Dante®

Model 352A Talk Station

Key Features

- Dante audio-over-Ethernet technology
- Supports GME-3-12 gooseneck microphone
- Voice-paging capability with background music support
- Broadcast talent cueing (IFB) capability
- Two independent output channels
- Analog monitor output with input selection
- Stores and replays two digital audio (WAV) files
- Excellent audio quality
- Uses STcontroller for configuration
- Power-over-Ethernet (PoE) powered

Introduction

The Model 352A Talk Station is a unique product intended to support a variety of voice paging, background music, audio-file playback, broadcast talent cueing (IFB), and general-audio applications. The unit supplies two "talk" channels that can be individually configured to match the needs of an application. A monitor section allows one of two Dante® receiver (input) channels to be routed as desired to a connected amplifier or amplified speaker.

The Model 352A supports Dante audio-over-Ethernet digital media technology with AES67 compatibility for integration into contemporary applications. In addition, the unit is compatible with the Dante Domain Manager™ (DDM) software application.

Voice audio is supplied by way of a gooseneck microphone, purchased separately, which is attached to the connector/threaded bushing combination that is located on the front of the Model 352A. (Using the Studio Technologies' GME-3-12 Gooseneck Microphone is highly recommended.) Two audio sources or "messages," up to 40 seconds each in length, are stored in non-volatile memory within the Model 352A. These two audio sources, saved in the common 16-bit monoaural WAV format, can be easily updated by way of a standard USB flash drive. The audio files can contain whatever "pre-page" (preamble) audio signals that are desired to support an application. They could be the sound of natural bells or chimes, an "electronically created" audio sequence, or a voice message. By using WAV files, the desired sources can be created outside of the Model 352A and then easily loaded for use. Configuration choices determine how and when the preamble audio files will be played.

The Model 352A is extremely simple to deploy, is "pro" quality throughout, and provides an intuitive user experience. The Model 352A's audio quality is excellent, with low distortion, low noise, and ample headroom. Careful circuit design and rugged components ensure long, reliable operation. Only a Power-over-Ethernet (PoE) connection and attachment of a gooseneck microphone are required for operation. If desired, the analog monitor output can be connected to the input of an associated device. Two "custom" WAV audio files can be loaded by way of a USB flash drive.

Model 352A operating features are configured using the STcontroller software application. An extensive set of parameters allows the unit's functions to be tailored to meet the needs of a myriad of applications. STcontroller is a fast and simple means of confirming and revising the unit's operating parameters. The Model 352A is housed in a compact, rugged steel enclosure that's intended for table-top use. Its small size makes it ideal for applications in space-constrained locations.

Applications

The Model 352A is ideal for creating a single- or dual-channel voice paging application. A signal processor or secondary device would not be required for full-featured deployments. Support for other applications abound. For theater or live-event spaces the Model 352A can be a complete 2-channel solution for routing background audio along with voice paging and preamble audio capability. Pre-recorded "tones" or "chimes" can be configured to play prior to voice audio being presented on one or both of the outputs. The unit could be conveniently located in a box office, manager's office, or back stage location.

More sophisticated applications can also be supported. A Model 352A function allows high-frequency "call" audio signals present in a Dante receiver (input) channel to be detected and displayed in the LED indicators that are associated with a "talk" pushbutton switch. This allows users to observe the on or off status of call signals that may be associated with intercom audio channels. In addition, this function can be used to display when a channel is active. This allows multiple Model 352A units to be deployed in a single application, with each unit displaying the talk status of all the others. A configuration option allows an associated pushbutton switch to be disabled whenever talk activity is taking place in other units. This feature will help prevent multiple users from simultaneously activating a specific talk function.

Broadcast applications can utilize the Model 352A to create a simple 2-channel talent cue (IFB) master station. Program audio sources can be routed to two of the unit's Dante receiver (input) channels and then interrupted with gooseneck microphone audio as desired.

Deployment of Dante applications could benefit from the unit's ability to store and playback WAV audio files. Test messages could be created in WAV format and then loaded into the Model 352A for continuous playback during system commissioning.

Security or management personnel can use the Model 352A as a combination voice paging console and monitor resource. Audio from the gooseneck microphone can be routed to either or both of Dante transmitter (output) channels. The monitor section allows either of the two Dante receiver (input) sources to be monitored by way of a connected amplifier or amplified speaker. Both analog and Dante monitor outputs are provided.

