XAP™ Audio Conferencing

White Paper

ClearOne®

You're Virtually There℠

XAP White Paper
ClearOne Part No. 803-151-101
August 2002 (Rev. 2.0)

XAP White Paper

Table of Contents

CHAPTER 1: Introduction

Overview

Audio is critical to human communication. Media such as voice mail, the Internet, conference calling, video conferencing, and electronic presentations are driving the demand for better audio-communication technologies. The audio conferencing and video conferencing arenas also demand higher-quality sound. The XAP™ product line provides better sound quality as well as the ability to provide total sound system performance for telecommunications and any other multimedia audio event.

The XAP 800 and XAP 400 Audio Conferencing Systems meet the demands of a wide variety of conferencing and sound reinforcement requirements with digital matrix mixing, ClearOne's proprietary Gentner® Distributed Echo Cancellation®, noise cancellation, parametric equalizers, filters, and 32 customizable presets. These features enable the XAP 800 and XAP 400 to create a quality audio experience in many venues—from corporate briefing centers and courtrooms to training rooms and boardrooms.

A quality teleconferencing experience is one where the audio or audio source material is the message, not the audible inadequacies of a poorly designed or configured teleconferencing system. With a properly configured XAP system, participants and observers do not become fatigued by reverberated audio and poor sound clarity.

For ease of use, XAP systems facilitate local and remote PC setup, control, and diagnostics; compatibility with custom control panels; integration with popular control systems; logic outputs, and gated microphone operation. Microphone inputs and mic mixing parameters can be individually customized, while automatic gain control keeps the overall sound level consistent. Input channels can be configured as an automatic microphone mixer.

Any combination of inputs can be routed to any combination of outputs, allowing flexibility in accommodating different applications and customer requirements.

Adjustments in routing, level, and other functions can be made through presets activated through a closure on the rear panel or through an RS-232 serial interface.

New XAP Enhancements

The G-Ware™ 4.0.1 and 4.5 releases provide several enhancements to the XAP product line that are designed to help with system configuration.

- **Virtual references.** Improves operation in speech reinforcement and stereo applications by increasing flexibility in setting up echo canceller reference points. The virtual reference allows an installer to select sources going to multiple outputs or to subtract sources from an output for accurate echo cancellation. Previously, an analog output was used to set up a pseudo reference.
- **Front panel gain and mute control.** Gain and mute adjustments can be made from the front panel without connecting through G-Ware.
- **Safety Mute.** Mutes all outputs with one simple click if feedback or audio problems occur during system configuration.
- **Preset and macro passwords.** Presets and macros can be password protected to prevent unauthorized users from making changes.
- **Clear Matrix.** This new button on the Matrix Screen makes it easy to clear all cross point cells in the matrix.
- **Ramp serial command.** The new Ramp serial command gives you greater control over gain adjustments. You can specify the rate at which the gain increases and decreases, as well as the target level.Multiple outputs may be controlled simultaneously by writing a macro containing multiple ramp commands.
- **800x600 resolution.** A scroll bar has been added to the Inputs configuration window so you can view Input 8 when using 800x600 resolution. 1024x768 is still recommended.
- **Signal Generator indicator.** This toolbar indicator illuminates when the signal generator is active.
- **Preset/Programming Output Mute.** When synchronizing to the unit, all outputs will be muted to prevent extraneous noises or popping sounds. When changing a preset, the outputs associated with a particular preset will be muted again to prevent unwanted noise.
- **The new XAP IR Remote Control.** Upgraded systems now support the XAP IR Remote Control.
- **G-Switcher.** G-Switcher is a new program that allows users to maintain multiple versions of G-Ware on the same computer and easily switch between them. (G-Switcher is available only in the 4.5 release.)

XAP 800 Product Description

XAP 800 overview

The XAP 800 is a highly-advanced audio conferencing system with a twelve-by-twelve digital matrix mixer, Gentner® Distributed Echo Cancellation® (Gentner® D.E.C.®), noise cancellation, and audio processing. It utilizes an internal macro language and 32 user-definable presets to quickly adapt to a variety of sound reinforcement and room-combining applications in courtrooms, multimedia centers, hotel/convention centers, conference rooms, training rooms, and boardrooms.

The XAP 800 features:

- Gentner D.E.C. places an acoustic echo canceller on each mic input for greater echo cancellation flexibility and effectiveness.
- Noise cancellation on each mic input to reduce unwanted ambient noise.
- 8 mic/line inputs:
  - Input gain control.
  - Configurable processors with four filters (parametric EQ, high-pass, low-pass, all-pass, and notch).
  - Automatic gain control with speech leveler for consistent audio levels, 4 line inputs.
- Twelve line output channels. All output levels are adjustable and can be muted.
- 12x12 matrix mixer with level control at the cross points.
- Any combination of inputs can be routed to any combination of outputs.
- Adjustable automatic mic mixer:
  - 8-channel automatic mix mixer. Up to four mixers operate across bus.
  - Four internal mixers and four global mixers linked across expansion bus.
  - Adjustable parameters.
- Eight audio processing buses, each with 15 filters, can be placed anywhere within the matrix mixer audio path.
- Remote and local PC setup and diagnostics.
- 32 programmable presets for instant configuration changes.
- Network-based interconnectivity allows up to eight XAP 800s to be connected and controlled as a single unit
- Logic outputs.
- ClearOne service and support. One-year limited warranty.

XAP 800 front panel

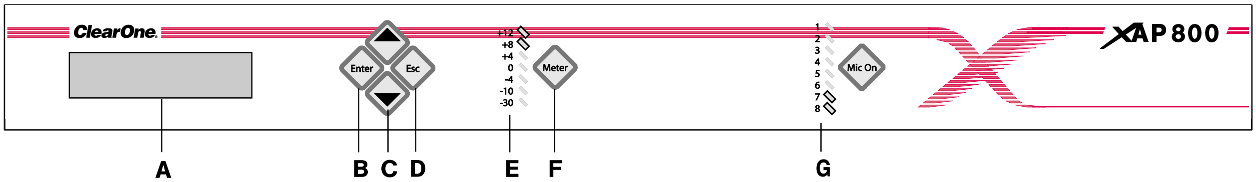


Figure 1.1. XAP 800 front panel

A. **LCD.** Used for numeric display of audio levels, gain readouts, and limited set-up and programming functions.

B. **Enter/▲▼/ESC.** Used to navigate the XAP 800's menu system.

C. **LED Meter.** This LED bar meter is used to display the audio level of an input, output, or processing channel of the XAP 800.

D. **Meter**. Takes you directly to the Meter branch of the XAP 800's LCD programming tree.

E. **Mic On LED.** These LEDs indicate microphone gate status.

XAP 800 rear panel

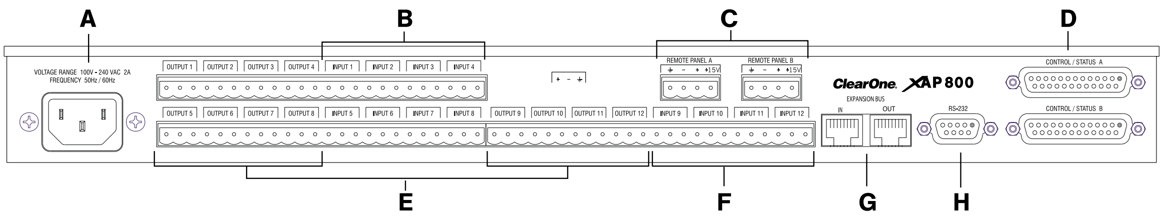


Figure 1.2. XAP 800 rear panel connections

A. **Power.** The power module accommodates an AC voltage input of 100–240VAC, 50/60Hz, 30W. No switching is required.

B. **Inputs 1–8.** For mic and/or line level inputs.

C. **RS-485 Remote Control Ports.** These four-pin Phoenix ports allow you to control the XAP 800 with a ClearOne Control Panel or the XAP IR Remote Control.

D. **Control/Status Ports A and B.** These DB25 connectors connect control devices. The control devices have access to the command set for the XAP 800 and can be used for functions such as volume, muting, preset change, etc.

E. **Outputs 1–8, 9–12.** Line level outputs that may be configured for any combination of gated and non-gated inputs, as well as a mix of mic and line level inputs.

F. **Inputs 9–12.** For line level inputs such as video codecs, XAP TH2s, and other auxiliary audio sources.

G. **Expansion Bus In/Out.** Used for daisy-chaining XAP units in a network.

H. **RS-232.** This DB9 serial port is for interconnection between the XAP 800 and a PC, modem, or other custom remote controller.

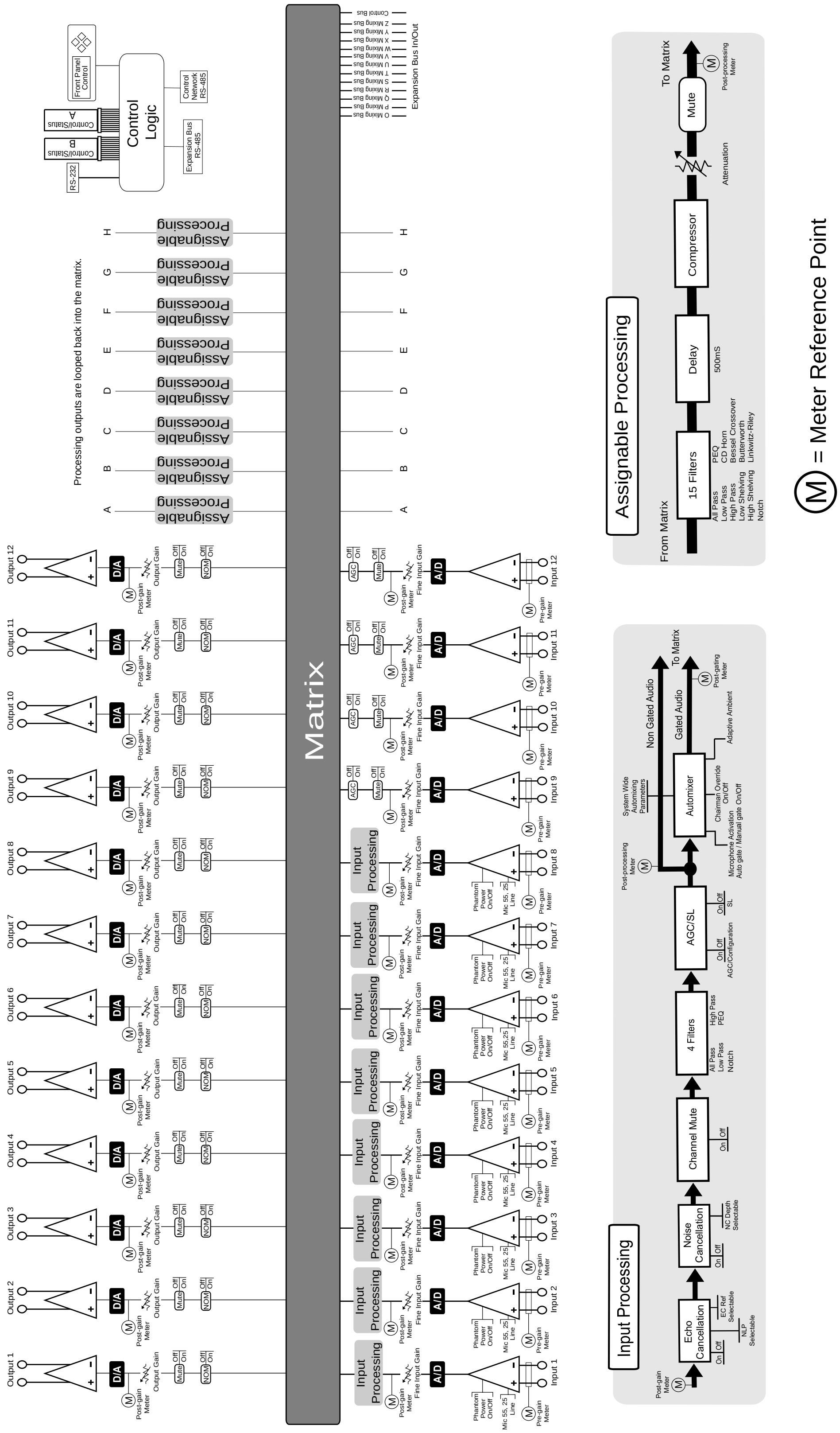


Figure 1.3. XAP 800 block diagram

XAP 400 Product Description

XAP 400 overview

The XAP 400 combines a highly advanced eight-by-eight digital matrix mixer that features Gentner® Distributed Echo Cancellation® (Gentner® D.E.C.), noise cancellation, and audio processing with a a single-line digital telephone hybrid. This combination provides high quality audio for a variety of audio and video conferencing applications—all in a single unit.

The XAP 400 hybrid uses digital signal processing (DSP) to separate the transmit and receive audio, eliminating distortion, weak signals, and feedback and continually filters low and high frequency noise to provide pure sound.

XAP 400 features

- Gentner D.E.C.—four microphone echo cancellers remove echo in even the most difficult environments.
- EC Reference summing enables the XAP 400 to reference various signals for an echo cancellation reference without requiring the use of a dedicated output.
- Noise cancellation reduces background noise such as that caused by fans or air conditioning.
- Four independent processing blocks, each with 15 filters, delay, and compressors, provide pinpoint audio configuration.
- Four mic/line inputs, four line inputs, and eight outputs.
- Enhanced expansion bus provides network capabilities. Eight XAP 400s or XAP 800s can be linked for up to 64 microphone inputs.
- Fully configurable matrix which allows you to route any input to any output or combination of outputs.
- Front panel control of mute and gain on inputs and outputs.
- Safety mute button on the toolbar that mutes all outputs if feedback occurs during the configuration process.
- ClearOne's 100% digital signal processing (DSP) technology ensures crystal-clear audio with the deepest, most reliable hybrid null.
- International configuration of telephone hybrid setting
- Built-in telephone interface with:
  - Telco noise cancellation
  - Touch-tone dialing capability (40 character dial string)
  - Full-time telco echo cancellation with 31 millisecond tail time
  - Selectable caller automatic level control (ALC)

- ~ Adjustable dial tone, DTMF attenuation
- ~ Continual adaptation to telephone line conditions
- ~ Digital anti-alias filter minimizes hum and Central Office switching noise
- ~ Analog telephone line compatibility

- 10W speaker amp output.
- Program and operate with a connected PC or any other type of serial remote control device via expansion bus or RS-232 port.

XAP 400 front panel

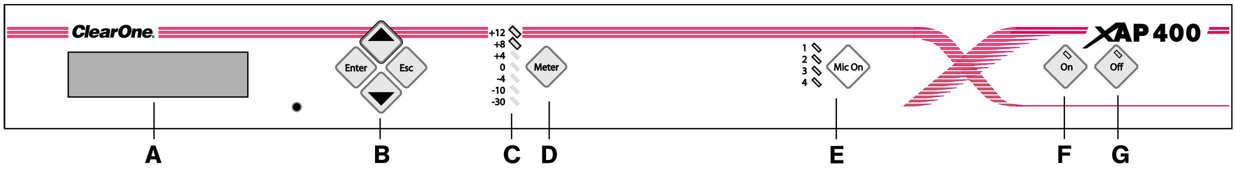


Figure 1.4. XAP 400 front-panel controls

A. **LCD.** The LCD is used for numeric display of audio levels, gain readouts, and limited set-up and programming functions.

B. **Enter/▲▼/ESC.** These buttons are used to navigate the XAP 400's menu system.

C. **LED Meter.** The LED bar meter is displays the audio level of a selected input, output, or processing channel of the XAP 400. The audio level of Output 8 is displayed by default.

D. **Meter.** The Meter button takes you directly to the Meter branch of the XAP 400's LCD programming tree.

E. **Mic On LED.** These LEDs indicate microphone gate status.

F. **On LED/button.** The bicolor LED on the button illuminates green when the hybrid is on. The On button connects the XAP 400 to the telephone line and automatically adapts the hybrid to the line. Pressing and holding the On button for more than a half-second while the hybrid is active will readapt the hybrid to the telephone line.

G. **Off LED/button.** The bicolor LED on the button illuminates red when the hybrid is off. The Off button disconnects the hybrid from the telephone line.

XAP 400 rear panel

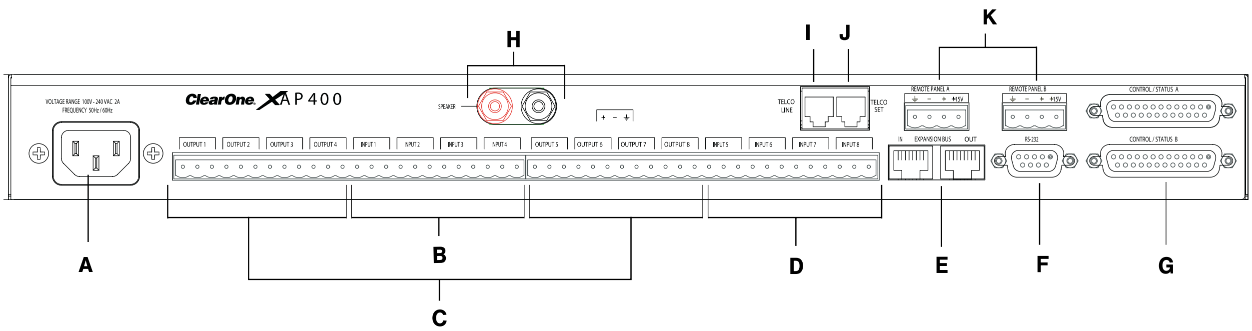


Figure 1.5. XAP 400 rear-panel connectors

A. **Power.** The AC power cord input is a IEC type connector allowing 100–240VAC, 50/60Hz.

B. **Inputs 1–4.** These Phoenix-type connection blocks are for mic and/or line level inputs.

C. **Outputs 1–8.** These Phoenix-type connection blocks are for line level outputs that may be configured for any combination of gated and non-gated inputs, as well as a mix of mic and line level inputs.

D. **Inputs 5–8.** These Phoenix-type connection blocks are for line level inputs.

E. **Expansion Bus In, Out.** This RJ-45 connector is used to connect XAP units. G-Ware is capable of accessing and controlling an expansion bus network of up to eight XAP 400/800/PSR1212 units and 16 XAP TH2 units, where the total number of microphone inputs does not exceed 64. The expansion bus supports a distance of up to 80 feet between each connected XAP 400/800 or PSR1212.

F. **RS-232.** This female DB9 serial port connects the XAP 400 to a PC, modem, or other custom remote controller.

G. **Control/Status Ports A and B.** These two female DB25 connectors are for general purpose input/output (GPIO) control of custom or unique control devices. The control devices access the command set for the XAP 400 and can be used for common functions such as volume control, muting, preset changes, room combining, etc.

