (PoE) capability should have been made. Alternately, a midspan power injector, in "series" with the Ethernet connection, should have been put into place. A headset terminated on a 4-pin female XLR connector should be plugged into the headset connector. Only a headset with a dynamic microphone will function correctly.

The Model 371's Dante configuration settings should have been selected using the Dante Controller software application. In this way the two audio output (Dante transmitter) channels and two audio input (Dante receiver) channels should have been routed to the Dante receivers and transmitters on associated equipment. The Model 371's three configuration switches should have been used to select the desired operating characteristics.

Initial Operation

The Model 371 will begin to function as soon as a Power-over-Ethernet (PoE) power source is connected. However, it may take 20 seconds or longer for full operation to commence. Upon initial power up the three status LEDs located on the bottom panel below the etherCON RJ45 jack will begin to light as network and Dante connections are established. The three pushbutton switches on the top panel will light in a test sequence to indicate that the main operating 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 connector on the Model 371'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 371 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 371 is not synchronized with a Dante network. It will light solid green when the Model 371 is synchronized with a Dante network and an external clock source (timing reference) is being received. It will slowly flash green when the Model 371 is part of a Dante network and is serving as a clock master. It's possible that up to 20 seconds may be required for the SYNC LED to reach its final state.

How to Identify a Specific Model 371/Remote Talk Off Function

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

The identify command also causes a talk off ("mic kill") function to activate. If either or both talk buttons are configured to the push to talk/tap to latch mode, and they are latched on, the identify command will cause them to latch off. This allows talk channels on a specific Model 371 that have been accidentally enabled to be remotely turned off. A subtle but interesting action will also occur if the identify command is received when the Model 371 is in a specific configuration and talk operation is active. If either or both of the talk buttons have been configured for push to talk operation and either or both buttons are pressed to enable talk, when the identify command is received the microphone audio will be muted for approximately one second. This is implemented to allow technical personnel to use the identify command to locate a specific Model 371 that might have had one or both of its buttons accidentally activated.

For example, if a Model 371 was removed from a user's belt or other clothing and then pushed into the corner of a table or studio set it's possible that either or both talk buttons could become depressed. The identify command would cause the audio path on that specific unit to be momentarily disabled, providing an aural "clue" as to which unit is in an unwanted state and helping guide personnel to identify its location. This feature only disturbs the audio path for a brief interval, ensuring that a valid talk condition won't be unduly impacted.

Listen Level

Two rotary level controls ("pots"), located on the Model 371's top panel, allow adjustment of the level of the two audio input signals as they are sent to the headphone output. The resulting audio signal that is sent to the headphone output will be a mix (combination) of the two input signals. The pots are "push-in/push-out" type which allows the associated knobs to be in their "out" positions to be adjusted and their "in" positions when protection from changes is desired.

The headphone output audio quality should be excellent, with high maximum output level and low distortion. High-frequency audio content is limited above 10 kHz, preventing unwanted signals from reaching the transducer(s) of the associated headset. Analog audio signals do not pass directly through the rotary level controls. The position of the pots is recognized by the Model 371's processor which then adjusts the signal level in 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 channels, or the status of the call function, does not impact the headphone output. They are all independent functions.

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 is such that the dynamic range of the talk signal is being controlled. It's perfectly acceptable for the LED to light intermittently while a user is talking in a normal voice level into the microphone. But if the COMP LED lights solid while a user is talking at a normal voice level and the mic preamp gain is set for 46 dB, this will typically indicate that the mic gain setting should be changed to 40 dB. Conversely, if the LED almost never lights when normal talking is taking place and the mic preamp gain is set for 40 dB, it's possible that changing the gain to 46 dB would be beneficial. The compressor active LED will function whether or not either or both of the talk channels are active.

Talk Buttons

Two pushbutton switches are associated with the Model 371's audio output channels. How they function will depend on the configuration of the unit. Each button can be configured independently. When a button has been set for 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 audio output channel. The button's green LED will light to indicate that the output is active. 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 changes states; off-to-on or on-to-off. Whenever the output is active the green LED will light.