The range of resources provided by the Model 352A also makes it suitable for use in many other applications. It's designed to support the needs of varied applications across a wide range of audio system designs. The unit is intended to provide a "palette" of resources that meet real-world needs, allowing great things to be accomplished rather than presenting limitations that dictate what can and can't be done.

Setup and Operation

Set up, configuration, and operation of the Model 352A are simple. An RJ45 jack is used to interconnect the unit's Ethernet interface with a standard twisted-pair Ethernet port associated with a PoE-enabled network switch. This connection provides both power and bidirectional digital audio. A gooseneck microphone is attached to the Model 352A using the front-panel 3-conductor ¼-inch jack with integrated threaded bushing. The Studio Technologies' GME-3-12 Gooseneck Microphone is available as an option and will perform very well with the Model 352A.

Model 352A Talk Station shown with optional GME-3-12 Gooseneck Microphone

The monitor output (available in both analog and Dante digital formats) can assist users in confirming system operation. A 3-pin male XLR connector provides access to the analog monitor output. This would typically be connected to the analog input on an amplified loudspeaker. Amplified loudspeakers are also available with a Dante digital audio interface. One or two "custom" preamble audio files can be created and loaded into the Model 352A. They utilize the common WAV file format supported by many personal computer applications. The STcontroller software application is used to configure the wide range of Model 352A operating parameters. This allows the unit's performance to be optimized to meet the needs of specific applications.

The user is presented with three pushbutton switches and a push-in/push-out rotary level potentiometer. Two of the pushbutton switches are lighted with dual-color LEDs to represent the operating status of the unit's talk channels.

Ethernet Data and PoE

The Model 352A connects to a local area network (LAN) by way of a 100 Mb/s twisted-pair Ethernet interface. The unit's physical 100BASE-TX interconnection is made by way of a Neutrik® etherCON® RJ45 jack. While compatible with standard RJ45 plugs, an etherCON CAT5-compatible plug allows a ruggedized and locking interconnection for harsh or high-reliability environments. The Model 352A'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 an associated data network. To support PoE power management, the Model 352A's PoE interface enumerates (reports) to the power sourcing equipment (PSE) that it's a class 1 (very low power) device.

Dante Audio-over-Ethernet

Audio data is sent to and received from the Model 352A using the Dante audio-over-Ethernet media networking technology. As a Dante-compliant device, the Model 352A's three Dante transmitter (output) channels and four Dante receiver (input) channels can be assigned (routed or "subscribed") to other devices using the Dante Controller software application. The Dante transmitter (output) and receiver (input) channels are limited to supporting four Dante flows, two in each direction. The digital audio's bit depth is up to 32 with a sampling rate of 48 kHz. Two bi-color LEDs provide an indication of the Dante connection status. An additional LED displays the status of the associated Ethernet connection.

The Model 352A is compatible with the AES67 interoperability standard. When configured in Dante Controller to support AES67, the unit's Dante transmitter (output) channels will function in multicast; unicast is not supported. In addition, the Model 352A is compatible with the Dante Domain Manager (DDM) software application.

Audio Quality

The Model 352A provides excellent audio performance. A low-noise, wide dynamic-range microphone preamplifier and associated voltage-controlled-amplifier (VCA) dynamics controller (compressor) ensures that gooseneck microphone 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 32.

As previously discussed, the Model 352A can store and replay two audio files. These uncompressed PCM audio signals utilize the high-quality 16-bit, monaural, WAV format.

Audio sources can arrive in the Model 352A by way of four Dante receiver (input) channels. The supported sampling rate is 48 kHz with a bit depth of up to 32. These signals pass into the Model 352A's 32-bit microcontroller integrated circuit and can be used as part of monitor, background music, or talent cue (IFB) functions. The source selected for the monitor output is sent to both a Dante transmitter (output) channel and a high-performance digital-to-analog (DAC) integrated-circuit converter. The output of the DAC is then connected to a robust balanced (differential) analog output circuit. This analog output is protected from damage should it be accidentally connected to low-voltage DC, ESD ("static"), or other potentially damaging transients.