H. **Speaker.** This is a 10W power amp connector. A 4–16Ω speaker can be directly connected to the XAP 400, eliminating the need for an external power amplifier.

I. **Telco Line.** This RJ-11 connector provides connection of a standard analog telephone line to the hybrid.

J. **Telco Set.** This RJ-11 connector allows connection to a standard telephone set. Tip and ring from the phone line are present at this connector when the hybrid is off. Tip and ring from the phone line are not present at this point when the hybrid is on.

K. **RS-485 Remote Panel A/B Port.** These four-pin Phoenix connector ports allow you to control the XAP 400 with the ClearOne Control Panel or XAP IR Remote. Power is supplied through the RS-485 ports to the Remote Control Panels from the XAP 400.

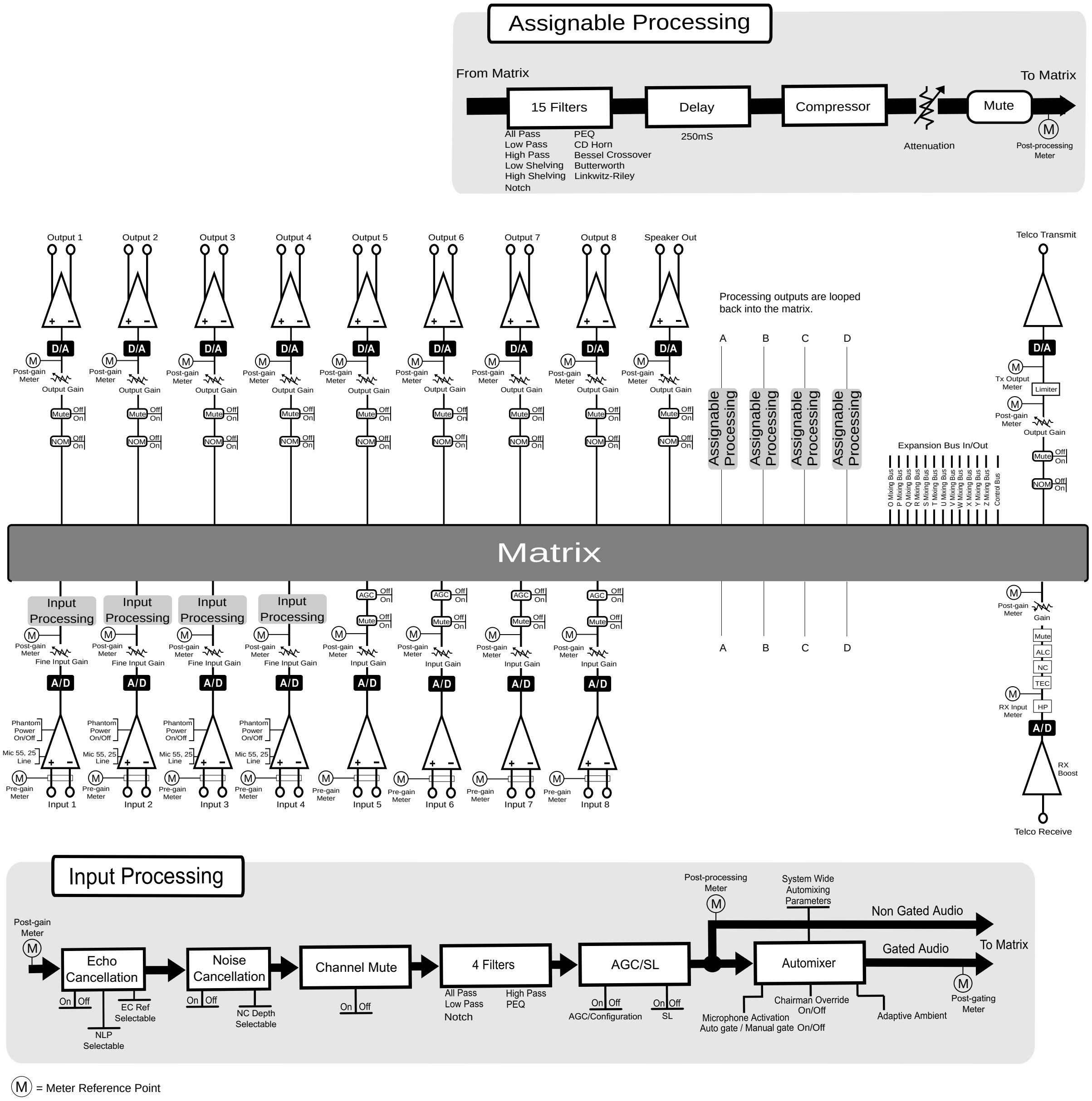


Figure 1.6. Meter Reference Point diagram

XAP TH2 Product Description

XAP TH2 overview

The XAP TH2 is a single-line digital hybrid which uses digital signal processing (DSP) to separate the transmit and receive audio, eliminating distortion, weak signals, and feedback. It continually filters low and high frequency noise to provide pure sound.

The XAP TH2 is designed to function as a stand-alone telephone hybrid or as an accessory to the XAP 800 (echo cancelling, audio processing, microphone mixing matrix), enabling you to add remote callers to your audio conferences.

XAP TH2 features

- ClearOne's 100% digital signal processing (DSP) technology ensures crystal-clear audio with the deepest, most reliable hybrid null.
- Balanced line-level input and output.
- Touch-tone dialing capability (40 character dial string).
- Full-time telco echo cancellation with 31 millisecond tail time.
- Conference up to 16 callers (with 16 XAP TH2s) within a XAP 800 system.
- Adjustable audible connect and disconnect tones.
- Selectable caller automatic level control (ALC).
- Adjustable dial tone, DTMF attenuation.
- Simultaneous two-wire/four-wire operation.
- Continual adaptation to telephone line conditions.
- Digital anti-alias filter to minimize hum and Central Office switching noise.
- Compatible with analog telephone lines.
- Program and operate with a connected PC or any other type of serial remote control device via the Expansion Bus or RS-232 port.

XAP TH2 front panel

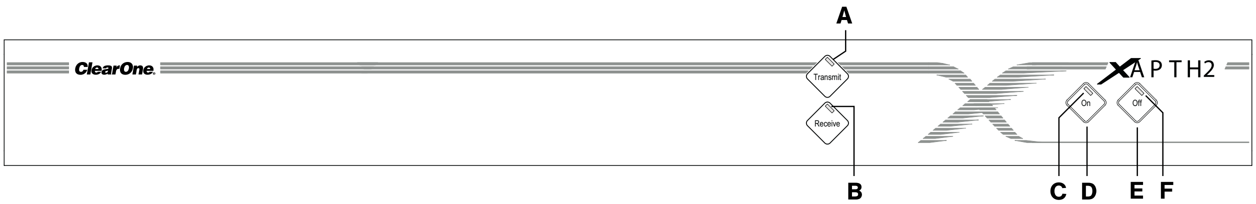


Figure 1.7. XAP TH2 front-panel controls

A. **Transmit LED.** This bicolor LED indicates the audio levels being transmitted from the room to the telephone line.

B. **Receive LED.** This bicolor LED indicates the audio level the room is receiving from the telephone line.

C. **On LED.** This LED indicates the hybrid's On state. The LED will illuminate green when the hybrid is on.

D. **On.** The On button connects the XAP TH2 to the telephone line and automatically adapts the hybrid to the line.

E. **Off.** The Off button disconnects the hybrid from the telephone line and mutes all audio.

F. **Off LED.** This LED indicates the hybrid's Off state. The LED will illuminate red when the hybrid is off.

XAP TH2 rear panel

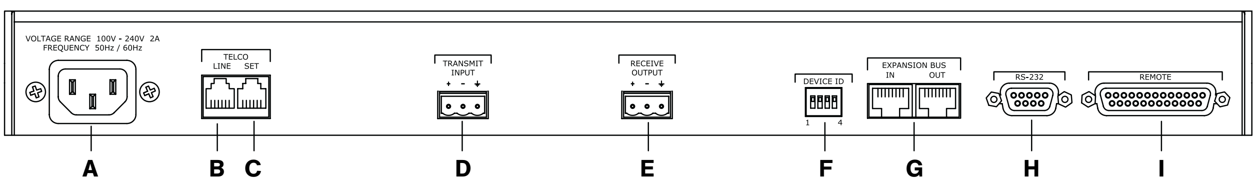


Figure 1.8. XAP TH2 rear-panel connectors

A. **Power.** The AC power cord input is a IEC type connector allowing 100–240VAC, 50/60Hz.

B. **Telco Line.** This RJ-11 connector provides connection of a standard analog telephone line to the hybrid.

C. **Telco Set.** This RJ-11 connector allows connection to a standard telephone set.

D. **Transmit Input.** This Phoenix connection provides a non-gated electronically balanced line level input. The nominal input level is 0dBu. This line input is mutable. The default setting is off (not muted).

E. **Receive Output.** This Phoenix connection provides a balanced line level output. The nominal output level is 0 dBu. The output adjusts for line imbalances and maintains a constant output level. This line output is mutable. The default setting is off (not muted).

F. **Device ID.** This four-position DIP switch is used to assign a device ID number to the XAP TH2.

G. **Expansion Bus In, Out.** This RJ-45 connector is used to connect the XAP TH2 to the XAP 800 for control. Up to eight XAP 800/XAP400/PSR1212 units and 16 XAP TH2 units can be connected together.

H. **RS-232.** This female DB9 serial port is for interconnection between the XAP TH2 and a PC, modem, or other custom remote controller.

I. **Remote.** This DB25 connector provides control and status of the XAP TH2 and unbalanced audio.

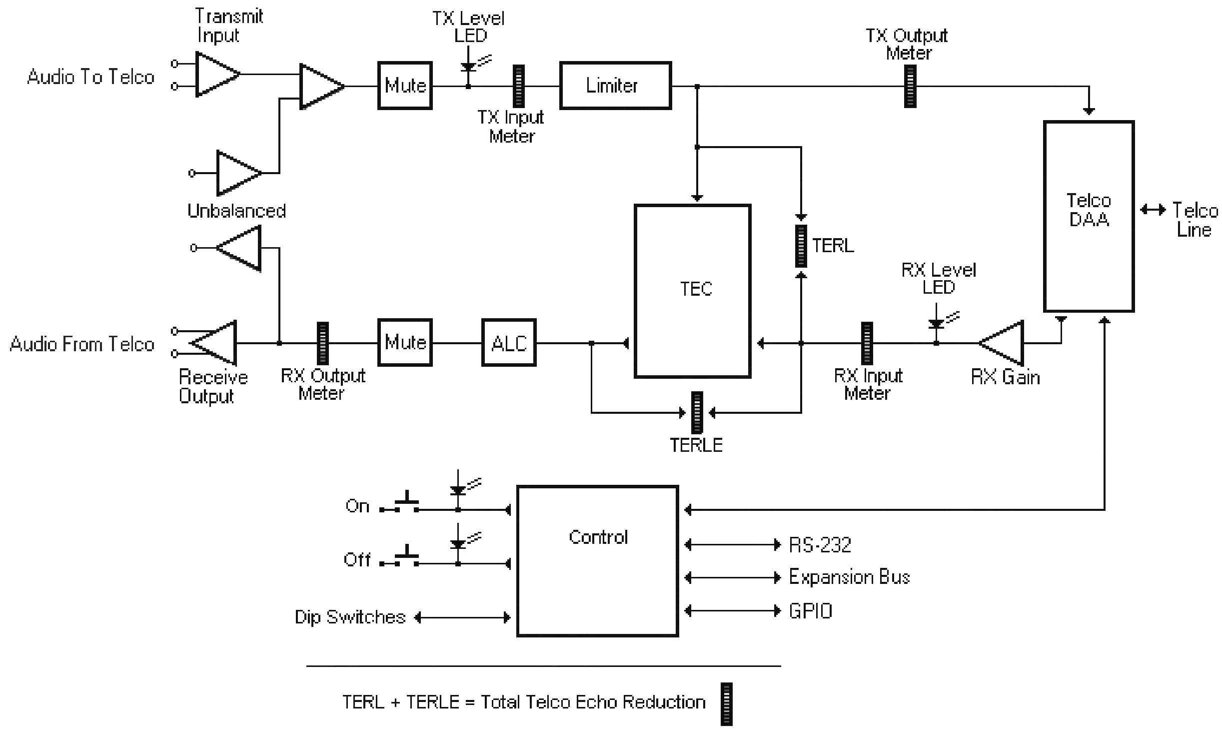


Figure 1.9. Meter Reference Point diagram

G-Ware Description

ClearOne's G-Ware software provides an easy interface for configuring and controlling your XAP products. While some configuration can be done using the front panel LCD menus, G-Ware is required to complete the custom configuration of your audio conferencing system.

G-Ware has three modes: Configuration, Preset, and Macro Recorder. Configuration is used to configure the unit and is the default mode. The Preset and Macro modes are used to create presets and macros for system control. You can switch between modes by clicking on the corresponding toolbar button. The current mode is displayed on the status bar.

Figure 1.10. Configuration, Preset, and Macro toolbar buttons.

There are two main configuration screens, the Flow Screen and the Matrix Screen. All unit configuration and audio routing is accessed through these screens.

Flow Screen

The G-Ware Flow Screen is the main access window for G-Ware's features and functions. Using the menus and toolbar at the top of the screen, you can access general configuration windows. Unit specific configurations are accessed through the buttons and labels on the Flow Screen itself. If you have multiple units, click on the unit icon in the Site pane to access that particular unit's Flow Screen.

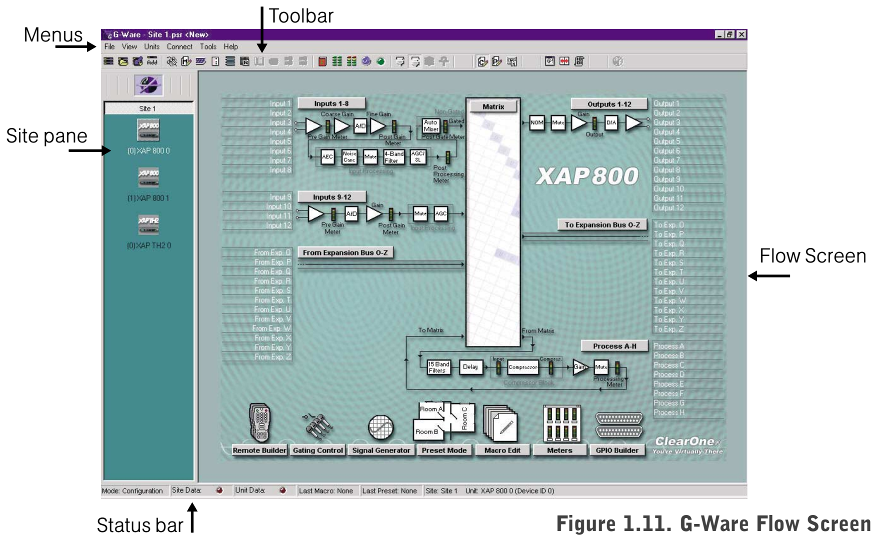


Figure 1.11. G-Ware Flow Screen

Matrix Screen

The Matrix Screen is used to route audio to one or more destinations (outputs, expansion bus, or processing). Any number of connections to outputs can be made in the matrix. The Matrix Screen can also be used to access the Input, Output, Telco, and Processing configuration windows by clicking on the buttons along the top and left side of the routing matrix.

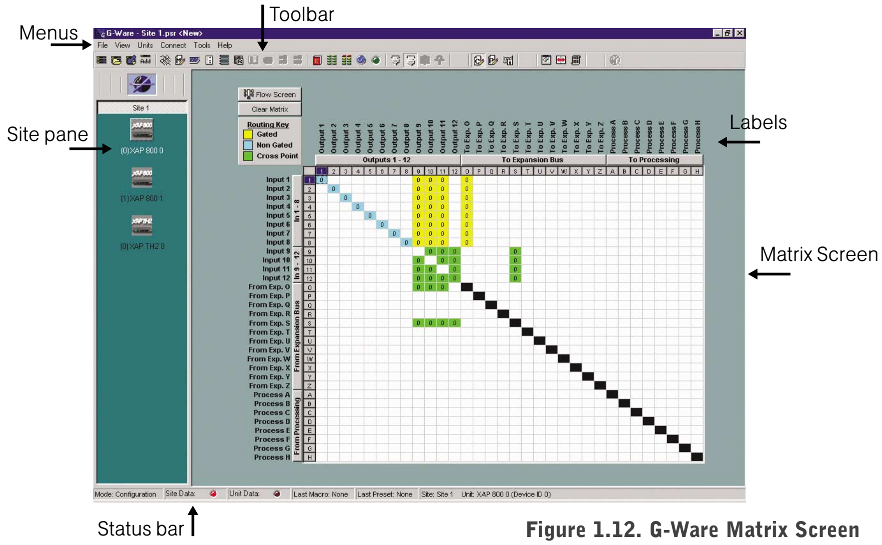


Figure 1.12. G-Ware Matrix Screen

Equipment Placement

Each XAP unit is designed for mounting in a 19" equipment rack. Do not block any of the ventilation holes. With a desktop kit, it can be modified for tabletop placement.

Environmental requirements

XAP units can safely operate in temperature environments between 32°–110° F/0°–38° C.

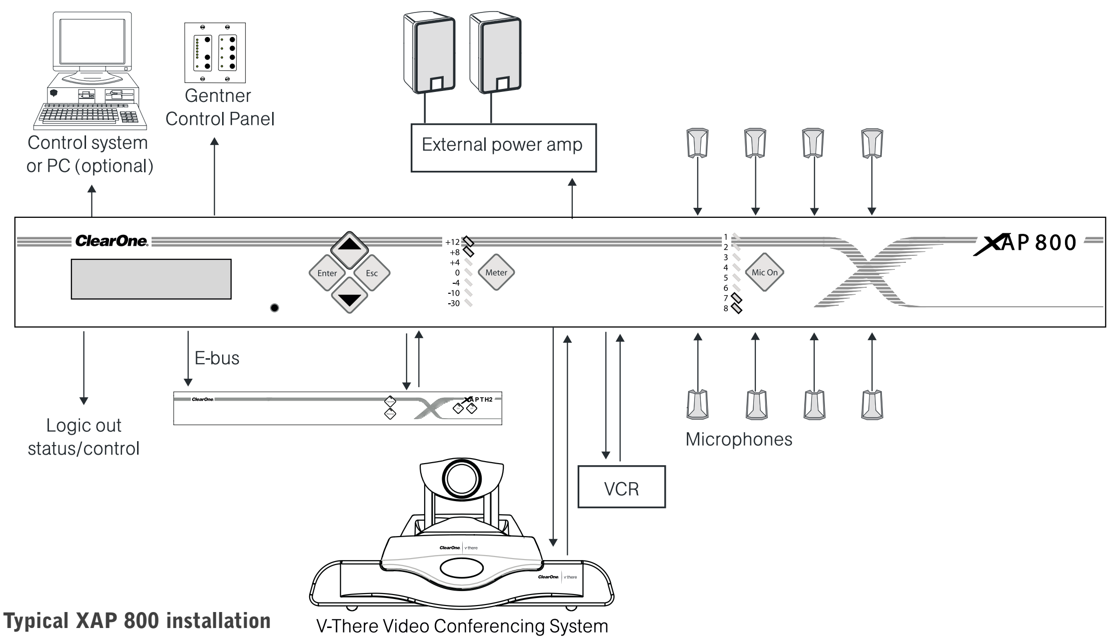


Figure 1.13. Typical XAP 800 installation (simplified for illustrative purposes)

CHAPTER 2: Echo and Noise Cancellation

Echo Cancellation

Acoustic echo is a significant challenge to overcome in virtually any teleconferencing environment. The effects of acoustic echo can destroy a teleconference because it impairs participants' abilities to understand and communicate.

