Model 212
Announcer's Console

The Model 212 Announcer's Console is designed to serve as the audio control center for announcers, commentators, and production talent. With its digital audio input and output resources the tabletop unit is well suited for numerous applications including on-air television sports broadcasting. The broadcast-standard interfaces allow direct integration into contemporary "all digital" environments. The Model 212 integrates all on-air, talkback, and cue audio signal routing into one compact system. Ease of use, flexible in configuration, and sonically excellent are some of the unit's highlights.

Whether it's microphone pre-amplification, switching of main and talkback audio signals, or the headphone cue feed output, superior audio quality is maintained throughout. A microprocessor provides the Model 212's logic power, allowing precise control of the unit's operation. A range of configuration choices allows the desired operating configuration to be easily selected. While flexible, the user is presented with an easy-to-use set of controls and indicators. The digital audio interfaces are directly compatible with system environments that use AES digital audio signals as their preferred transmission method. Unbalanced, 75 ohm interfaces (AES3id) are standard with balanced 110 ohm (AES3) support also available.

A truly next-generation product, extensive research into the needs and desires of field production personnel was integral to the Model 212's creation. While primarily targeted for on-air television applications, specialized features are included to allow the Model 212 to be used in a variety of other audio applications. These include on-air radio broadcasting, stadium announcement, and voice-over/narration booths.

Model 212 front panel shown with buttons labeled for on-air applications

STUDIO TECHNOLOGIES INC.

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Microphone Input
A high-performance microphone preamplifier circuit provides low-noise/low-distortion amplification over a 10 to 50 dB gain range. The gain is adjustable in 10 dB steps. The input is compatible with balanced dynamic or condenser microphones. The microphone power source is 48 volts nominal and meets the worldwide P48 phantom standard. An LED indicator serves as an aid for optimizing the setting of the preamplifier's gain. The output of the microphone preamplifier is routed to the compressor circuit that supports the main and talkback audio channels.

Dynamic Range Control
A studio-quality compressor circuit is provided to control the dynamic range of the analog signal coming from the microphone preamplifier. Far from a simple "clipper," the circuit utilizes a sophisticated laser-trimmed voltage-controlled-amplifier (VCA) integrated circuit for quiet, low-distortion level control. The output of the compressor is used by the main and talkback channels, being routed to the inputs of their respective analog-to-digital converter circuits. The threshold of the compressor can be configured from among two values. When configured for –4 dBFS dynamic range control will only take place as the signal approaches the level limit of the digital signal. During normal operation the compressor would not be active. With the compressor configured to –14 dBFS some dynamic range control would be expected during normal operation. While not appropriate for major on-air situations, having dynamic range control of the signal can offer increased performance for many applications. This would be appropriate for applications such as stadium announcement positions, sports events using nonprofessional on-air talent, as well as situations where the risk of digital "clipping" is present.

Main and Talkback Audio Channels
Internally, the Model 212 creates analog main and talkback audio channels. The main audio channel is designed to serve as the on-air, stadium announcement, or other primary audio feed. The talkback audio channel is intended to provide production trucks, control rooms, or support personnel with a talent-originated cue signal. The two analog channels are converted to 24-bit PCM digital audio and multiplexed and formatted into an AES digital audio "stream." The A/D converters and digital transmitter circuitry is synchronized ("locked") to the selected digital input signal. The system will automatically revert to the internal clock, which provides a 48 kHz sampling rate. The main audio channel is assigned to digital channel 1 with talkback audio on digital channel 2. The composite digital audio signal is routed to two physical interfaces. One output is AES3id compliant (unbalanced, 75 ohms) with a BNC connector for interfacing. This professional interface is transformer-coupled and capable of driving long broadcast cable runs.

The digital audio output signal is also routed to a unique bidirectional digital interface. Similar to AES3id, it uses a BNC connector and supports two channels of audio in each direction over a single 75 ohm cable. The bidirectional audio interface will find use in specialized applications, including direct integration with Riedel® intercom systems. Note that audio signals received from the bidirectional interface are described in the Cue Sources section of this document.

For enhanced flexibility, the Model 212 includes the resources to support a third digital output. Circuitry, located on the printed circuit board assembly, creates an AES3 compliant, 110 ohm nominal, transformer-balanced digital output. Gaining access to this output simply requires adding a 3-pin male XLR-type connector to one of the two spare connector locations on the Model 212's back panel. Connector kits, available from Studio Technologies, make this a simple task.