Depending on the unit's configuration and operating states, the audio content of the two Dante talk transmitter (output) channels will consist of a combination of gooseneck microphone, WAV audio file, and Dante program receiver (input) sources. The signals remain in the digital domain and are routed through the 32-bit logic circuitry and on to the Dante interface section where they are packetized and prepared for transport over Ethernet.

Configuration Flexibility

The Model 352A can easily be configured to meet the needs of many specific applications and user preferences. All configuration choices are performed using the STcontroller software application. Versions of STcontroller are available to support the WinOS® and macOS® operating systems. Selectable configurable parameters include microphone preamplifier gain, WAV file level trim, LED indicator intensity and actions, and monitor output source selection and operation. In addition, both of the talk functions can be optimized from numerous configuration choices. These choices allow the unit to be configured to support voice page, talent cueing (IFB), and other specialized applications.

As previously described, the functions associated with the two talk pushbutton switches and their Dante transmitter (output) channels can be individually configured. As an example, for a voice page application one of the talk channels might be configured to utilize a program music source and play, before the microphone is active, a pre-page message. The second talk channel might be used as a voice-page output with no background music or pre-page audio.

While a range of choices is available, for voice-paging applications the two talk pushbutton switches would typically be configured to provide a momentary (push to activate) function. Red and green LEDs are associated with the pushbutton switches. When lit singly, the LEDs will provide a red or green indication. When both LEDs are lit simultaneously, they will provide an orange indication. A configuration choice allows selection of how the LEDs will light under various operating conditions. This capability is provided to assist users who have trouble observing the differences between colors, typically issues differentiating between red and green. A person experiencing a "color blindness" condition is not uncommon in men and can make effective use of some electronic equipment difficult. The Model 352A's ability to control the talk pushbutton switches' colors can also be useful for supporting international applications where compliance to specific regulatory requirements is required. This may dictate which LED color is associated with a function being "on" and which color is associated with a function being "off."

The two program and two monitor Dante receiver (input) audio sources and the way in which they are assigned to the talk and monitor outputs can be configured. Unique choices allow a number of audio monitoring situations to be implemented. Whether for use in voice paging, talent cueing (IFB), or monitoring applications, the Model 352A should be able to achieve the desired configuration. Several special functions allow the WAV files and gooseneck microphone audio to be routed to the analog and Dante monitor outputs. Configured using STcontroller, these functions can allow a user to hear confirmation audio related to the unit's real-time operation.

Future Capabilities and Firmware Updating

The Model 352A was designed so that its capabilities and performance can be enhanced in the future. The unit implements a USB host function which allows the application firmware (embedded software) to be updated using a standard USB flash drive. And, as previously discussed, the USB receptacle is also used to load WAV audio files that can be stored on the same USB flash drive. The Model 352A uses the Audinate UltimoX4™ integrated circuit to implement its Dante interface. The firmware in this integrated circuit can be updated via the Ethernet connection, helping to ensure that the unit's capabilities remain up to date

Model 352A Specifications

Applications: Dante-based voice paging, intercom, talent cue (broadcast IFB), and general audio installations

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

General Audio:
Internal Digital Audio Processing: 32-bit, fixed
Nominal Digital Input and Output Level: –20 dBFS

Network Audio Technology:
Type: Dante audio-over-Ethernet
AES67-2018 Support: yes
Dante Domain Manager™ (DDM) Support: yes
Bit Depth: 16, 24, or 32
Sample Rate: 48 kHz
Number of Transmitter (Output) Channels: 3
Number of Receiver (Input) Channels: 4
Dante Audio Flows: 4; 2 transmitter, 2 receiver

Network Interface:
Type: 100BASE-TX, twisted-pair Ethernet with Power-over-Ethernet (PoE) supported
Data Rate: 100 Mb/s (10 Mb/s and 1000 Mb/s GigE Ethernet not supported)

Microphone Input:
Compatibility: Studio Technologies' GME-3-12 Gooseneck Microphone or equivalent
Microphone Power: 3.3 volts DC via 2.49 k resistor
Impedance: 2.0 k ohms, nominal
Gain: 12, 18, 24, 30, 36 dB, selectable
Frequency Response: –3 dB at 40 Hz, –1 dB at 16 kHz
Distortion (THD+N): 0.022%, measured at –20 dBFS, 22 Hz to 22 kHz bandwidth, 12 dB of gain
Noise Floor: –95 dBFS, A-weighted, 12 dB of gain
Dynamic Range: >76 dB, A-weighted, measured at 36 dB of gain