Acoustic echo is created when microphones pick up loudspeaker audio and return it to the originating teleconference site. This echo will cause a conference participant to stop speaking while trying to listen to the echo.

One way to remove echo from your teleconference is to use an acoustic echo canceller. An acoustic echo canceller samples audio coming in from the remote site and prevents this audio from being sent back to the originating site. To be most effective, each site should utilize an echo canceller.

There are several factors that contribute to poor echo cancellation. These include:

- Poor room acoustics
- High reverberation
- High noise
- Rapidly changing acoustical environment
- Wireless or other moving microphones
- Poor microphone/speaker placement
- Automatic mic mixers that are not properly configured to work with an echo canceller
- Other audio devices such as audio processors and user gain controls that change the acoustic gain to which the echo canceller must adapt.

Figure 2.1 on the following page shows how a single echo canceller attempts to cancel echo in a room. Audio from the distant room is sampled and used as a reference for the echo canceller. When far-end audio is picked up by the microphone (acoustic echo), the acoustic echo canceller senses the echo and builds an adaptive filter that eliminates the echo.

✍ Visit ClearOne Technology Lab at www.clearone.com to learn more about echo cancellation and to hear the Gentner Distributed Echo Cancellation difference.

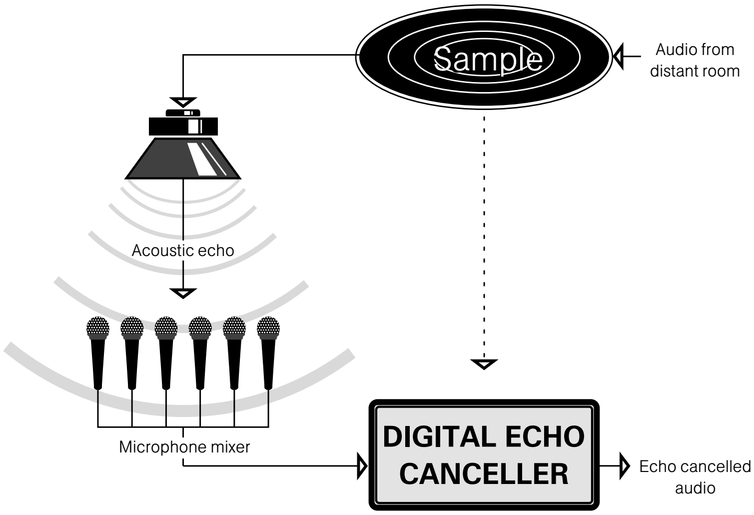


Figure 2.1. Single echo canceller

The XAP 800's Distributed Echo Cancellation places an echo canceller on each mic input for dramatically improved echo cancellation (see Figure 2.2).

In this example, audio from a distant room is sampled. This audio is a reference for each echo canceller on every mic. When sampled audio (acoustic echo) is detected by the echo canceller, the echo is eliminated.

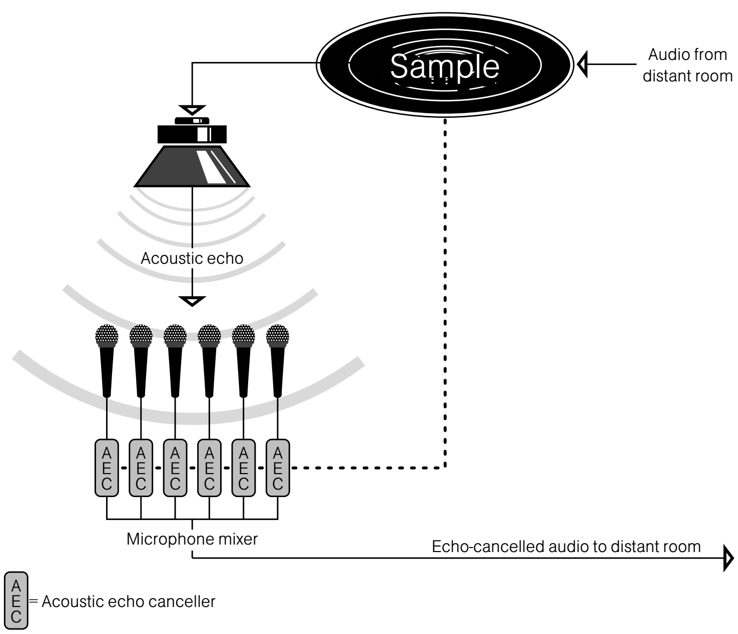


Figure 2.2. ClearOne's Distributed Echo Cancellation

The XAP advantage

XAP audio conferencing systems overcome acoustic echo cancellation challenges through the use of ClearOne's proprietary Distributed Echo Cancellation (Gentner D.E.C.) technology. Unlike older systems, which use a single echo canceller across all audio sources in the system, Gentner D.E.C. dedicates an echo canceller to each mic input. The D.E.C. system is far more effective in canceling echo, resulting in clearer, more accurate echo cancellation. The D.E.C. system can also track changes in the room environment more effectively, keeping the audio quality at the highest level.

The advantages of Gentner D.E.C. include:

- Significantly better echo cancellation in a wide variety of acoustical environments
- Plug and play echo cancellation
- Faster convergence time
- Better full duplex
- Reduced noise and suppression
- Increased gain
- Higher tolerance to room and network audio level changes

Instead of a single echo canceller covering the entire room, an echo canceller is put on every acoustic (mic) input. In a room of eight microphones, a D.E.C. system would have eight echo cancellers. Each echo canceller must work only on one acoustic reference. Obviously, such an echo canceller has a far easier time canceling echo than a single echo canceller with eight acoustic references. In addition, when compensation techniques are required for times when echo cannot be fully cancelled (suppression, center clipping, etc.), the compensation effects are heard only on the single microphone channel, rather than the entire mixed audio source. This greatly improves full duplex, noise control, and compensating audio level reduction.

Noise Cancellation

Today's hi-tech microphones are becoming more sensitive, which means the chance that simple sounds will be detected and amplified is much higher. Some of these sounds include:

- Heating, ventilation, and air conditioning (HVAC)
- Fluorescent light ballasts, generators, power cords, and other electrical items that generate low-level (60Hz) hum in amplification equipment
- Portable devices such as laptops and overhead projectors have fans that can transmit on frequencies that are occasionally detected and amplified

Elimination of noise

Since the term "noise" covers a variety of unwanted audio generated by many different sources, there is also a variety of ways to cancel the noise.

Most noise can be eliminated by:

- Acoustical treatments in and around noise sources
- Conducting microphone placement tests
- Reducing the amount of reflective surfacing
- Identifying and reducing or eliminating hum
- Using quality, shielded cable and connectors
- Using a noise cancellation device, such as that found in the XAP

ClearOne's XAP systems feature a revolutionary new algorithm that actively eliminates background noise within the vocal range on each mic input. Unlike other noise cancelling devices, the XAP does not simply gate audio off when a mic input drops below a specified level. ClearOne's noise cancellation actively separates and attenuates ambient noise from the speech signal, leaving speech audio virtually untouched. It effectively reduces noise between 20Hz and 20kHz.

ClearOne's noise cancellation also ensures that your entire system functions more efficiently. Noise not only interferes with human communication, it unnecessarily hampers all sound system components (i.e., DSPs, amplifier power, loudspeaker bandwidth), thereby significantly decreasing a system's power and responsiveness. By reducing signals unassociated with human speech, the XAP ensures the entire system sounds clearer and is much more efficient and responsive.

Noise cancellation vs. echo cancellation

Since the job of any sound system is to amplify audio signals, all noise is picked up and amplified. Even barely audible signals may become very loud once they are picked up by a sound system. However, unlike echo cancellation, noise cancellation is much more challenging because it lacks a reference signal. Despite these challenges, ClearOne's revolutionary noise canceller discerns and actively tracks ambient noise and eliminates it from the final mix.

The XAP's noise canceller is an excellent complement to the acoustic echo canceller. This is because it reduces ambient noise, which can cause the echo canceller to deviate from an optimal echo cancellation solution. By reducing much of the ambient noise, it ensures the echo canceller has the purest, most accurate room reference by which to cancel echo.

CHAPTER 3: Processing Blocks

The XAP 800 features the power of eight processing blocks, which are dedicated signal processing resources that can be applied to any mic or line input routed to it. Within each processing block, compression, delay, gain adjustments, and up to 15 different filters may be applied to any mic/line input or combination or inputs routed to the block. Each input also has an additional four assignable filters for shaping the input's audio response.

Filters

Filters can be used to isolate and attenuate specific frequencies. For example, you have a conferencing scenario in which mics connected to Inputs 1 and 4 are having feedback problems. The problem frequencies could be attenuated in each mic input, or both mics could be assigned to processing block A, where problem frequencies are attenuated. In the future, other microphones can also be routed to the same processing block and will therefore receive the same feedback cancellation—without any additional adjustments. Microphones not routed to processing block A would be unaffected by these filters.

Feedback

Perhaps the most common (and most annoying) problem for professional sound engineers is feedback. Feedback is often a high, piercing tone caused when mics pick up a particularly resonant frequency and amplify it through the sound system, where it is picked up again by the microphones—this time louder—and the process is repeated.

In addition to being painful to hear, feedback can also cause amplifiers to clip, and can destroy drivers and loudspeaker components. While most feedback can be controlled by proper microphone placement, resonant frequencies can be attenuated using filters such as parametric equalizers.

Low frequency feedback isn't necessarily audible, but it decreases the system's power and responsiveness. By attenuating the low-end frequencies (low cut/high pass) not needed for vocals, the XAP system operates more efficiently because it doesn't need to allocate resources to produce low frequencies throughout the system.

XAP 800 feedback control system

The XAP 800 includes a number of highly customizable filters that are excellent tools for feedback control. These filters boost or attenuate certain audio ranges, compensate for poor acoustical environments, and generally fine-tune your sound system. Some filters, such as pass filters, allow you to select a range of audio frequencies above or below a given point and attenuate it, while others allow you to attenuate specific frequencies. Filters are also excellent for use in cancelling out frequency ranges that cannot be transmitted through telephone lines, thereby freeing up the system's resources.

The XAP 800 and XAP 400 feature the following filters:

- **All Pass.** A filter that provides only phase shift or phase delay without significantly changing the magnitude of the signal. Useful in custom crossovers designed to compensate for loudspeaker driver deficiencies.
- **CD Horn Equalizer.** A constant directivity horn driver has an inherent 6dB/octave high frequency roll-off. This device produces a 6dB/octave boost to compensate. The CD horn device is implemented using a high shelving filter. The level control is fixed at +12dB, and is hidden.
- **High Pass.** Allows frequencies above a designated frequency to pass while attenuating those below it. Perfect for attenuating low, rumbling noises like those captured by a microphone when placed on a stage, table, or in a microphone stand.
- **High Shelving.** Boosts or attenuates frequencies above a designated frequency while leaving those below it unaffected. The transition between the spectrum above and below the designated frequency occurs at a fixed 6dB/octave rate. When boosted, it enhances the higher, more intelligible aspects of the vocal range. When cut, it is excellent for avoiding resonant frequencies (feedback) and ranges of extraneous sibilance (like a constant "ssssssss" noise).
- **Low Pass.** Allows frequencies below a designated frequency to pass while attenuating those above it. Useful for reducing overall sibilance and avoiding shrill resonant frequencies (feedback).
- **Low Shelving.** Boosts or attenuates frequencies below a designated frequency, leaving those above it unaffected. The transition between the spectrum above and below the designated frequency occurs at a fixed 6dB/octave rate. Excellent for enhancing the low-end range of a signal.
- **Notch Filter.** A band-stop filter that can remove a select range of frequencies. Commonly used for removing specific resonant frequencies from a system.
- **Parametric Equalizer.** A multi-band variable equalizer that allows the user to define the amplitude of the filter, shift the center frequency of the filter, and control how wide the range is to which the equalizer is applied. Excellent for general tone shaping or feedback removal.

Filters for the XAP 800 and XAP 400 are configured in G-Ware software.

Crossovers

The XAP 800/400 also features a crossover function. The crossover combines high-pass and low-pass filters that divide a full-range signal into separate frequency ranges. These ranges can then be sent to amplifiers and loudspeakers optimized for producing those respective frequency ranges. Band-pass filters can be designed by overlapping high-pass and low-pass filters.

For example, the bottom end of a frequency range might be defined as 400Hz and lower. This signal might then be heavily compressed and sent to bass bins and woofers powered by amplifiers. The midrange (say, 400Hz to 5kHz) can be sent to full-range loudspeakers. The high end (say, 5kHz and above) can be directed to smaller amplifiers and associated tweeters, piezos, horns, etc.

The XAP audio conferencing system includes three types of crossovers: Butterworth, Bessel, and Linkwitz-Riley. Each of these is characterized by the steepness of their roll-off slopes (the rate of attenuation outside their passbands). Crossovers in the XAP are created by assigning the appropriate type of high-pass filter in a processing block with the complementary low-pass filter in a separate processing block.

- **Bessel Crossover.** Utilizes a low-pass filter design characterized by having a linear phase response (or maximally flat phase response), but also a monotonic decreasing passband amplitude response (which means it starts rolling off at DC and continues throughout the passband). Linear phase response (e.g., a linear plot of phase shift vs. frequency producing a straight line) results in constant time delay (all frequencies within the passband are delayed the same amount). Consequently, the value of linear phase that reproduces a near-perfect step response (there is no overshoot or ringing resulting from a sudden transition between signal levels). The drawback is a sluggish roll-off rate. For example, for the same circuit complexity, the response for a Butterworth crossover rolls off nearly three times as rapidly.
- **Butterworth Crossover (1 pole).** A type of crossover circuit low-pass filter design characterized by having a maximally flat magnitude response (i.e., no amplitude ripple in the passband).
- **Linkwitz-Riley (LR-4) Crossover (2 poles).** The 4th-order (24dB/octave slope) Linkwitz-Riley (LR-4) design represents a vast improvement over the previous 3rd-order (18dB/octave) Butterworth standard. It consists of a cascaded 2nd-order Butterworth low-pass filter, and is considered the de facto standard for professional active audio crossovers.

XAP also has an intuitive filter display, which shows the aggregate filter response in a particular processing block. This display can also overlay the filter responses of other processing blocks, which is useful in designing active crossovers. The aggregate phase response of the processing block can also be shown superimposed on the filter response curve.

CHAPTER 4: Inputs and Outputs

Level Control

The XAP 800 has 12 inputs, consisting of eight mic/line inputs and four line inputs, and 12 line outputs. The XAP 400 has eight inputs, consisting of four mic/line inputs and four line inputs, and eight line outputs.

All inputs and outputs are actively balanced. Mic inputs have 5kΩ of terminating impedance while line level inputs provide >10kΩ of termination. Outputs provide a source impedance of 50Ω. All levels are referenced to a 0dBu level.

Input and output level control is executed in the digital domain. As a result, input levels should never exceed +20dBu. The unit will deliver a maximum output level of +20dBm. XAP utilizes 24-bit A/Ds and D/As while sampling at a 48kHz rate. This results in a system-wide dynamic range of 100dB and a pass band from 20Hz to 20kHz. Input and output levels can be monitored in real time on the front-panel LED display and through the RS-232 serial port. The LCD display and RS-232 port provide precise numeric readouts indicating level. This allows extremely precise level calibration. Additionally, while monitoring numeric dBu audio levels, input and output gains can be adjusted for optimum audio performance.

Mic inputs

Balanced audio is input at the rear panel Phoenix connector. Mic or line level is selected and phantom power is provided (if required). The XAP then converts this audio from analog to digital for processing by the DSP engine. Once converted to digital, audio is level controlled. This function, along with all other input and output controls, can be adjusted via the RS-232 port and/or the control pins on the control/status connector. This provides for real-time audio volume control, muting, etc. Minimum and maximum levels can also be set to limit the range of gain to suit audio requirements.

The acoustic echo canceller/speech leveler is the first option in the audio signal flow. Here, you set the echo cancellation reference, non-linear processing (optional), and meters reflecting echo cancellation activity. You can also enable the noise canceller and depth of cancellation.

Next, four configurable filters can each be set individually or in any combination. Available filter types are: all-pass, low-pass, high-pass, parametric equalizer (PEQ), or notch. Each may be activated to equalize different microphones to sound similar, filter out unwanted hum, etc. You can increase or decrease each band up to 15dB, in increments of .5dB on each input.

The next option is the automatic gain control (AGC) and speech leveler. The purpose of the AGC is to automatically increase gain when the level is too low and decrease gain when it is too high. AGC is provided at all inputs and should be activated for microphones or line inputs that experience audio level fluctuation.

The speech leveler targets speech audio and equalizes the audio levels of someone who is speaking close to a mic and someone who is speaking from farther away. For example, if audio coming from a video codec fluctuates (depending on the connection at the other end), the AGC will compensate for these differences.

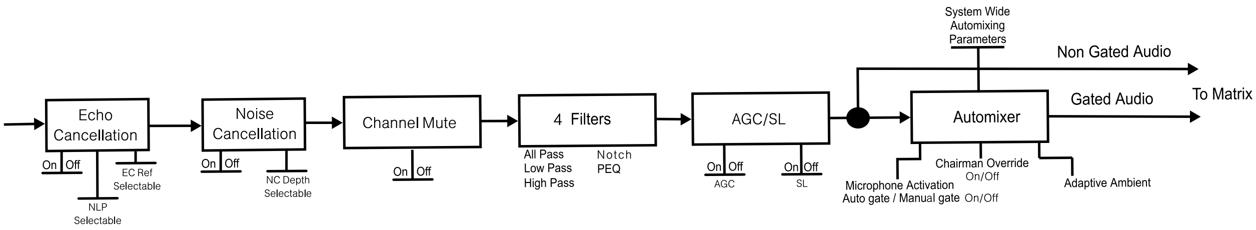


Figure 4.1. Mic input parameters

After the channel mute stage, non-gated audio is applied to the routing matrix for outputs that need direct audio. The final stage (automixing) determines how the audio is directed into the post-gating input to the routing matrix. Each input can be set for a variety of automixing functions, including activation settings, chairman mic, and adaptive ambient mode. The functions determine when, how, and why an individual microphone will gate on or off.