For specialized applications an optional direct microphone output card is available. This allows the Model 212 to support an analog signal chain for the main output channel. Of course this output also provides "click-free" microphone on/off ("muting") control.

The Model 212 isn't limited for use in "on-air" applications. A special feature can be enabled, placing the unit in a "production" mode. When this mode is active enabling the talkback audio channel will no longer force the main audio channel to mute. This allows the main audio channel to be used as a second talkback output. In this configuration the unit can be even more powerful when used in a live event application, such as serving as a master console for a production director.

User Controls and Status Indicators
Two pushbutton switches, three LED indicators, and two rotary controls provide the user with a clear, easy-to-use interface. One pushbutton switch controls the status of the main audio channel. This is the audio signal intended for

Product Highlights:

- Excellent Audio Quality
- Simple User Interface
- Standard Connectors
- Configuration Flexibility
- Next-Generation Performance
- Digital Audio Inputs and Outputs

on-air, announcement, or other primary uses. Two LEDs display the on/off status of the main audio channel. A second push-button switch controls the status of the talkback audio channel. This is the audio signal used to communicate with producers, directors, spotters, or other behind-the-scenes production personnel. A status LED is associated with the talkback button. Two rotary controls allow the user to adjust the level of the stereo headphone output.

Operational Flexibility

Much of the Model 212's unique capabilities derive from the ability to configure the on/off control of the main and talkback audio channels. To meet the needs of the many specific broadcast and production applications, a variety of button operating modes are available. The main button, located on the left side of the front panel, can be selected to operate from among four modes. In the "push-to-mute" mode the button performs a momentary mute of the main audio channel. In this way a "cough" button function is created, something typically required for television sports broadcasting. In the "push-to-talk" mode the button provides a momentary active function for the main audio channel. This mode would be appropriate for applications such as stadium announcement. An alternate action "latching" configuration allows the button to enable or disable the main audio channel as desired. This is useful in radio broadcasting, announce-booth, or voice-over applications. The fourth mode provides a hybrid function, supporting both push-to-talk and tap-to-enable/tap-to-disable operation. This operation is similar to that found in many broadcast intercom system user stations.

The button associated with the talkback function can be configured to operate from either of two modes. One of the modes supports a "push-to-talk" function. This is typically used for on-air broadcast applications. The other mode provides a hybrid function, the operation of which is discussed in the previous paragraph. The hybrid mode is especially useful when the Model 212 is used in a production-support application.

Cue Sources

The Model 212 allows two audio channels from one of three digital audio sources to be routed to the stereo headphone output. Configuration switches select which digital input is active. The available sources are the AES3id input, the previously discussed bidirectional digital interface, and the optionally implemented AES3 input. The AES3id input is an unbalanced, 75 ohm interface that uses a BNC connector. Transformer-coupled, it is compatible with standard unbalanced digital audio signals. The bidirectional digital interface is similar to AES3id but carries two audio channels in each direction. It is intended for use in special applications. The AES3 input is balanced, 110 ohms, and to use requires the addition of a connector on the unit's back panel. No matter which of the digital inputs is selected, the receiver circuitry can accept bit depths of 16 to 24. While the nominal sample rate is 48 kHz,

Model 212 back panel

any signal in the range of 32 to 96 kHz is acceptable. Note that the received clock signal is used for two purposes; it allows correct receiver synchronization as well as serving as the transmitter clock source for the digital outputs.

Digital-to-analog conversion circuitry creates two analog "cue" audio signals from the selected digital interface. In on-air broadcast applications these cue signals, originating in production trailers, control rooms, or remote locations, supply program-with-interrupt audio on one channel and program-only audio on the other. Each source can be individually assigned to the left channel, right channel, or both left and right. This allows a wide variety of stereo and mono headphone mixes to be created.

Some applications may benefit by being able to connect standard line-level audio signals to the Model 212. To meet this need one or two optional line input cards can be installed in the Model 212's back panel. Each card provides a female XLR-type connector and transformer-isolated +4 dBu nominal input circuit. Each source can be individually assigned to the left channel, right channel, or both left and right.