Compressor:
Application: applies only to gooseneck microphone audio
Threshold: 2.7 dB above nominal Dante output level (–17.3 dBFS), ±0.3 dB
Slope: 2:1
Status LED: compressor active

Preamble Audio:
Number of Channels: 2, field updatable using USB flash drive
Source Type: 16-bit monophonic, 48 kHz sample rate, WAV (.wav) files, stored in non-volatile memory
Level: –20 dBFS nominal, adjustable ±12 dB in 3-dB steps
Duration: up to 40 seconds per WAV file
Distortion: <0.0001%
Frequency Response: 20 Hz to 20 kHz, +0/–0.7 dB

Background Music or Talent Cue (Broadcast IFB) Capability:
Number of Channels: 1 or 2, configurable
Audio Sources: two Dante receiver (input) channels
Frequency Response: 20 Hz to 20 kHz, +0/–0.7 dB
Program Audio Attenuation (Dim): 0, 5, 10, 15, 20 dB, Full Mute, configurable

Dante Talk Outputs: 2
Type: Dante digital
Nominal Level: –20 dBFS

Analog Monitor Output:
Type: electronically balanced, capacitor coupled, intended to drive balanced or unbalanced loads of 2 k ohms or greater
Audio Source: two Dante receiver (input) channels, two preamble audio sources, and microphone audio, configurable and selectable
Source Impedance: 200 ohms
Nominal Level: –10, 0, or +4 dBu, reference –20 dBFS, configurable
Maximum Level: +20 dBu, with 0 dBFS input, measured at 1 kHz
Dynamic Range: >106 dB, A-weighted
Distortion (THD+N): 0.0011% (–99 dB), measured at –1 dBFS input, 22 kHz bandwidth
Frequency Response: +0/–1.5 dB, 20 Hz to 20 kHz
Level Reduction (Dim) Capability: 0, 5, 10, 15, 20 dB, or full mute, configurable

Dante Monitor Output:
Type: Dante digital
Audio Source: two Dante receiver (input) channels, two preamble audio sources, and microphone audio, configurable and selectable
Nominal Level: –30, –24, or –20 dBFS, reference –20 dBFS, configurable

Channel Status Signals:
Action: independently configurable per channel
Send Tone Frequencies: 16, 18, and 20 kHz, sine wave
Send Tone Frequency Accuracy: <10 ppm
Send Tone Level: –20 dBFS, nominal
Send Tone Distortion: <0.0001%
Receiver Tone Detect Level Threshold: –23 dBFS at 16 kHz; –28 dBFS at 18 kHz; –30 dBFS at 20 kHz
Receiver Tone Minimum On Time: 80 milliseconds
Program Input-to-Monitor Output Low-Pass Filter: –6 dB at 10 kHz; –28 dB at 16 kHz; –55 dB at 20 kHz
Tone Detect Tone-to-Monitor Output Rejection Filter: –31 dB at 18 kHz; –46 dB at 20 kHz; –70 dB at 22 kHz

Connectors:
Gooseneck Microphone: 3-conductor ¼-inch jack with 7/16-20 UNF threaded bushing; 4-40 hex head socket set screw allows microphone to be secured into bushing
Analog Monitor Output: 3-pin male XLR
Ethernet: Neutrik NE8FBH etherCON RJ45 jack (compatible with etherCON CAT5 plug)
USB: type A receptacle (used only for updating WAV audio and firmware files)

Configuration: Studio Technologies' STcontroller personal computer application

Software Updating: USB flash drive used for updating main firmware and WAV audio files; Dante Controller's Dante Updater application used for updating Dante interface firmware

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: 5 to 95%, non-condensing
Altitude: not characterized

Dimensions:
3.6 inches wide (9.1 cm)
2.7 inches high (6.9 cm)
4.7 inches deep (11.9 cm)

Deployment: intended for tabletop applications

Weight: 1.1 pounds (0.50 kg) without gooseneck microphone; 1.4 pounds (0.64 kg) with GME-3-12 Gooseneck Microphone

Specifications and information subject to change without notice.

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