- **Microphone Activation.** There are two modes of mic activation that can be selected on a per-input basis: auto-gate and manual gate on/off. In auto-gate mode, the input channel is voice activated, based on the programmed automixing system parameters. In manual gate mode, the mic is activated by manually switching it on or off and allowing the input to contribute to automixing parameters.

- **Chairman Override (On or Off).** Each gated input may be selected as a chairman override input. This feature adds this input to the chairman override group and, when gated on, gates off inputs with in the same group that are not chairman override enabled.

- **Adaptive Ambient (On or Off).** In the ON mode, the ambient level used to calculate gating is based on the room's actual noise floor, integrated over time, as measured by the input in the room. In the off mode, the manual ambient level is set by the integrator, and will be used to calculate gating.

Figure 4.2 shows line level inputs. These line level inputs can be level controlled, muted, and gain controlled through G-Ware. All of these functions operate identically to the mic inputs.

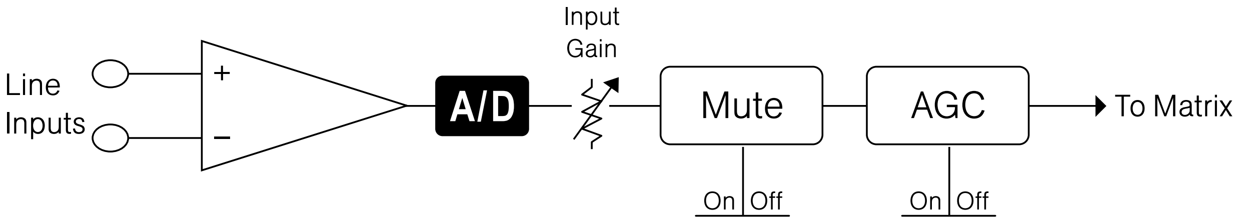


Figure 4.2. Line level inputs

Outputs

All line outputs are identical, as shown in Figure 4.3. Three functions are associated with each output: gain control, mute, and NOM (number of open mics). Gain control allows you to set the output level. The min/max controls allow you to set minimum and maximum gain levels that the user will be limited to. The mute function essentially turns the volume off. All of these functions can be controlled via the RS-232 port or the control/status connector. An example would be if you want to control the volume of the speakers—you could use two control pins on the control/status connector for volume up and volume down. Another pin could be used for mute.

Activation of NOM places this output **only** in a mode where, as more microphones routed to this output are gated on (either by auto gate or manual gate), the total overall output gain will remain constant. This reduces the possibility of feedback occurring.

A feature of the XAP is its ability to provide NOM at every output. Most automixers have a single master NOM output. NOM is used to maintain a constant acoustic gain in the room, permitting the system to optimize its gain before feedback status. This is most useful in sound reinforcement applications.

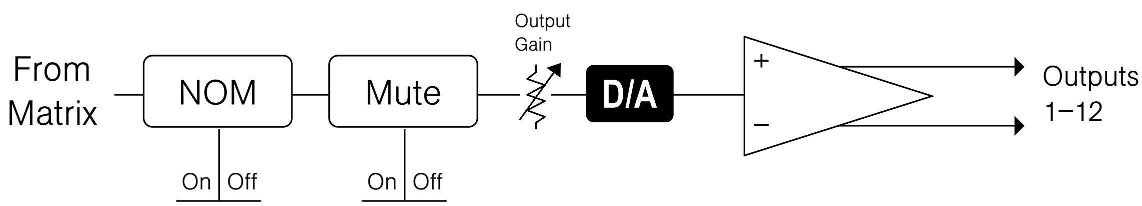


Figure 4.3. Outputs from Matrix

CHAPTER 5: Automatic Mic Mixing

Intelligible, Reliable Audio

Conferencing systems are in constant use in conference rooms, boardrooms, training rooms, and many other applications. Systems that produce intelligible and reliable audio are key to facilitating effective communication. Quality conferencing systems meet the following objectives:

- The audio must be transparent. Users should not have to think about the audio.
- The audio must not fatigue the users. Distorted, noisy audio will cause users to break off discussions before a natural conclusion occurs. It will also fatigue the users, producing a less-than-effective outcome.
- Since 10% of our population is hearing impaired, the audio system must be capable of producing effective results for all users.
- The audio system must be reliable.

Automatic microphone mixing is a key part of producing highly intelligible and reliable conferencing audio. An automatic microphone mixer, in conjunction with directional microphones, will reduce reverberation and noise—the two major culprits in making voice communications difficult to understand.

In Figure 5.1, direct audio from a person's voice is picked up by several microphones connected to a microphone mixer that has all microphones on at all times. Direct and reflected audio (reverberation) is picked up by all the microphones. In addition, the reverberated audio will have a variety of delays, depending on how far it has traveled in the room and how many surfaces reflected it. When this happens in an actual audio setting, we have a difficult time understanding the audio.

We have all experienced trying to speak in a room that has a lot of reverberation—it's difficult. When people hear reverberated audio, their initial response is to turn up the volume. This does not help make the audio more understandable; in fact, in audio room systems, turning up the volume will almost always degrade the performance of the entire system. In addition, with more microphones on, more noise is picked up by the system. Clearly, increased noise and reverberation hurts audio intelligibility and increases listener fatigue.

There are several strategies that can be used to reduce reverberation and noise:

- Keep microphones close to the participants.
- Activate only those microphones where voice audio is present.
- Use directional microphones.
- Acoustically treat the room to reduce reverberation and noise.
- Eliminate or reduce the source of noise.

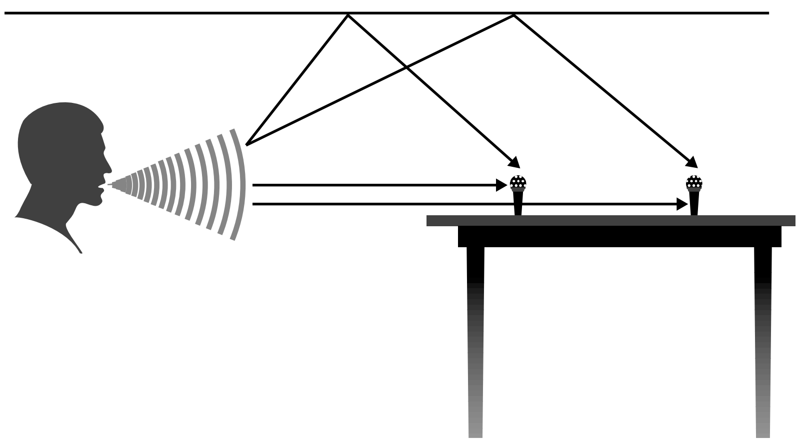


Figure 5.1. Microphones pick up direct and reflected audio

The XAP 800/400 was designed to implement automatic microphone mixing that increases audio intelligibility by reducing overall multiple microphone pickup of reverberation and noise. Unlike most automixers, the XAP implements its mixing function completely in the digital domain. This greatly increases precision in making automixing decisions.

All audio is routed through the XAP (both microphone and speaker audio), which means the XAP can more accurately make microphone activation decisions.

For example, when PA Adaptive is enabled, audio from another source (such as conference audio from another room) is amplified through the speakers in the room. Typically, an automixer would activate at least one microphone, as if that audio were a voice in the room. This false activation will not occur with the XAP (see Figure 5.2) because the unit can determine that this audio is coming from the loudspeaker.

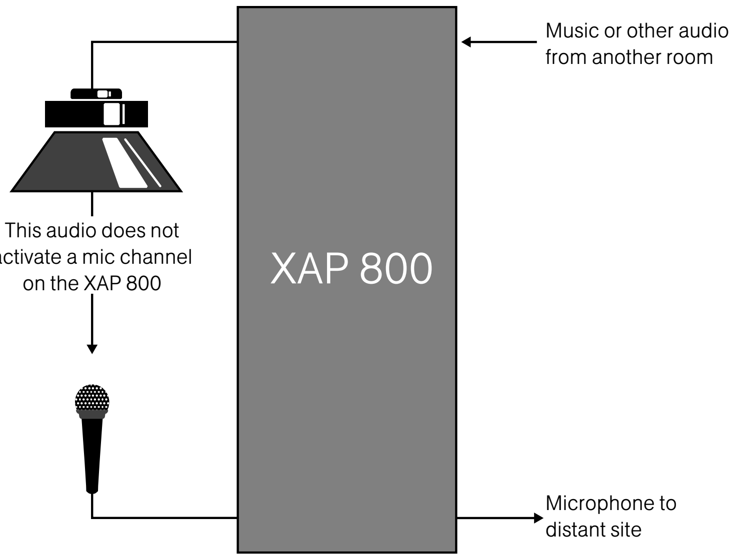


Figure 5.2. Microphone activation

Microphone Parameters and Modes

A XAP system has a variety of automixing functions that are implemented on both a per-channel basis and across the entire automatic mixer. These functions are described on the following pages. Each XAP 800/400 can have up to four separate automatic mixers working independently within a single unit, and up to four independent global mixers across multiple units. In addition, more microphone channels can be added by linking XAP units via the expansion bus, the digital network bus. Unlike other "expandable" automatic microphone mixers, the XAP works as a single unit for up to eight units networked together. Expanded analog automixers can offer only limited functionality such as NOM (number of open microphones). Multiple XAP units can operate as a single unit because all functions are implemented digitally and all units are connected together using the high-speed digital network bus (expansion bus), which passes both audio and control information. See Figure 5.3.

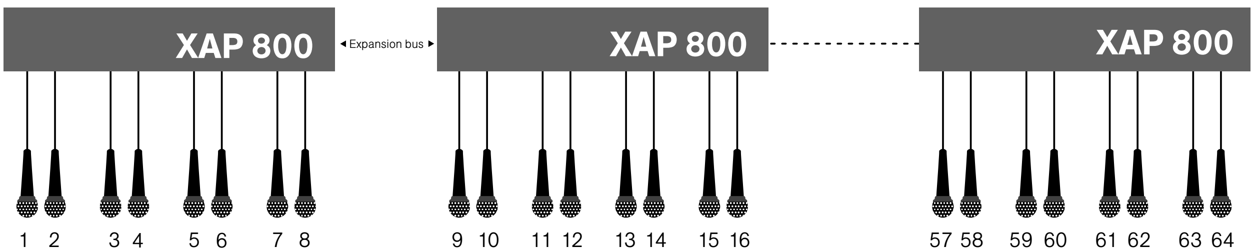
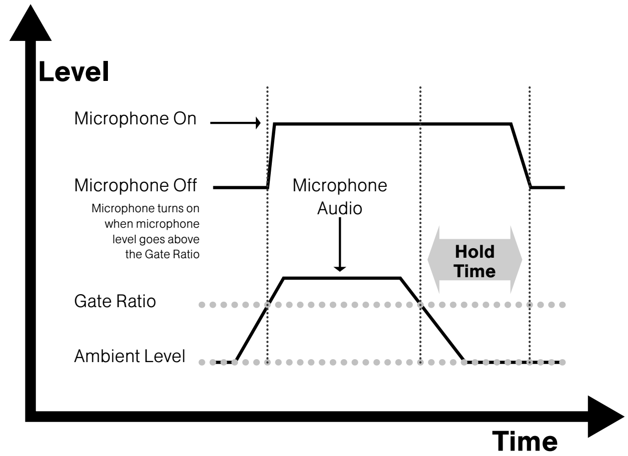


Figure 5.3. Expansion bus control of 64 mics

The following parameters and modes are used on the XAP to provide high precision and reliability in microphone mixing.

- **Mixer Mode.** The XAP 800 can be set in two different mixer modes to accommodate a variety of installation needs: master or slave. When placed in the master mode, the unit acts as a stand-alone eight channel automatic microphone mixer. The master mode is also used for the master unit in a multiple-unit configuration. Slave mode is used for slave units in a multiple-unit configuration.
- **Chairman Override.** This provides gating priority for all microphones selected for the chairman override group. When a mic in this group gates on, all microphones within this group that are not chairman override enabled gate off.
- **PA Adaptive (On or Off).** The problem: Speaker audio gates on microphones when it shouldn't. The solution: PA adaptive knows when speaker channels are activated and prevents the speakers from gating the mics on. The reason the XAP 800 can accurately determine when loudspeaker audio is present is because audio goes through the XAP 800 a few milliseconds before it reaches the microphone.
- **Maximum Number of Microphones On (1–8 or Off).** This mode allows you to program how many microphones (maximum) can be activated simultaneously. Generally, there isn't a need for more than two or three people to speak at the same time.

Figure 5.4. Automixing gate functions



- **First Mic Priority Mode (On or Off).** This feature increases the audio level required to gate on additional microphones after the first mic gates on. This helps ensure that only one mic gates on when a person speaks.

- **Last Mic On/Mic1-8/Off.** Last Mic On leaves the last activated mic on until a new one is activated. Mic 1–8 mode reverts back to the mic you've selected for Mic 1–8 On when all other mics gate off. These features are useful to ensure the audio never goes completely away. Without it, you might even think that you have lost connection to the other room. You can set this parameter to Off, which disables this function.

The following are depicted in Figure 5.4:

- **Gate Ratio Adjust (0 to 50dB).** This specifies how much louder the audio level must be above the ambient level to gate on. If, for example, the gate threshold is set at 35dB, it will take 35dB plus the gate ratio the room to activate the microphone. The ambient audio level can be specified or the adaptive ambient mode can be turned on. In this case, the ambient room level changes or adapts as the noise floor changes.

- **Off Attenuation (0 to 50dB).** This sets how much a gated input is attenuated when it is not on. This feature help prevent the first consonant of words from being lost when the microphone gates back on.

- **Hold Time (.1 to 8.0 seconds).** This programs how long the mic stays gated on after audio is no longer present and keeps the mic from gating off to pauses in speaking.

- **Decay Rate (slow, medium, fast).** This programs how quickly the audio level attenuates to the Off Attenuation level after the hold time has expired.

- **Manual Ambient Level (0 to -80dB).** This setting is relevant only if the adaptive ambient mode is disabled on the individual gated inputs. This ambient level is then used in conjunction with the gate threshold to determine whether or not the mic should turn on.

Automatic microphone mixing is a key part of the XAP 800/400 solution set. Because all decisions regarding automixing are made by the same digital engine, better decisions in automixing can be made.

Parameter	Effect	Range	Description
Mixer mode	System-wide	Master, slave	Selects mixer mode of operation.
Microphone Activation	Inputs 1–8	Auto gate, manual gate, on/off	Sets the method of microphone gating.
Chairman Override	Inputs 1–8	On/off	When a chairman override channel is gated on, all non-chairman inputs are gated off.
Adaptive Ambient mode	Inputs 1–8	On/off	Automatically sets the ambient audio level of the room averaged over time.
PA Adaptive mode	Inputs 1–8	On/off	This prevents mic channels from gating on to loudspeaker audio.
Maximum number of mics on	Mixer-wide	1–8 or off	Sets the maximum number of microphones allowed to be gated on at a time.
First Mic Priority mode	Mixer-wide	On/off	Increases the audio level required to gate on additional microphones after the first mic is gated on.
Last Mic mode	System-wide	Last, Mic 1–8, Off	Keeps the last gated microphone or one Mic 1–8 on when no mics are providing a gating input.
Gate Ratio Adjust	Inputs 1–8	0 to 50dB	Specifies how much louder above the ambient level the audio level must be to gate on.
Off Attenuation Adjust	Inputs 1–8	0 to 50dB	Sets how much the microphone will be attenuated when it is not gated.
Hold Time	Inputs 1–8	.1 to 8.0 seconds	Programs the amount of time it takes until the mic starts the off attenuation process.
Decay Rate	Inputs 1–8	Slow, medium, fast	Programs how quickly the audio level is attenuated once an input hold time has expired.
Manual Ambient Level	Inputs 1–8	0 to -80dB	Sets the ambient audio level when the adaptive ambient mode is off.
NOM/Constant Gain mode	Output sensitive	On/off	Maintains constant gain of a selected output. As more mics gate on, each mic is appropriately attenuated.

Figure 5.5. Mixing parameters

CHAPTER 6: Audio Routing

Matrix Mixing

One of the most important functions of the XAP audio conferencing system is matrix routing of audio signals. Like all device functions, all routing is executed in the digital domain. In addition, changes in routing can be executed via the RS-232 port and/or via presets on the control/status connectors.

The XAP audio matrix has 32 possible input sources and 32 output destinations, with level control at each cross point. The routing chart (Figure 6.2) describes the default XAP routing. Inputs and outputs are labeled for this default routing diagram, but virtually any input and output scheme could be used. Inputs and outputs to the matrix are described below.

Inputs

- **Gated and non-gated inputs.** The mic/line inputs are located on the rear terminal block. Both gated and non-gated inputs are provided on the matrix for delivery to desired destinations. This is provided because, in some applications (such as a courtroom), direct, non-gated outputs are required. Default routing for gated microphone inputs are to the O-Bus. Non-gated outputs are routed by default to their corresponding output number (i.e., input 1 is routed to output 1).
- **Inputs 9–12/5–8.** These are line level inputs that appear on the rear panel terminal blocks. This is typically audio that comes from a CD player, video codec, XAP TH2 telephone interface, and other auxiliary audio sources. In typical applications, this audio must be heard in the local PA system (as well as networked XAP units). In the default routing, audio is routed to every other device except itself.

Outputs

- **Outputs 1–8/1–4.** These are exactly the same as outputs 9–12/5–8. Their default routing is for each non-gated input 1–8/1–4 to go directly to these outputs.
- **Outputs 9–12/5–8.** These are line level outputs on the rear panel terminal blocks. This is typically audio that goes to a XAP TH2 telephone interface, tape recorder, power amp, and other audio devices. Normally, this audio contains auxiliary audio and audio from other networked XAP units. In the default routing, Inputs 9–12/5–8 (minus your channel input) and master auxiliary mix (all auxiliary audio from other XAP units) are contained in this audio.

Expansion bus

The expansion bus is a digital bus that is used to network XAP 800 and XAP 400 units. It is a mix-minus bus. Any audio placed on the bus for a particular unit is not fed back to that unit when audio is taken off that bus. Audio on any networked XAP can be placed on a bus or audio can be taken off a bus and routed to any destination within the unit. The XAP system has 12 digital mix-minus buses with the following default programming.

- **O–R Buses.** These four audio buses are defaulted as the mic mix buses; they can communicate the NOM count (see page 25) across the network to other XAP units. Otherwise, these buses are identical to buses S–Z.
- **S–Z Buses.** These eight buses are defaulted as auxiliary mix buses. They are used to route auxiliary audio, such as from a XAP TH2 Telephone Interface, between units on the network. These buses are used as mic mix buses when NOM count is not required.