Sidetone Function

The Model 371 includes a sidetone function that sends microphone audio to the headphone output whenever either or both of the talk functions are active. The audio quality should be excellent and will provide the user with a confidence signal that they are actively talking to other intercom users.

The level of the sidetone audio is adjusted using the rotary level control ("pot") located on the bottom of the unit. A small straight blade screwdriver (e.g., a "greenie") can be used to adjust the pot. Although with some practice one's thumb and index finger can also be effective. Typically the exact sidetone level is not critical and most users will not be concerned about changing it. But setting sidetone to a reasonable level 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. The two level controls on the top panel do not impact the sidetone level. Audio does not pass directly through the sidetone pot but instead is used by the Model 371's processor to control the level in the digital audio domain.

Call Function

An integrated call function allows Model 371 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 the 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 371's two audio input channels continually monitor for the presence of 20 kHz. A call signal is recognized on a channel 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 talk signal. Digital filters within the Model 371's processor integrated circuit limit the high frequency response of the audio signals to about 10 kHz in both the input and output signal paths. This helps to ensure that false call detection won't take place as well as limiting the chance of causing issues in events where headphone extended high-frequency response is possible, e.g., dog shows or other animal events.

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 or turn the call function(s) off and sending of the call signal will cease. As expected, pressing the call button when neither talk channel is active will result in nothing occurring.

Whenever a Model 371 audio input channel receives a call signal (20 kHz audio tone) the orange LED on its companion 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 channels the orange LED associated with the call button will not light.

By using a 20 kHz tone the Model 371 is compatible with legacy intercom equipment, including the venerable RTS BP-325. When interconnecting Model 371 and BP-325 units using an appropriate Dante-enabled interface, such as the Studio Technologies' Model 45DR, call signaling is fully compatible. Compatibility with the Clear-Com method of call functionality is possible by using the Model 45DC party-line interface. Also, devices such as the Studio Technologies' Model 44D Audio Interface product will send and receive 20 kHz signals that will be compatible. The Model 44D refers to these signals as GPI (general purpose input) and GPO (general purpose output) but the actual signaling utilizes 20 kHz tones which are transported “in band” via the Dante audio paths.

Technical Notes

IP Address Assignment

By default the Model 371's Ethernet interface will attempt to automatically obtain an IP address and associated settings using DHCP (Dynamic Host Configuration Protocol). If a DHCP server is not detected an IP address will automatically be assigned using the link-local protocol. This protocol is known in the Microsoft® world as Automatic Private IP Addressing (APIPA). It is also sometimes referred to as auto-IP (PIPPA). Link-local will assign an IP address in the IPv4 range of 169.254.0.1 to 169.254.255.254. In this way multiple Dante-enabled devices can be connected together and automatically function, whether or not a DHCP server is active on the LAN. Even two Dante-enabled devices that are directly interconnected using an RJ45 patch cord will, in most cases, correctly acquire IP addresses and be able to communicate and transport audio. An exception does arise when trying to directly interconnect two Dante-enabled devices that use the Ultimo Dante implementation, such as two Model 371 units. An Ethernet switch is required to provide the Ethernet ports to correctly interconnect two Ultimo-based devices directly with each other. (The technical reason relates to the need for the slight latency provided by an Ethernet switch.)

Using the Dante Controller software application the Model 371's IP address and related network parameters can be set for a fixed ("static") configuration. While this is a more involved process than simply letting DHCP or link-local "do their thing," if fixed addressing is necessary then that 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 IP address. If knowledge of a Model 371'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 personal computer connected directly to the Model 371. 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 QoS capability is recommended. This can typically be implemented on virtually all contemporary managed Ethernet switches. There are even specialized switches that are optimized for entertainment-associated applications. Refer to the Audinate website (www.audinate.com) for details on optimizing networks for Dante applications.