Headphone Output

Two rotary controls are provided for user adjustment of the headphone output levels. For application flexibility the actual function of the two "pots" is configurable. For traditional on-air sports applications they can be selected to the dual-channel ("level/level") mode which provides independent control of the left- and right-channel volume. For use with dual-channel cue signals, or to support user preference, the stereo ("level/balance") mode can be selected. In this mode one control adjusts the overall level of both the left and right channels, while the other allows adjustment of the left/right level balance. To help minimize the chance of broadcast cues being missed, both level control modes can be configured so that a minimum headphone output level is maintained. Alternately, the headphone output can be set to fully mute when the controls are at their minimum position. The rotary controls can also be set for a reverse left/right mode. This is provided for cases where a user needs to wear their headset "reversed" to support alternate camera angles. By using this special mode intuitive operation will be maintained and a comfortable work environment assured.

Provision has been made to support applications where a monaural cue feed is desired. A configuration switch allows the summing (combining) of the selected left and right headphone sources. In addition to creating a dual-channel mono output it also allows the level controls to be configured as a simple 2-channel mixer.

The headphone output was designed to meet the needs of contemporary headphones and headsets. Specifically, the output circuits act as voltage, rather than power, drivers. In this configuration they can provide high output levels with very low distortion and noise, along with minimal current consumption. The output circuits are configured to safely drive stereo or mono loads. This ensures that all types of headphones, headsets, and earpieces can be directly connected.

Audio Quality and Protection

The Model 212's circuitry was carefully tailored to provide excellent audio performance. Professional-quality components are featured throughout. For reliability all audio routing is performed using solid-state devices. In all critical audio paths, "clickless" performance is provided. All audio inputs and outputs make extensive use of protection components. This limits the chance of damage from ESD and other undesirable, yet real-world, hazards.

Power Sources

The Model 212 requires an external source of nominal 24 volt DC, 150 milliamperes maximum, for operation. Inside the Model 212, an internal switch-mode power supply creates the various voltage "rails" required by the internal circuitry.

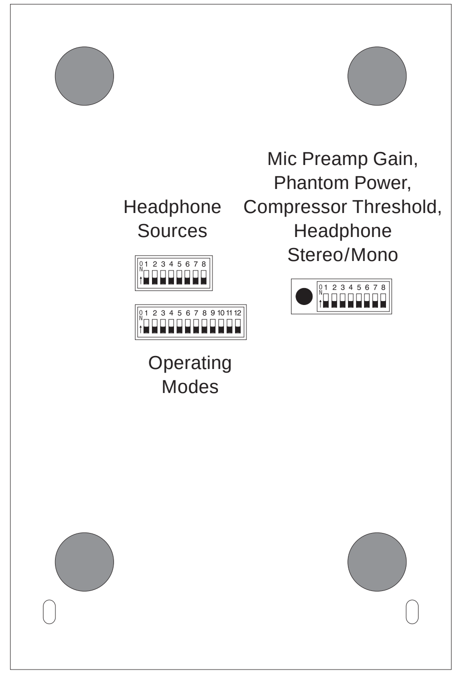
An external power supply is included with each unit. As an alternative, the Model 44 Interface can be used to power the Model 212. Available from Studio Technologies, it serves as an interconnection "hub," providing power and signal routing for up to six Model 212 Announcer's Consoles. Refer to the Studio Technologies website for details.

Auxiliary Relay

Model 212 resources include a general-purpose relay, allowing specialized configurations to be created. Under software control, the relay can be configured to follow the state of the main and talkback audio channels. Taking advantage of the back-panel locations provided for additional XLR-type connectors, a technician may easily implement a variety of functions such as providing an "on-air" indicator or performing loudspeaker muting during talkback. The auxiliary relay is also used by the optional direct microphone output card.

Configuration

Model 212 configurations are made using a number of DIP-type switches. One 8-position switch array is used to set the gain of the microphone preamplifier, the on/off status of phantom power, the compressor threshold, and the headphone stereo/mono mode. Another 8-position switch array configures which of the audio sources is routed to the headphone output. A 12-position switch array communicates the desired operating modes to the microprocessor. All switches are accessible via the bottom of the Model 212's enclosure; the unit does not have to be disassembled. Changes made to any of the configuration parameters become active immediately. To prevent access to the configuration switches a security panel, included with each unit, is attached to the bottom of the enclosure.



Model 212 bottom view showing configuration switches

Connectors

The Model 212 uses standard connectors throughout. The microphone input uses a 3-pin female XLR-type connector. The digital audio inputs and outputs use BNC connectors. A ¼-inch 3-conductor jack is used for the headphone output. The external source of 24 volt DC power is connected by way of a 2.1 x 5.5 mm "locking" coaxial power jack.