PA Adapt expansion bus reference buses

The expansion bus reference buses provide a system-wide bus for mic channels to receive a reference input for PA Adaptive Mode and acoustic echo cancellation. For example, say you have four XAP units expansion bused together. Audio on Output 12 of unit 1 is audio routed to the PA system in the room. This audio is needed as a reference for mics on units 2, 3 and 4 so that speaker audio does not gate on the mics and the echo canceller cancels the appropriate audio. This is accomplished by selecting Output 12 on unit 1 as expansion bus reference and then selecting microphones of units 2, 3, and 4 to use expansion bus reference.

Assignable processing

There are eight assignable processing buses in the XAP 800 and four in the XAP 400. Unlike with other matrix mixers, these buses can route any input or group of inputs to any output or group of outputs. Also, these combined sources can be filtered, delayed, compressed, and attenuated to provide specific enhancements to the audio (see Figure 6.1). These buses are typically used to reduce feedback in the venue and provide crossovers for different speaker systems.

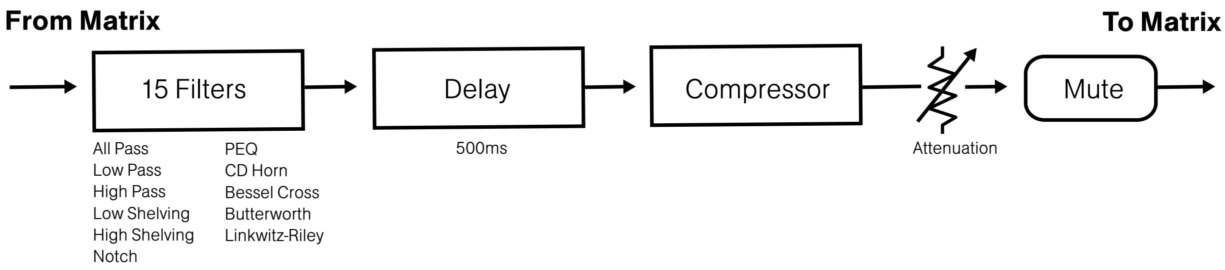


Figure 6.1. Assignable processing

XAP 800
Default Routing

		Output 1	Output 2	Output 3	Output 4	Output 5	Output 6	Output 7	Output 8	Output 9	Output 10	Output 11	Output 12	Inputs 1 - 8 to other units	Inputs 9 - 12 to other units																		
		Outputs												To Expansion Bus												To Processing							
Routing Matrix Setup		1	2	3	4	5	6	7	8	9	10	11	12	O	P	Q	R	S	T	U	V	W	X	Y	Z	A	B	C	D	E	F	G	H
	Input 1	N								G	G	G		G																			
	Input 2		N							G	G	G		G																			
	Input 3			N						G	G	G		G																			
	Input 4				N					G	G	G		G																			
	Input 5					N				G	G	G		G																			
	Input 6						N			G	G	G		G																			
	Input 7							N		G	G	G		G																			
	Input 8								N	G	G	G		G																			
	Input 9										X	X	X					X															
	Input 10									X		X	X					X															
	Input 11									X	X		X					X															
	Input 12									X	X	X	X					X															
Inputs from other units	O Bus									X	X	X																					
Inputs from other units	P Bus																																
	Q Bus																																
	R Bus																																
	S Bus									X	X	X	X																				
	T Bus																																
	U Bus																																
	V Bus																																
	W Bus																																
	X Bus																																
	Y Bus																																
	Z Bus																																
	Processing A																																
	Processing B																																
	Processing C																																
	Processing D																																
	Processing E																																
	Processing F																																
	Processing G																																
	Processing H																																

* G = Gated N = Non-Gated

(Software shows actual cross point values.)

Figure 6.2. Default Routing diagram for the XAP 800

Default Routing diagram

The Default Routing diagram (see Figure 6.2) shows how all inputs, outputs, and buses are default routed. There is level control at each cross point which is adjustable from 0 to -20dB.

Room Preset/Configuration worksheet

The Room Preset/Configuration worksheet (see page 67) is used for recording preset information such as description, command list, port usage, and other parameters. The XAP 800/400 has 32 configurable presets.

Input/Output Parameters worksheet

The Input/Output Parameters Worksheet (Figure 6.3) details each configurable input, output, and processing channel parameter and provides space to record settings for each. Default settings appear in bold.

XAP 800 Input/Output Parameters Worksheet

Input Channel		1	2	3	4	5	6	7	8	9	10	11	12
Program Parameter	**Selection Range**												
Input Type	**Mic 55dB**, Mic 25dB, Line												
Phantom Power	**On**, Off												
Input Gain Adjust	-60dB to +20 dB (**0**)												
AGC	On, **Off**												
Mute	On, **Off**												
Input Filters 1-4	*See Processing Filters Worksheets*												
Input Activation	**Auto**, Manual												
Chairman Mic	On, **Off**												
Gate Ratio	0 - 50 dB (**15**)												
Off Attenuation	0 - 50 dB (**12**)												
Hold Time	.1 - 8.0 seconds (**.3**)												
Decay Rate	Slow, **Medium**, Fast												
Manual Ambient	0dB to -70dB (**-30**)												
Adaptive Ambient	**On**, Off												
PA Adaptive Mode	On, **Off**												
PA Adapt Reference	Output 1-12, AEC/Expansion Bus Ref E1-E4									See System Parameters to define E1-E4.			
Mixer Group Select	Internal 1-4 or Global A-D (A)									See System Parameters to define Mixers.			
Acoustic Echo Cancellation	**On**, Off, adapt reference, non-linear process												
AEC Reference	Output 1-12, AEC/Expansion Bus Ref E1-E4												
Noise Cancellation	On, **Off**, 6dB to 15dB cancellation depth												
Speech Leveler	**Off**,On												
Output Channel		**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**
Program Parameter	**Selection Range**												
Output Gain Adjust	-60dB to +20 dB (**0dB**)												
Mute	On, **Off**												
NOM	**On** or Off												
Processing Channel		**A**	**B**	**C**	**D**	**E**	**F**	**G**	**H**				
Program Parameter	**Selection Range**												
Processing Filters 1-15	*See Processing Filters Worksheets*												
Delay	0-500ms (**0ms**) .02ms steps												
Compressor	On, **Off**												
Threshold	-30dB to +20dB **(0dB)**												
Ratio	**1:1** - 1:20												
Attack Time	0.5ms to 100ms in 0.5ms steps **(1ms)**												
Release Time	5ms to 2sec/Increment of 5ms **(1s)**												
Processing Attenuation	0dB to -60dB **(0dB)**												

ClearOne XAP 800

Figure 6.3. Input/Output Parameters worksheet

CHAPTER 7: System Control

The XAP 800/400 provides a variety of options for system control. You can create up to 32 presets and up to 255 macros to change whole room configurations or run a series of commands. Presets, macros, and commands can be executed using any of the following control options: custom control through Control/Status port A, contact closure through Control/Status port B, ClearOne Control Panels or XAP IR Remote through the RS-485 ports, serially with a touch panel, modem, or PC through the RS-232 port, or front panel LCD menus.

Presets and Macros

A preset is a simply a group of routing and configuration settings stored in the XAP. These settings are applied to the unit when the preset is executed. A good way to think of presets is to consider each preset as a room configuration option. You can create up to 32 presets which enables you to accommodate changing room requirements quickly and efficiently. XAP presets are unique in the sense that they operate independently of other presets in the unit. When a preset is run, only the selected inputs/outputs are changed—all other settings in other presets remain unchanged and are not reset. This means you can change audio routing and configuration settings in a room without affecting settings in other rooms (such as gain).

Presets can be executed in a variety of different ways including the Execute Preset utility in G-Ware, the front panel controls of the XAP unit, RS-232 external control devices, RS-485 control devices (ClearOne Control Panels and XAP IR Remote), logic in/out, and contact closure. You can also create macros which can run multiple presets. These options give you tremendous flexibility when designing your installations and are described in more detail in the XAP 800 and XAP 400 Installation and Operation manuals.

Macros provide powerful options for controlling and operating your XAP system. A macro can contain multiple commands that can reference a single unit or multiple units across the expansion bus. Each XAP 800 and XAP 400 is designed to support up to 255 macros, with an average of 150 command lines each. Macros are created in G-Ware using the Macro Recorder, which records you onscreen selections, or the Macro Editor, which allows you to directly create command lines. The Macro Editor is also used to edit macros created using the Macro Recorder. For more information on creating and using macros, consult the XAP 800 or XAP 400 Installation and Operation manual.

Control and Status Connectors

Control/Status connections are provided on two DB-25 connectors. These connectors are labeled Control/Status A and Control/Status B and contain different types of pins. Control pins on Control/Status A are momentary while control pins on Control/Status B are latching. The inputs on these connectors are internally pulled high and are activated by connecting the pin to ground. The outputs are open collectors, which are open when inactive and grounded when active. This allows the XAP to control and be controlled by a wide variety of external devices, including relays, lamps, switches, and other equipment.

Control/Status A

The GPIO (general-purpose input/output) Builder in G-Ware is used to establish the pin assignments for the 16 user definable pins on Control/Status Port A. These pins provide control via contact closure and status via open collector functions in the unit. Note that the pins numbered in blue are command pins; the pins numbered in green are status pins. The default pin assignments are listed on page 40.

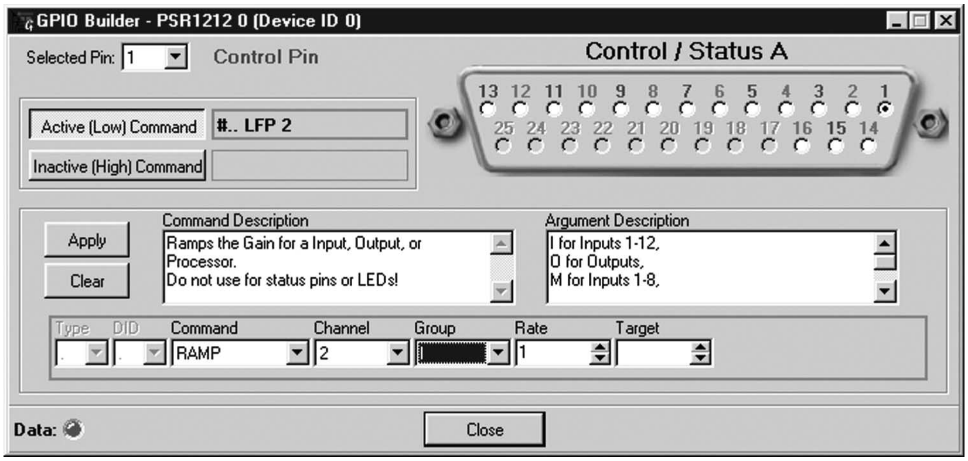


Figure 7.1. Customizing pin assignments with GPIO Builder

Control/Status B

The Control/Status B port is designed to run presets. Using the Preset Mask Control Status B in the Preset window, you can require an active high (H) or active low (L) contact on a control pin (1–19 odd numbers) or combination of several contacts in order to run the preset.

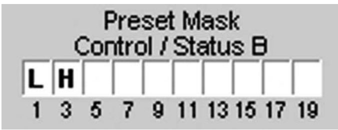


Figure 7.2. Preset Mask Control/Status B

A typical use for Preset Mask Control Status B is a room combining application which uses automatic partitions with sensors or triggers which set the pin to high (H) when the partition is closed and to low (L) when the partition is open. For example, if pin 1 is connected to the first partition and Pin 3 is connected to the second partition, then the Preset Mask Control Status B settings shown in Figure 7.2 will activate the preset when the first partition is open and the second partition is closed.

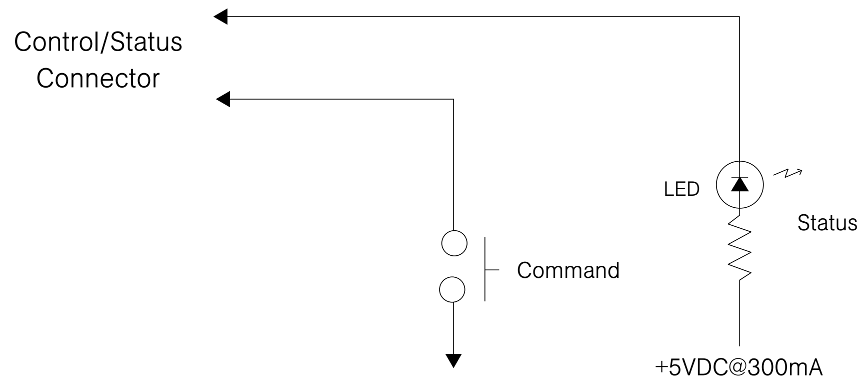


Figure 7.3. Direct control/status operation

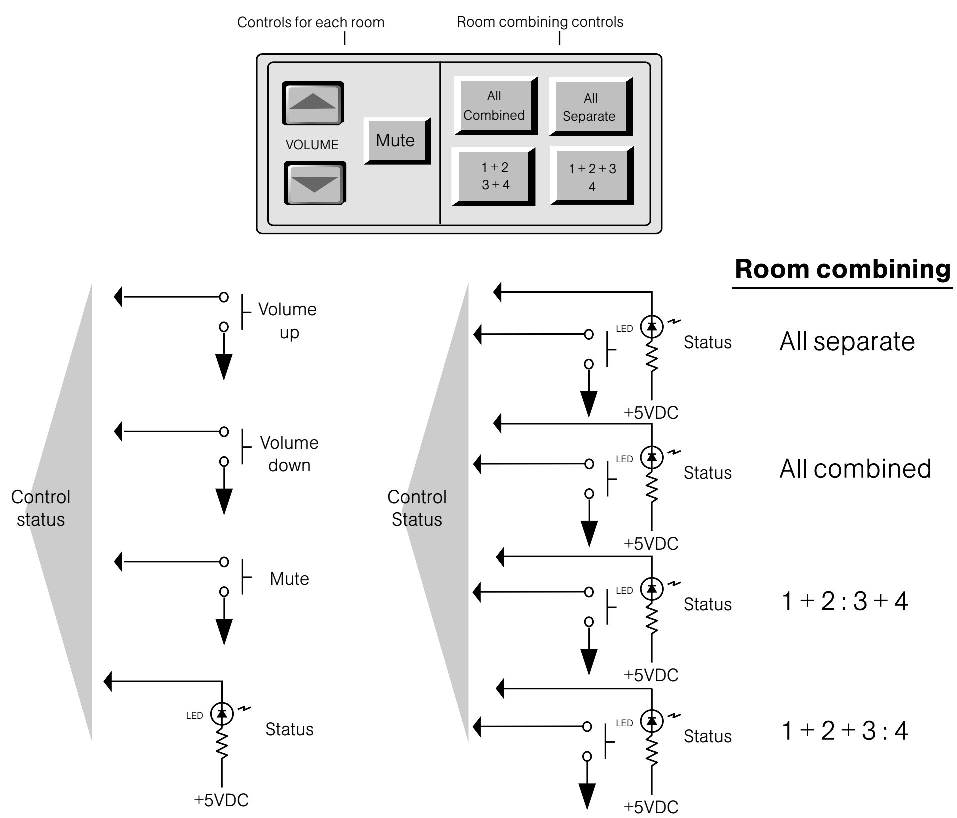


Figure 7.4. Room combining using control/status pins

Control/Status A

Pin	Definable	Type	Default Description
1	Yes	Control	Lock front panel toggle
2	Yes	Status	Status of front panel lock
3	Yes	Control	Mute all mics toggle
4	Yes	Status	Status of mute all mics
5	Yes	Control	Mute Output 9 toggle
6	Yes	Status	Status of Output 9 mute
7	Yes	Control	Mute Output 10 toggle
8	Yes	Status	Status of Output 10 mute
9	Yes	Control	Mute Output 11 toggle
10	Yes	Status	Status of Output 11 mute
11	Yes	Control	Mute Output 12 toggle
12	Yes	Status	Status of Output 12 mute
13	Yes	Control	Output 1 volume up (1dB)
14	Yes	Status	Not programmed
15	Yes	Control	Output 1 volume down (1dB)
16	Yes	Status	Not programmed
17	No	Status	Mic 1 gate status
18	No	Status	Mic 2 gate status
19	No	Status	Mic 3 gate status
20	No	Status	Mic 4 gate status
21	No	Status	Mic 5 gate status
22	No	Status	Mic 6 gate status
23	No	Status	Mic 7 gate status
24	No	Status	Mic 8 gate status
25	No	Ground	Ground

Control/Status B

Pin	Definable	Type	Default Description
1	Yes	Control	Preset select bit
2	Yes	Status	Preset select status (Pin 1)
3	Yes	Control	Preset select bit
4	Yes	Status	Preset select status (Pin 3)
5	Yes	Control	Preset select bit
6	Yes	Status	Preset select status (Pin 5)
7	Yes	Control	Preset select bit
8	Yes	Status	Preset select status (Pin 7)
9	Yes	Control	Preset select bit
10	Yes	Status	Preset select status (Pin 9)
11	Yes	Control	Preset select bit
12	Yes	Status	Preset select status (Pin 11)
13	Yes	Control	Preset select bit
14	Yes	Status	Preset select status (Pin 13)
15	Yes	Control	Preset select bit
16	Yes	Status	Preset select status (Pin 15)
17	No	Status	Preset select bit
18	No	Status	Preset select status (Pin 17)
19	No	Status	Preset select bit
20	No	Status	Preset select status (Pin 19)
21	No Connection		
22	No Connection		
23	No		+5VDC
24	No		+5VDC
25	No	Ground	Ground

Figure 7.5. Default pin programming

ClearOne Control Devices

ClearOne manufactures three control devices designed for use with the XAP units: Volume Control Panel, Select Control Panel, and XAP IR Remote Control. These devices are programmed using the Remote Builder in G-Ware. These control devices are connected to Remote Panel A or Remote Panel B—the RS-485 connectors.

Volume and Select Control Panels

ClearOne Control Panels are convenient wall panels which provide control over the XAP system. There are two Control Panel models: Volume and Select. Volume can be programmed to make specific gain adjustments and Select can be programmed to execute various commands such as presets for room combining applications. You can connect up to six Control Panels in daisy chain fashion to each RS-485 port.

XAP IR Remote

The XAP IR Remote provides remote control of volume and mute for a XAP system. You can connect up to two XAP IR Remote Controls—one to each RS-485 port. See the XAP IR Remote user manual for more information. The XAP IR Remote has five programmable buttons and one programmable LED.