Application Firmware Version Display

As part of the Model 371's power-up sequence the unit's application firmware version number can be displayed. This is useful when working with factory personnel on application support and troubleshooting.

Before connecting the PoE Ethernet cable, press and hold the call button. Then connect the Ethernet cable. Upon application of power the Model 371 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. As an example of what would be a typical firmware display, if the channel 1 talk button “flashes” once followed by the channel 2 talk button “flashing” twice this would indicate that application firmware version 1.2 was present in the Model 371.

Application Firmware Update Procedure

It’s possible that updated versions of the application firmware (embedded software) that is utilized by the Model 371’s processor (microcontroller or MCU) integrated circuit will be released to add features or 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 memory by way of a USB interface. The Model 371 implements a USB host function that directly supports connection of a USB flash drive. The Model 371’s MCU updates its firmware using a file named **m371.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 firmware file in the root directory with a name of **m371.bin**. Studio Technologies will supply the application firmware file inside a .zip archive file. While the firmware file inside of the zip file will adhere to the naming convention required by the Model 371, the name of the zip file itself will include the file’s version number. For example, a file named **m371v1r1MCU.zip** would indicate that version 1.1 of the application firmware (**m371.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 load. The precise steps required will be highlighted in the next paragraphs of this guide. After the firmware has been updated the LEDs associated with the talk buttons should be used to confirm that the desired application firmware version has been successfully installed.

To install the application firmware file follow these steps:

1. Disconnect power from the Model 371. This will entail removing the Ethernet connection that is providing PoE power.
2. Remove the cover from the Model 371. Begin by removing the four Phillips-head screws (#1 screwdriver 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, then lift it off.
3. Locate the USB connector on the main circuit board. It’s adjacent to the call button. Insert the prepared USB flash drive into it.

4. Apply power to the Model 371 by connecting to a Power-over-Ethernet (PoE) Ethernet signal.
5. After a few seconds the Model 371 will run a “boot loader” program that will automatically load the new application firmware file (**m371.bin**). This load process takes only a few seconds. During this time period the call button’s LED will flash slowly in alternate colors. Once the entire loading process is over, taking approximately 10 seconds, the Model 371 will restart using the newly loaded application firmware.
6. At this time the Model 371 is functioning with the newly loaded 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. Press and hold the call button, apply power to the Model 371, and “read” the application firmware version number by observing the two talk button LEDs. Ensure that this is the desired version.

Note that upon power being applied to the Model 371 if the USB flash drive doesn’t have the correct file (**m371.bin**) in the root folder no harm will occur. Upon power up the call button’s LEDs will flash on and off rapidly for a few seconds to indicate this 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 371 implements Dante connectivity using the Ultimo 2x2 channels integrated circuit from Audinate. The Dante Controller software application can be used to determine the version of the firmware (embedded software) residing in the Ultimo “chip.” This firmware can be updated by way of the Model 371’s Ethernet connection. The latest Dante firmware file is available on the Studio Technologies website. The Dante Firmware Update Manager (FUM) application is used to install the firmware. This program is also available for download on the Studio Technologies website.

Specifications

Audio Channels: 2 talk, 2 listen

Power Source:

Power-over-Ethernet (PoE): class 1 (very low power, ≤3.84 watts)

Network Audio Technology:

Type: Dante Audio-over-Ethernet

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: twisted-pair Ethernet, Power-over-Ethernet (PoE) supported

Data Rate: 100 Mb/s (10 Mb/s Ethernet not supported)

Microphone Input:

Type: unbalanced, for use with dynamic microphones

Gain: 40 or 46 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 (combination of audio input 1 and audio input 2)

Compatibility: intended for connection to headsets or earpieces 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 Signal Support:

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 (intended for interfacing with a 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 etherCON RJ45

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

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 mounting adapter kit allows Model 371 to be permanently mounted

Weight: 0.6 pounds (0.3 kg)

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