In the world of broadcast and production audio it's fair to say that applications vary widely. To this end, one or two additional XLR-type connectors can easily be mounted into the Model 212's back panel. Multiple 3-position "headers" located on the Model 212's circuit board provide technician-access to all input and output connections. Using a factory-available interface cable kit allows a Model 212 to be optimized to meet the exact needs of specific applications. For example, some applications may prefer to use a multi-pin XLR-type connector to interface with a headset. This can easily be accomplished by adding the appropriate 5-, 6-, or 7-pin XLR-type connector and making a few simple connections. Other applications may benefit from having "mult" or "loop-through" connections, something easily incorporated into a Model 212. One or two optional line input cards, as previously discussed, can also be mounted in the spare XLR positions.

Model 212 Specifications

General Audio:

Frequency Response: < –0.1 dB at 20 Hz, –1 dB at 18 kHz, mic in/main out

Distortion (THD+N): 0.025%, measured at 1 kHz, mic in/main out

S/N Ratio: 71 dB, referenced to –46 dBu mic in/–14 dBFS out

Common Mode Rejection Ratio: 68 dB at 60 Hz

Connectors:

Mic In: 3-pin female XLR-type

AES3id Out, AES3id In, Bidirectional Interface: 75 ohm BNC

Headphone Out: ¼-inch 3-conductor phone jack

24 Vdc Power In: coaxial power jack, 2.1 x 5.5 mm, locking bushing, compatible with Switchcraft S760K plug

Spare Connector Locations: 2

Allows one or two Neutrik NC*D-L-1 connectors to be installed (*=3F, 3M, 5F, 5M, 6F, 6FS, etc.)

Digital Inputs/Outputs:

A/D, D/A Converters: 24 bit, 100 dB dynamic range

Sampling Rate: internal clock source 48 kHz, external clock source 32 to 96 kHz

AES3id In: unbalanced, transformer-coupled
Impedance: 75 ohms
Minimum Input Carrier Amplitude: 320 mVpp (SPDIF compatible)

AES3id Out: unbalanced, transformer-coupled
Impedance: 75 ohms
Output Carrier Amplitude (Loaded): 1 Vpp

Bidirectional: unbalanced transformer-coupled
Impedance: 75 ohms
Output Carrier Amplitude (Loaded): 1.75 Vpp
Minimum Input Carrier Amplitude: 320 mVpp

AES3 In: balanced, transformer-coupled (via optional connector)
Impedance: 110 ohms
Minimum Input Carrier Amplitude: 200 mVpp

AES3 Out: balanced, transformer-coupled (via optional connector)
Impedance: 110 ohms
Output Carrier Amplitude (Loaded): 3.5 Vpp

Microphone Input/Preamplifier:

Type: electronically balanced

Input Impedance: 2 k ohms, nominal

Gain Range: 10 to 50 dB, nominal, adjustable in 10 dB steps

Compatibility: dynamic or phantom-powered mics

Phantom Power: 48 Vdc, nominal, meets IEC 61938

Optional Line Inputs: 2

Type: balanced, transformer-coupled

Impedance: 10 k ohms, nominal

Nominal Level: +4 dBu

Compressor:

Threshold: switchable –14 dBFS/–4 dBFS

Attack/Release Time: 2 mSec/100 mSec, nominal

Slope: 5:1, nominal

Status LED: compressor active

Headphone Output: 1, stereo

Compatibility: intended for connection to mono or stereo headphones or headsets with nominal impedance of 100 ohms or greater

Type: voltage driver

Maximum Output Voltage: 8 Vpp, 150 ohm load

Auxiliary Relay:

Function: software configurable

Contacts: 2, form C (Common, Normally Closed, Normally Open)

Rating: 1 A, 30 W (resistive)

Access: requires user-implemented connector scheme

Power Source:

24 Vdc nominal, 150 mA max @ 24 Vdc; acceptable range 20-30 Vdc. Units shipped to North America and Japan include a 120 V input/24 Vdc output power supply. Units shipped to all other locations include a universal input/24 Vdc output power supply.

Options: one or two optional XLR cards can be installed to provide support for connection of AES3, line-level balanced or unbalanced audio sources

Dimensions (Overall):

5.6 inches wide (14.2 cm)
3.3 inches high (8.4 cm)
8.5 inches deep (22.4 cm)

Weight: 3.2 pounds (1.45 kg)

Features and specifications subject to change without notice.

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