Figure7.6. Volume (left) and Select Control Panels

Figure 7.7. XAP IR Remote

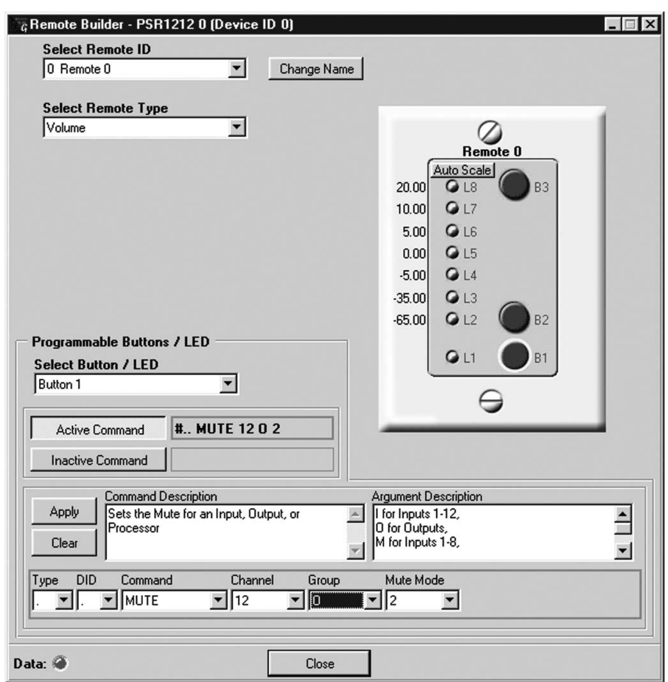


Figure 7.8. Programming a Volume Panel in Remote Builder

Serial Control (RS-232)

Operation of linked XAP units can be done with one RS-232 serial connection. Functions which can be controlled via this connection include audio level control, muting, audio signal routing, telephone dialing, remote diagnostics, and many other functions.

While any external device with an RS-232 serial connection can communicate with the XAP, the system was designed primarily to be programmed and set up using G-Ware, and operated using a custom remote controller.

The XAP provides real-time control and status via the RS-232 port of all system functions, including:

- Input and output audio levels in dBu
- Input and output gain in dB
- Channel input and output muting control and status
- Mic/line input select and phantom power on/off control and status
- Microphone gate activation status
- Control and status of AGC and equalization
- Routing
- Automixing functions and modes
- Control/Status connector configurations
- Preset/macro configurations
- Password protection
- Expansion bus setup
- System setup

Front Panel

The XAP's front panel is intuitive to operate, thanks to its simple interface: a 2x16 character LCD, menu buttons, and a peak-level LED bar meter. Although most of the XAP's features are programmed with G-Ware software, the front panel can be used for simple adjustments and meter monitoring.

To prevent unauthorized changes, the XAP 800 and XAP 400 units can be password protected. When the unit is locked, navigation of the menus is allowed without a password; however, changes to programming require a valid password.

LCD menu tree

The menu tree features five main menus, each with submenus. These branches typically end when an adjustable parameter or viewable value is reached. The diagram below shows the LCD menu tree.

The five main menus are: System, RS-232, Meter, Inputs, and Outputs. All submenu items are arranged under these menus. Use the Enter button to select items and the ▲ and ▼ buttons to scroll through menus and submenus. When the last menu item is reached, the display scrolls back to the beginning of the list. The Esc button allows you to back out of the menus.

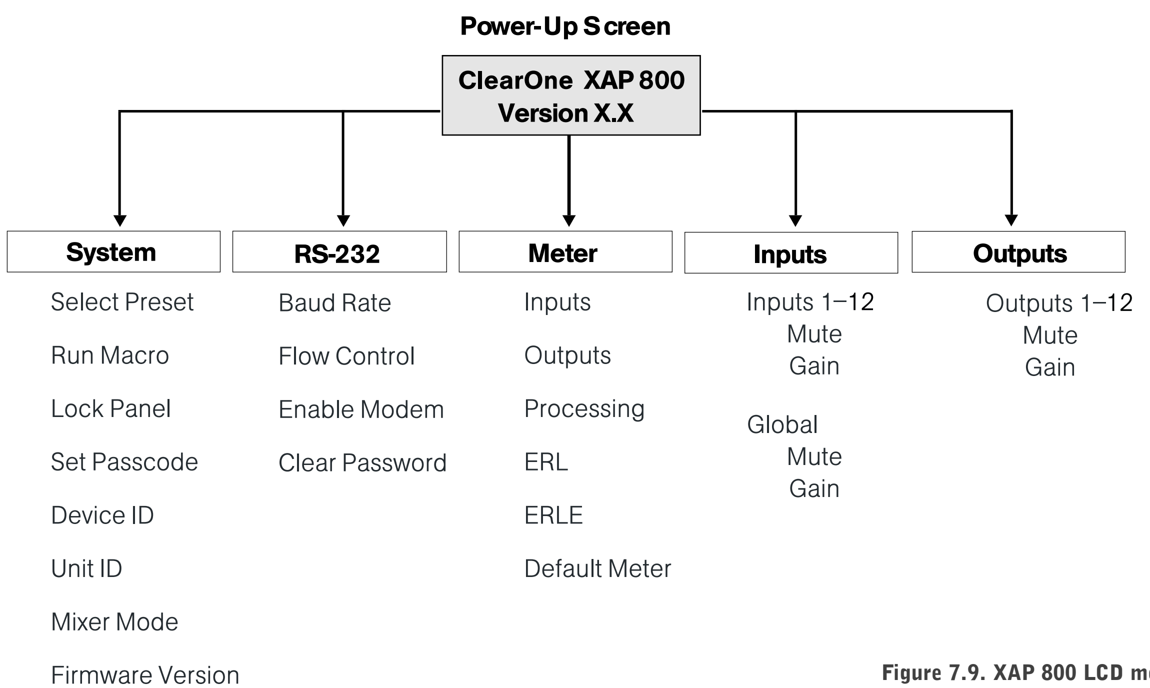


Figure 7.9. XAP 800 LCD menu tree

CHAPTER 8: XAP Connections

System Connections

Audio connections

The XAP 800/400 utilizes removable Phoenix block connectors that are supplied with the unit. To connect, standard audio cables should be stripped and inserted into the terminal block. The terminal screw in the block is then tightened, providing a secure and reliable audio connection. The terminal block can then be inserted into the rear panel connectors. These connectors maximize reliability and ease of use.

Control/Status connections

Direct remote control and status outputs are provided on two DB25 connectors on the rear of the XAP 800.

Expansion bus connection

The expansion bus consists of two RJ-45 connectors. An 18" cable is provided. Additional expansion bus cables are available.

Serial RS-232

The serial RS-232 communications port is connected via a standard DB9 connector. The RS-232 baud rate can be programmed for 9,600, 19.2K, 38.4K, or 57.6K baud rate. Flow control can be set for either hardware or none.

Passcodes

To prevent unwanted access via the front panel or modem, the unit can be programmed to require an access code. The RS-232 password is set from a PC. Should the RS-232 password be forgotten, it can be reset from the front panel.

Meters

The XAP 800 has an LED meter and an LCD. Whenever the input, output, or room loss menus are accessed, the meter displays the level of the parameter selected. When not in the input, output, or room loss menus, the default meter is shown. The default meter can be changed to any input, output, or room loss parameter by pressing the Meter button and selecting the desired reference.

Power

A universal power connector is provided. The XAP 800 will operate on all global voltages and cycles.

Expansion Bus

The expansion bus is a high-speed network protocol that provides two primary system functions: 1) communication among units, and 2) audio linking. All functions of the XAP 800/400 are available across a system of linked XAP units, which allows automixing of up to eight XAP 800/400 units.

XAP takes advantage of its DSP infrastructure in accomplishing this task. Networked XAP units communicate with one another via the expansion bus (see Figure 8.1). Control, status, and addressing functions are performed via the network bus.

Serial connection to any XAP 800 permits programming, operation and diagnostics to all XAP 800 units networked together. This permits a single connection for the installer and user, decreasing costs and complexity.

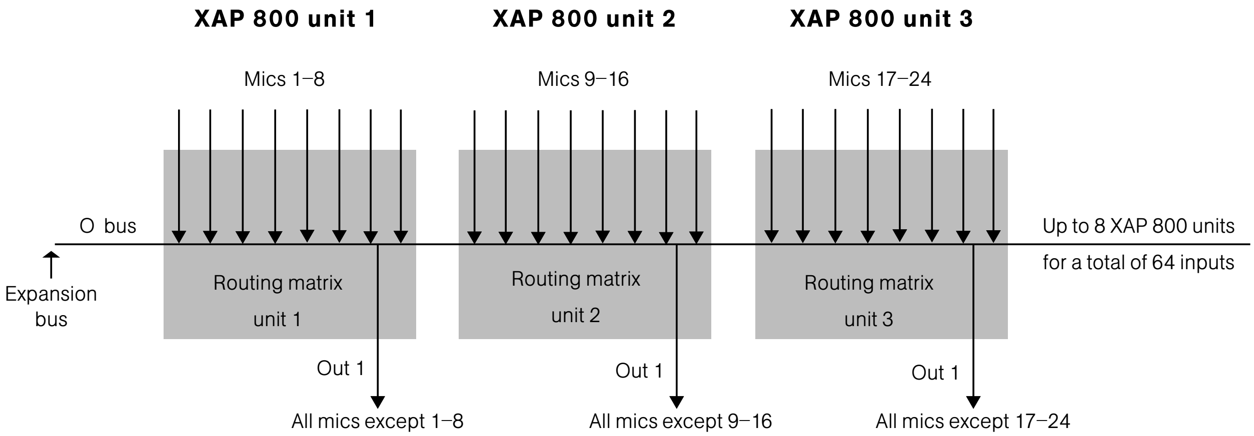


Figure 8.1. Mix-minus configuration of the O bus

Expansion bus audio functions

The expansion bus network architecture allows up to eight XAP 800/400s and up to 16 XAP TH2s to be controlled as if part of a single unit.

Expansion Bus This digital mix-minus bus allows audio routing to and from any destination on the expansion bus network. It contains 12 independent digital audio buses labeled O–Z and four PA Adapt/echo cancellation reference buses. Each audio bus can route mic or line level inputs, in any combination, across the expansion bus network. These buses are divided into two groups—O–R buses and S–Z buses— based on their capabilities and default settings.

O–R buses These four audio buses are defaulted as the mic mix buses; they can communicate the NOM count (see page 25) across the network to other XAP 800s. Otherwise, these buses are identical to buses S–Z.

S–Z buses These eight buses are defaulted as auxiliary mix buses. They are used to route auxiliary audio, such as from a CD player, telephone interface, or video codec, to and from other units on the network. These buses are also used as mic mix buses when NOM count is not required.

PA Adapt/AEC Reference buses These buses provide a system-wide bus for input channels to receive a reference input for PA Adaptive Mode. See page 29 for more information about PA Adaptive mode.

In addition, there are four global mixer groups (A-D). They support first-mic priority, maximum number of mics, etc., and work across all linked XAPs. Unlike the audio buses, they contain only mic status and gate parameters. All gated mics are default routed to the A mixer and to the O bus for routing.

Connecting to the expansion bus

Each XAP 800/400 comes standard with one expansion bus cable. The maximum (cable) distance allowed between any two XAP 800 or XAP 400 units on an expansion bus network is 80 feet (24 meters). ClearOne recommends that category five twisted-pair (10BaseT) cable be used.

CHAPTER 9: Applications

The sophistication and adaptability of the XAP 800 and XAP 400 allow them to control and enhance many conferencing applications. Following are four applications where the XAP 800 forms the centerpiece of a high-quality conferencing system.

Distance Learning

In a typical distance learning application, the primary source of audio comes from the instructor's wireless microphone. For phone-add capabilities, a telephone interface (such as the ClearOne XAP TH2) would be used to transmit sound to distant sites. Secondary audio for presentation segments can be sourced from a VCR or CD player. In larger classroom settings, participants have desktop mics that enable everyone to hear questions and comments.

Microphone mixing and gating parameters can be set to favor the instructor's microphone to facilitate effective dialogue in the room. When a particular microphone gates on, nearby speakers can be attenuated or muted to reduce feedback. Also, participant microphones can be set to gate off when secondary audio sources are in use. The left and right speakers at the front of the room can be configured to provide stereo sound from a (stereo) secondary audio source, enhancing the training experience. All speakers can be optimized to produce full-fidelity audio by programming the XAP 800's parametric equalizers and filters accordingly.

Typical distance learning applications also require that video be transmitted. A video codec provides high-quality video to facilitate a complete distance learning experience.

Figure 9.1 shows a scenario including a wireless lapel microphone for the instructor; desktop microphones for participants; ceiling speakers to carry voice (primary) audio and some secondary audio; and left and right speakers at the front of the room, which carry primary and secondary audio.

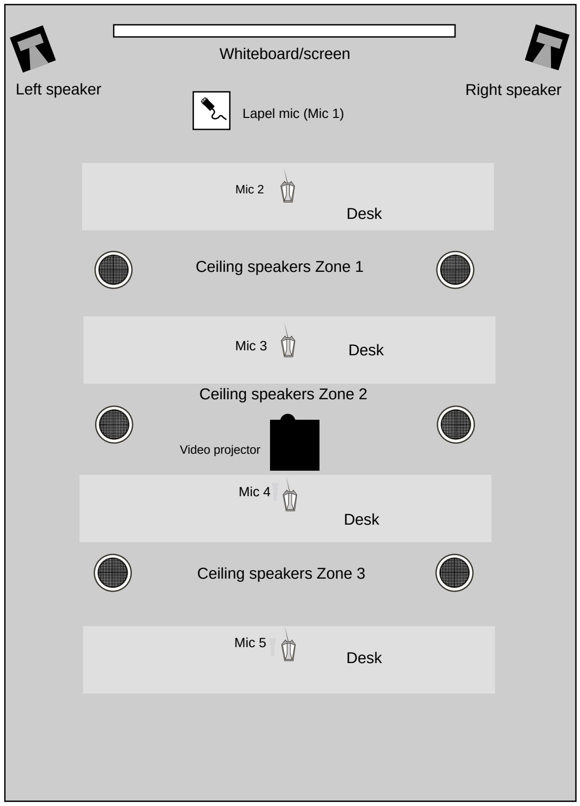


Figure 9.1. Distance learning room

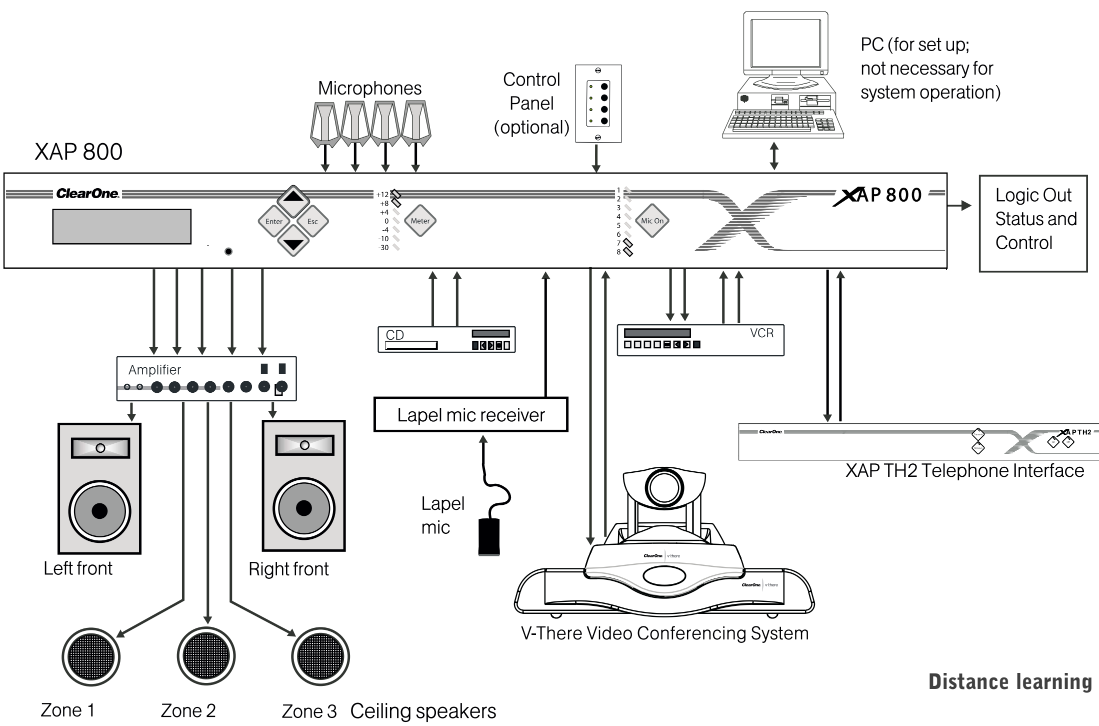


Figure 9.2. Distance learning installation

Hotel/Convention Center

A hotel or convention center sound system must adapt quickly to a variety of meeting scenarios to accommodate the changing needs of the group(s) throughout a meeting session or series of sessions. By configuring the presets on the XAP 800, the system can be quickly reconfigured in a way that accommodates the changing of room configurations in some meeting areas without disturbing meetings in other rooms where no room configuration changes are necessary.

Figure 9.4 shows four rooms with removable partitions. The XAP 800 can be pre-set to route microphone audio to one room or any combination of rooms. For example, say all four rooms are closed off for separate meetings; you can configure a preset to route the microphone audio only to the speaker in that room, with microphone gating properties applied as desired. Then, say the divider between Rooms A and B is removed for a combined meeting. You can use a preset that gates off microphones 3 and 4, while the audio from microphones 1 and 2 are routed to all speakers in Rooms A and B—while retaining the settings for the ongoing meetings in Rooms C and D. Later, when all partitions are removed for a final group meeting, you can use a preset that gates on only microphones 1 and 2, but routes audio to all speakers.

Conferencing capabilities via the XAP TH2 Telephone Interface can easily be routed to any room configuration. The XAP 800's Distributed Echo Cancellation quickly and accurately adjusts to any room. Multiple XAP TH2s can be used to accommodate simultaneous conference meetings in various rooms using just one XAP 800.

The use of other audio sources can be configured using the XAP 800's parametric equalizers and filters to enhance audio quality.

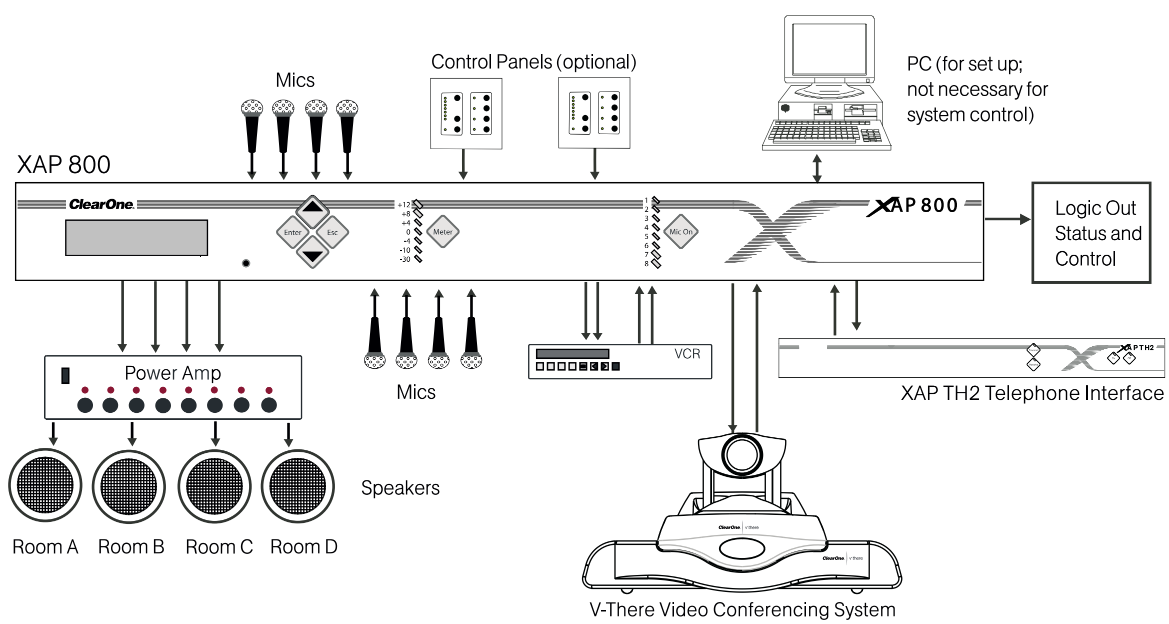


Figure 9.3. Hotel/convention center installation

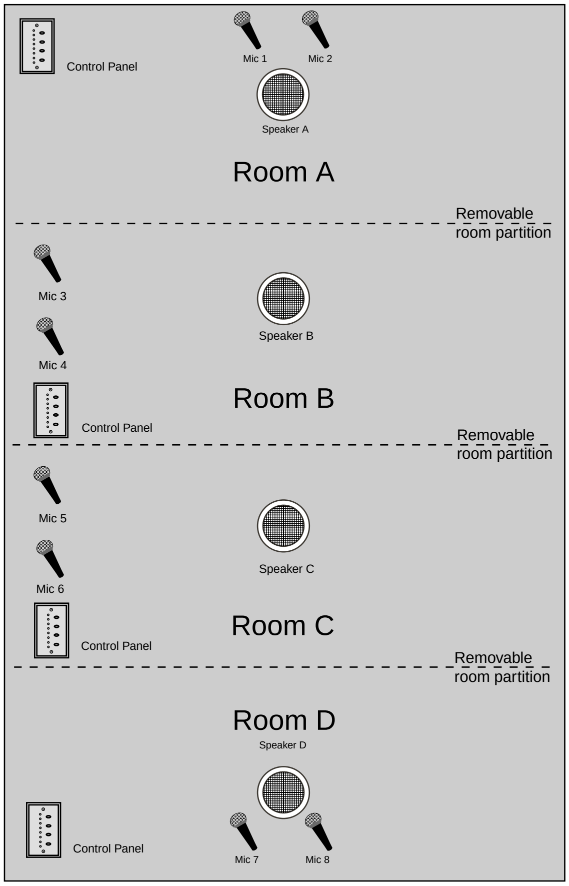


Figure 9.4. Hotel/convention center

Courtroom

A courtroom application requires that the sound system be precisely calibrated so that all voice audio—including that from the judge, legal counsel, witnesses, and the jury—is easy to understand. The XAP 800 includes many features which enhance the performance of any courtroom audio system.

A typical courtroom setting requires at least six microphone inputs and eight line outputs. The XAP 800 features eight microphone inputs, each with individual gain, parametric equalizer, automatic gain control, high/low-pass filters and more. This allows you to provide everyone in the courtroom a natural and clear listening experience.

ClearOne's XAP TH2 is easily integrated into the courtroom system, allowing participants from a phone line to be heard clearly throughout the courtroom. This is an ideal addition to the system because it allows people to appear in the court telephonically—reducing travel expenses and wasted time.

During courtroom sessions, judges might want to hold sidebar conversations with counsel. To prevent jurors from hearing these conversations, white noise masking can be sent out of the jurors' speakers.

Everything that transpires in the courtroom must be recorded. You can configure the XAP 800 to accommodate this requirement by routing all outputs to the court recorder–without gating.

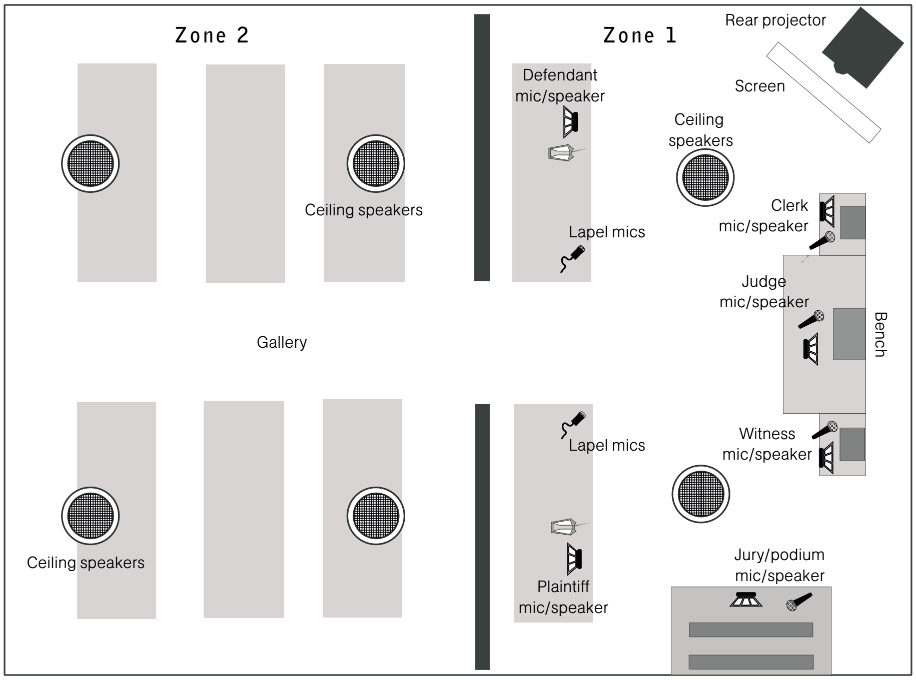


Figure 9.5. Courtroom

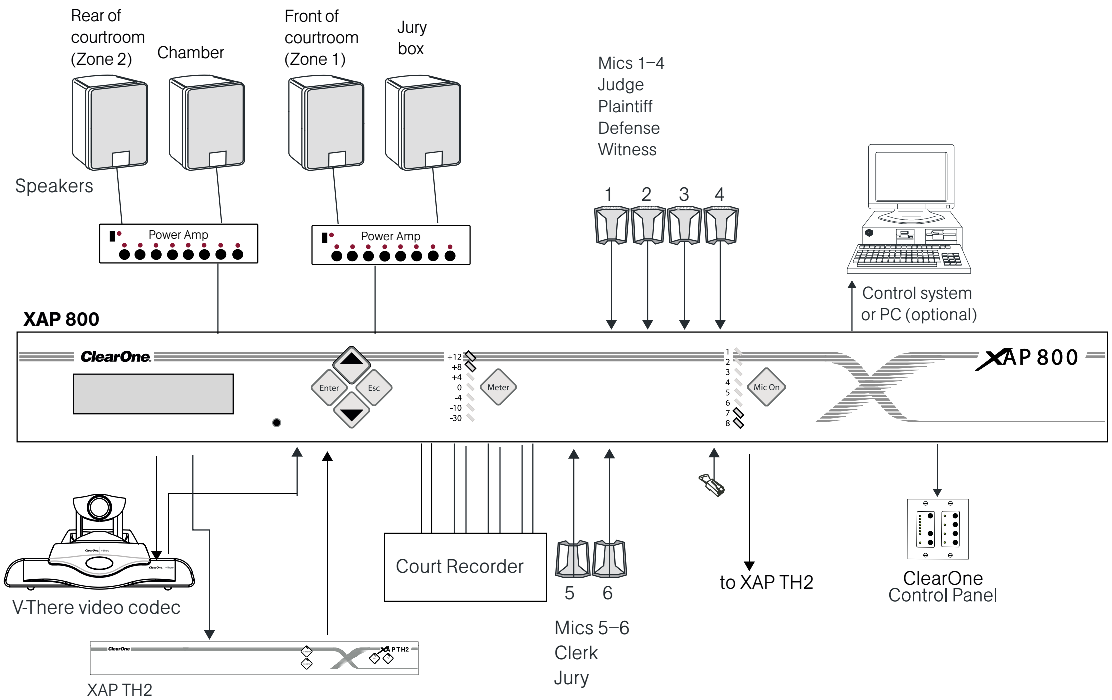


Figure 9.6. Courtroom Installation

Boardroom

A boardroom application is a good example of a situation where microphone mixing and gating become critical to providing seamless dialogue between several people seated around a large table and at distant sites. Figure 9.8 shows a boardroom scenario with seven participants, each with their own microphone and speaker. Also, there are observer seating areas on the periphery of the room with speakers for monitoring the discussion at the table.

Boardrooms typically use audio and video conferencing capabilities. In this scenario, a XAP TH2 is used for bringing in phone line audio and a video codec is used for video conferencing. The XAP 800 allows you to integrate both simultaneously.

Figure 9.7 shows how audio from the video conference will come from the front left and right speakers, which directs participants' attention to the person speaking on the video screen. Program audio and other mic audio is routed to the ceiling speakers for a natural-sounding conference.

Whenever a given mic in the room gates on, audio to that participant's speaker would gate off to prevent feedback, and the volume level of the microphone to adjacent speakers might be reduced below normal output levels, providing a more comfortable audio level. Various parametric equalizers and filters might be configured for particular microphones to enhance the voice qualities of regular participants who sit in assigned seats. All audio would be routed to the observation areas at normal levels.

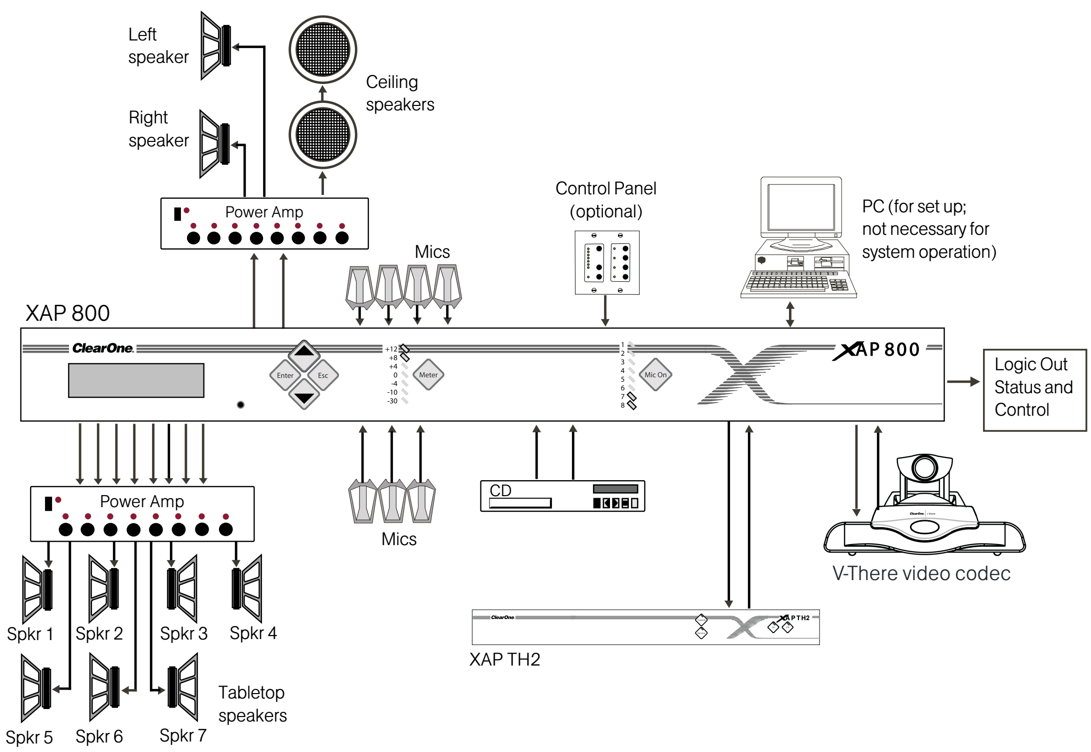


Figure 9.7. Boardroom Installation

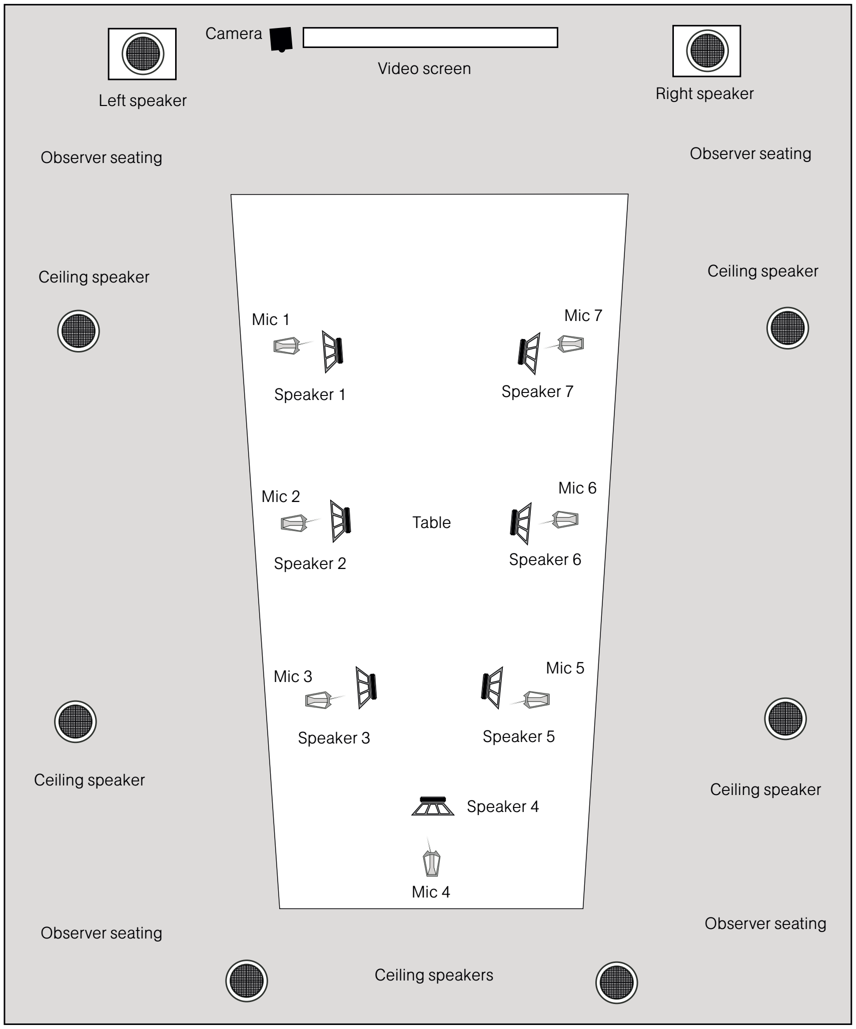


Figure 9.8. Boardroom

APPENDIX: Specifications

XAP 800 Specifications

Dimensions (LxDxH)
17.25″ x 10.25″ x 1.75″
43.8 x 26 x 4.5 cm

Weight
7 lb/4.5 kg dry
12 lb/5.9 kg shipping

Operating Temperature
32–100° F/0–38° C

Humidity
15% to 80%, non-condensing

Power Input Range
Auto-adjusting
100–240VAC; 50/60Hz

Power Consumption
30W typical

Expansion Bus In/Out
Proprietary Network
RJ-45 (2), 115.2kbps, 110kΩ impedance
Category five twisted-pair cable
80′ (24 meters) maximum cable length between any two XAP 800s, XAP 400s, or PSR1212s

RS-232
DB-9 female
9,600/19,200/38,400 (default)/57,600 baud rate; 8 bits, 1 stop, no parity
Hardware flow control on (default)/off

Control/Status
DB25 female A/B (2)
Inputs A/B: active low (pull to ground)
Outputs A/B: open collector, 40VDC max, 40mA each
+5VDC pins (2) (300mA over-current protected)

Remote Panels A/B
4-pin push–on terminal block

RS-485 Proprietary Protocol
Category five twisted–pair cable
1 pair data, 1 pair power and ground
+15VDC (300mA over-current protected)

Mic/Line inputs 1-8
Push-on terminal block, balanced, bridging
Impedance: 5kΩ
Nominal Level: adjustable -55dBu, -25dBu, 0dBu
Maximum Level: -35dBu, -5dBu, +20dBu
Echo Cancellation: 130ms tail time (works with 12dB of room gain)
Noise Cancellation: 6–15dB attenuation
Phantom Power: 24V, selectable

Line Inputs 9-12
Push-on terminal block, balanced, bridging
Impedance: >10kΩ
Nominal Level: 0dBu
Maximum Level: 20dBu

Outputs 1-12
Push-on terminal block, balanced
Impedance: 50Ω
Nominal Level: 0dBu
Maximum Level: 20dBu

Audio Performance
Conditions: Unless otherwise specified, all measurements are performed with a 22Hz to 22kHz BW limit (no weighting)
Frequency Response: 20Hz to 20kHz ±1dB
Noise (EIN): -126dBu, 20kHz BW, max gain, Rs=150Ω
THD+N: <0.02%
SNR: 80dB re 0dBu, (A-weighted)
Dynamic Range: 100dB (A-weighted)
Crosstalk <-91dB re 20dBu @ 20kHz channel to channel

Approvals
FCC, CSA, IC, CE, NOM, ACA, SABS, JATE

Assignable Processing Blocks
Filters:
- All pass
- Low pass
- High pass
- Low shelving
- High shelving
- Parametric EQ
- Notch
- CD Horn

Crossovers
- Bessel
- Butterworth
- Linkwitz-Riley
- Compressor
- Delay adjustable up to 500ms

Matrix Mixing Parameters
32x32 matrix
12 analog in/out
12 Expansion Bus in/out
8 assignable processing blocks in/out

Auto Mixer Parameters
Number of Open Microphones (NOM)
PA Adaptive Mode
First Mic Priority Mode
Last Mic Mode
Maximum # of Mics Mode
Ambient Level
Gate Threshold Adjust
Off Attenuation Adjust
Hold Time
Decay Rate

Microphone Input Configuration
Input Gain Adjust
Mic or Line Level
Phantom Power on/off
Echo Cancellation on/off
Noise Cancellation on/off
Filters
- All Pass
- Low Pass
- High Pass
- Notch
- PEQ

Mute on/off
Chairman Override on/off
AGC on/off
Speech Leveler on/off
Auto Gate/Manual gate
Adaptive Ambient on/off

Set-up Software
G-Ware

XAP 800 Architectural and Engineering Specifications

The echo canceller/automatic mic mixer shall incorporate microphone mixing, echo cancellation, matrix mixing, and signal processing in a single rack space unit.

The echo canceller/automatic mic mixer shall have 12 inputs and outputs: four line-level inputs, eight microphone/line selectable inputs, and 12 line-level outputs. Each mic/line input shall have four selectable filters, adjustable automatic gain control, phantom power, speech leveling, and automatic microphone mixing capabilities. Each mic/line input shall also feature acoustic echo cancellation with a maximum 130ms of tail time and noise cancellation with up to 15dB attenuation, adjustable in 1dB increments. Each input shall also be able to reference any output or any of four global reference buses for echo cancellation. The unit shall have four internal and four global automatic microphone mixers, each with fully adjustable parameters. The microphone mixer shall use PA adaptive, adaptive ambient, chairman override, first mic priority, last mic mode, number of open mics, and look-ahead gating.

The echo canceller/automatic mic mixer shall have a 32x32 internal matrix mixer with attenuation at every cross point in .5dB steps. Any input can be routed to any output or multiple outputs. The matrix shall consist of 12 analog inputs/outputs, 12 digital inputs/outputs from the network bus, and eight inputs/outputs from the processing blocks.

Signal processing shall be provided by eight assignable processing blocks, each with 15 programmable filters, delay, and compression. The processing blocks shall include such filters as high pass, low pass, all pass, low shelving, high shelving, notch, parametric EQ, CD horn, Bessel crossover, Butterworth crossover, and Linkwitz-Riley crossover. Filter setup shall be real-time. The unit shall include a signal generator for pink noise, white noise, and tone sweep capabilities, and shall be assignable to any input on any linked unit.

The echo canceller/automatic mic mixer shall have up to 32 presets. Multiple presets can be used simultaneously without interruptions or interference with other presets. The unit shall feature a macro recorder to create up to 255 macros for simple remote control management of the system.

The unit shall have a 12-channel bi-directional audio bus to pass audio, system control, four channels of echo cancellation, and four channels of NOM for four sub-mixers to other units. The maximum distance between linked units shall be 80 feet (24 meters). Up to eight units can be linked for up to 32 line inputs and 64 mic inputs.

System settings shall be saved in the unit, and shall include password protection.

The unit shall be set up and operated with intuitive software that allows complete configuration of the system. Additional control shall be handled via custom setup software, RS-232 protocol, RS-485 control panels, or contact closure with communication speeds up to 57,600 baud.

The unit shall have the ability to meter a group of inputs or an entire signal flow. Meters shall be provided on inputs, processing, and outputs for echo return loss, echo return loss enhancement, and gate parameters.

The unit shall have a frequency response of 20Hz to 20kHz and a signal to noise ratio of 80dB re 0dBu, A-weighted. It shall have a 48kHz sample rate. It shall operate in environments with up to 12dB of room gain.

The unit shall have an internal power supply that automatically adjusts between 100-240VAC of power input. The unit shall comply with FCC, CSA, IC, CE, NOM, ACA, SABS, VCCI, and JATE requirements.

The ClearOne XAP 800 is specified.

XAP 400 Specifications

Dimensions (LxDxH)
17.25" x 10.25" x 1.75"
43.8 x 26 x 4.5 cm

Weight
9.9 lb/4.5 kg dry
13 lb/5.9 kg shipping

Operating Temperature
32 to 100° F/0 to 38° C

Humidity
15% to 80%, non-condensing

Power Input Range
Auto-adjusting
100–240VAC; 50/60Hz

Power Consumption
30W typical

Expansion Bus In/Out
Proprietary Network
RJ-45 (2), 115.2kbps, 110kΩ impedance
Category five twisted-pair cable
80' (24 meters) maximum cable length between any two XAP 800s, XAP 400s, or PSR1212s

RS-232
DB-9 female
9,600/19,200/38,400(default)/57,600 baud rate; 8 bits, 1 stop, no parity
Hardware flow control on (default)/off

Control/Status
DB-25 female A/B (2)
Inputs A/B: active low (pull to ground)
Outputs A/B: open collector, 40VDC max, 40mA each
+5VDC pins (2) (300mA over-current protected)

Remote Panels A/B
4-pin push–on terminal block
RS-485 Proprietary Protocol
Category five twisted–pair cable
1 pair data, 1 pair power and ground
+15VDC (300mA over-current protected)

Mic/Line Inputs 1-4
Push-on terminal block, balanced, bridging
Impedance: 5kΩ
Nominal Level: adjustable -55dBu, -25dBu, 0dBu
Maximum Level: -35dBu, -5dBu, +20dBu
Echo Cancellation: 130ms tail time (works with 12dB of room gain)
Noise Cancellation: 6–15dB attenuation
Phantom Power: 24V, selectable

Line Inputs 5-8
Push-on terminal block, balanced, bridging
Impedance: >10kΩ
Nominal Level: 0dBu
Maximum Level: 20dBu

Outputs 1-8
Push-on terminal block, balanced
Impedance: 50Ω
Nominal Level: 0dBu
Maximum Level: 20dBu

Audio Performance
Conditions: Unless otherwise specified, all measurements are performed with a 20Hz to 20kHz BW limit (no weighting)
Frequency Response: 20Hz to 20kHz ±1dB
Noise (EIN): -126dBu, 20kHz BW, max gain, Rs=150Ω
THD+N: <0.02%
SNR: 80dB re 0dBu (A-weighted)
Dynamic Range: 100dB (A-weighted)
Crosstalk <-91dB re 20dBu @ 20kHz channel to channel

Assignable Processing Blocks
Filters:
- All pass
- Low pass
- High pass
- Low shelving
- High shelving
- Parametric EQ
- Notch
- CD Horn

Crossovers:
- Bessel
- Butterworth
- Linkwitz-Riley

Compressor; adjustable
Delay; adjustable up to 250ms

Matrix Mixing Parameters
25x26 matrix
8 analog in/out
1 speaker out (10W)
12 expansion bus in/out
4 assignable processing blocks in/out
1 telco in/out

Auto Mixer Parameters
Number of Open Microphones (NOM)
PA Adaptive Mode
First Mic Priority Mode
Last Mic Mode
Maximum # of Mics Mode
Ambient Level
Gate Threshold Adjust
Off Attenuation Adjust
Hold Time
Decay Rate

Microphone Input Configuration
Input Gain Adjust
Mic or Line Level
Phantom Power on/off
Echo Cancellation on/off
Noise Cancellation on/off
Filters
- All Pass
- Low Pass
- High Pass
- Notch
- PEQ

Mute on/off
Chairman Override on/off
AGC on/off
Speech Leveler on/off
Auto Gate/Manual gate
Adaptive Ambient on/off

Telco Line
RJ-11
POTS (plain old telephone service) or analog extension from a PBX
A-lead supervision provided

Telco Set
RJ-11
Connect analog telephone set
A-lead supervision provided

Telephone Audio Performance
Conditions: Unless otherwise specified, all measurements are performed with AGC disabled, referenced @ -15dBm on/off the telephone line
Frequency Response: 250Hz to 3.3kHz ±1dB
THD+N: <0.2%, 250Hz to 3.3kHz
SNR: >62dB re max level
Pre-emphasis: 4dB @ 2kHz

Telephone Echo Cancellation
Tail Time: 31ms
Null: >55dB

Telephone Noise Cancellation
Noise Cancellation: 6–15dB attenuation

Set-up Software
G-Ware

Approvals
FCC, CSA, IC, CE, NOM, ACA, SABS, JATE

XAP 400 Architectural and Engineering Specification

The echo canceller/automatic microphone mixer shall incorporate automatic microphone mixing, distributed echo cancellation, noise cancellation, matrix mixing, signal processing, a 10W power amp, and a telephone hybrid in a single rack space unit.

The echo canceller/automatic mic mixer shall have eight inputs and nine outputs: four mic/line selectable inputs, four line level inputs, eight line level outputs, and one 10W power amp. Each mic/line input shall have four selectable filters, adjustable automatic gain control, phantom power, speech leveling, and automatic microphone mixing capabilities. Each mic/line input shall also feature acoustic echo cancellation with 130ms of tail time and noise cancellation with up to 15dB attenuation, adjustable in 1dB increments. It shall operate in environments with up to 12dB of room gain. Each input shall also be able to reference any output, any of four global reference buses, or any of four summing virtual references for echo cancellation. The unit shall have four internal and four global automatic microphone mixers, each with fully adjustable parameters. The microphone mixer shall use PA adaptive, adaptive ambient, chairman override, first mic priority, last mic mode, number of open mics, and look-ahead gating. The unit shall have front panel control of gain and mute for inputs, outputs, and processing channels.

The echo canceller/automatic mic mixer shall have a 25x26 internal matrix mixer with attenuation at every cross point in .5dB steps. Any input can be routed to any output or multiple outputs. The matrix shall consist of eight analog inputs/outputs, one speaker output, 12 digital inputs/outputs from the network bus, four inputs/outputs from the processing blocks, and one telephone input/output.

Signal processing shall be provided by four assignable processing blocks, each supporting 15 programmable filters, delay, and compression. The processing block filters shall include: high pass, low pass, all pass, low shelving, high shelving, notch, parametric EQ, CD horn, Bessel crossover, Butterworth crossover, and Linkwitz-Riley crossover. Filter setup shall be real-time. The unit shall include a signal generator for pink noise, white noise, and tone sweep capabilities, and shall be assignable to any input on any linked unit.

The unit shall have a 12-channel bi-directional audio bus to pass audio, system control, four channels for echo cancellation referencing, and four channels of NOM for four sub-mixers to other units. The maximum distance between linked units shall be 80 feet (24 meters). Up to eight units can be linked together.

The echo canceller/automatic mic mixer shall have up to 32 presets and shall allow password protection of presets. Multiple presets can be used simultaneously without interruptions or interference with other presets. The unit shall feature a macro recorder to create up to 255 macros for simple remote control management of the system. System settings shall be saved in the unit, and shall include password protection.

The unit shall be set up and operated with intuitive software that allows complete configuration of the system. Additional control shall be handled via RS-232 protocol with communication speeds up to 57,600 baud, RS-485 control panels or remote control, or contact closure.

The unit shall have a frequency response of 20Hz to 20kHz and a signal to noise ratio of 80dB re 0dBu, A-weighted. It shall have a 48kHz sample rate.

The echo canceller/automatic mic mixer shall have an integrated standard 600Ω analog telephone hybrid with a nominal level of -9 to -15dB. The unit shall provide two RJ-11 connectors for connection to a telephone line and telephone set. The hybrid shall have telephone line echo cancellation with a tail time of no less than 31ms and noise cancellation with 6–15dB of attenuation. The audio frequency response of the hybrid shall be at least 250Hz to 3300Hz with a signal to noise ratio of >62dB re max level. DTMF dialing capability shall be provided on the hybrid.

The echo canceller/automatic mic mixer shall allow manual enabling/disabling of telephone calls through buttons on the front panel. Additional settings shall be enabled through configuration software. This software shall allow selection of either burst adapt or auto adapt hybrid null, receive reduction, selectable receive boost of 0, 3, 6, 9, or 12dB, on/off receive automatic level control (ALC), auto answer, auto disconnect, loop drop or call progress, DTMF level, dial tone level, audible hook indication, and audible ringer indication. The unit shall be able to store 10 speed dial numbers.

The unit shall have an internal power supply that automatically adjusts between 100-240VAC of power input. The unit shall comply with FCC, CSA, IC, CE, NOM, ACA, SABS, VCCI, and JATE requirements.

The ClearOne XAP 400 is specified.

XAP TH2 Specifications

Dimensions (LxDxH)
17.25" x 10.25" x 1.75"
43.8 x 26 x 4.5 cm

Weight
7 lb/3.18 kg dry
12 lb/5.4 kg shipping

Operating Temperature
32 to 100° F/0 to 38° C

Humidity
15% to 80%, non-condensing

Power Input Range
Auto-adjusting
100–240VAC; 50/60Hz

Power Consumption
15W typical

Device ID
4-position DIP switch

Expansion Bus In/Out
Proprietary Network
RJ-45 (2), 115.2kbps, 110kΩ Impedance
Category five twisted-pair cable 80' (24 meters) maximum cable length between any two PSR1212s, XAP 800s or XAP 400s

RS-232
DB-9 female
9,600/19,200/38,400 (default)/57,600 baud rate; 8 bits, 1 stop, no parity
Hardware flow control on (default)/off

Control/Status
DB-25 female
Inputs: active low (pull to ground)
Outputs: open collector, 40VDC max, 40mA each
+5VDC (300mA over-current protected)

Telco Line
RJ-11
POTS (plain old telephone service) or analog extension from a PBX
A-lead supervision provided

Telco Set
RJ-11
Connect analog telephone set
A-lead supervision provided

Transmit Input
Push-on terminal block, balanced, bridging
Impedance: >10kΩ
Nominal Level: 0dBu
Maximum Level: 15dBu

Receive Output
Push-on terminal block, balanced
Impedance: <50Ω
Nominal Level: 0dBu
Maximum Level: 19dBu

Audio Performance
Conditions: Unless otherwise specified, all measurements are performed with a 22Hz to 22kHz BW limit (no weighting). Transmit limiter and Receive ALC disabled
Frequency Response: 250Hz to 3.3kHz ±1dB
THD+N: <0.2% re max level
SNR: >62dB re max level

Telco Echo Cancellation
Tail time: 31ms
Null: 55dB nominal

Approvals
FCC, CSA, IC, CE, NOM, ACA, SABS, JATE

Set-up Software
G-Ware

XAP TH2 Architectural and Engineering Specification

The digital telephone hybrid shall integrate a standard 600-ohm analog telephone line with a nominal level of -9 to -15dB into an audio conferencing system. The hybrid shall provide two RJ-11 connectors for connection to a telephone line and telephone set.

The hybrid shall include a balanced line-level input and a balanced line-level output to pass audio to and from the audio conferencing device. The input and output shall each have a selectable mute accessed via serial command, custom software, or contact closure. The input and output shall use three-pin Phoenix connectors.

The hybrid shall have telephone line echo cancellation with a tail time of no less than 31ms. The audio frequency response shall be at least 250Hz to 3300Hz with a signal to noise ratio of >62dB re max level.

An audible/visual/serial ring indication and DTMF dialing capability shall be provided on the hybrid.

The unit shall have a digital network bus to receive system control. Up to sixteen units can be linked to add up to 16 telephone lines to the audio conferencing system.

The hybrid shall allow manual enabling/disabling of telephone calls through buttons on the front panel. It shall have selectable auto answer and auto disconnect. Additional settings shall be enabled through configuration software. This software shall select either burst adapt or auto adapt hybrid null, receive reduction, selectable receive boost of 0, 3, 6, 9, or 12dB, on/off receive automatic level control (ALC), auto answer, auto disconnect, loop drop or call progress, DTMF level, dial tone level, audible hook indication, and audible ringer indication. The hybrid shall be able to store 10 speed dial numbers.

The presence of transmit and receive audio and clipped audio signals, as well as on and off status, shall be indicated by LEDs on the front panel of the unit.

The unit shall have the ability to meter transmit and receive audio.

The unit shall be set up and operated with intuitive software that allows complete configuration of the system. Additional control shall be handled via custom control software, RS-232 protocol (with communications speeds up to 57,600 baud), or contact closure.

The hybrid shall use a single rack space and an internal 100-240VAC auto-adjusting power supply. It shall meet FCC, CSA, IC, NOM, Cofetel, ACA, SABS, JATE, and CE requirements.

The ClearOne XAP TH2 is specified.

Worksheets

ClearOne XAP 800

XAP 800 System Parameters Worksheet

System-Wide Parameters

Program Parameter	Selection Range		Program Parameter	Selection Range	
Timeout	0 - 15 (**5**)		Mixer Mode	**Master**, Slave	
Lock Front Panel	On, **Off**		RS-232 Baud Rate	9.6, 19.2, **38.4**, 57.6kbps	
Set Passcode	Any 5 Front Panel Keys		RS-232 Flow Control	On, **Off**	
Device ID No.	**0** - 7		Modem Mode	On, **Off**	
Unit ID No.	Factory Programmed		Clear Modem Password	**Enter**	
			Default Meter	Input/Output Channel 1-12 **(Out 12)**	

Mixer Group Parameters

Program Parameter	Selection Range	Internal/Unit Mixer Groups 1	2	3	4	Global/Site Mixer Groups A	B	C	D
Maximum No. of Mics	Off, 1-8 (**4**)								
First Mic Priority	**On,** Off								
Last Mic Mode	Last On. **Off**. Mic 1 - Mic 8								

Define PA Adapt/AEC Expansion Bus References

Reference #:	Define as Output: (1-12)	On Unit: (Device ID 0-7)
E1		
E2		
E3		
E4		

Tip:
Each Expansion Bus PA Adapt reference can only be defined once across the system (all linked XAP 800s).
Before defining, verify that the given reference has not already been defined on another unit.
See the Input - Output Parameters worksheet(s) to define a PA Adapt reference to a local output on this unit.

Application Notes

Date	Note

XAP 800 System Parameters worksheet

XAP 800 Input/Output Parameters Worksheet

ClearOne XAP 800

Input Channel		1	2	3	4	5	6	7	8	9	10	11	12
Program Parameter	**Selection Range**												
Input Type	**Mic 55dB**, Mic 25dB, Line												
Phantom Power	**On**, Off												
Input Gain Adjust	-60dB to +20 dB (**0**)												
AGC	On, **Off**												
Mute	On, **Off**												
Input Filters 1-4	*See Processing Filters Worksheets*												
Input Activation	**Auto**, Manual												
Chairman Mic	On, **Off**												
Gate Ratio	0 - 50 dB (**15**)												
Off Attenuation	0 - 50 dB (**12**)												
Hold Time	.1 - 8.0 seconds (**.3**)												
Decay Rate	Slow, **Medium**, Fast												
Manual Ambient	0dB to -70dB (**-30**)												
Adaptive Ambient	**On**, Off												
PA Adaptive Mode	On, **Off**												
PA Adapt Reference	Output 1-12, AEC/Expansion Bus Ref E1-E4									See System Parameters to define E1-E4.			
Mixer Group Select	Internal 1-4 or Global A-D (A)									See System Parameters to define Mixers.			
Acoustic Echo Cancellation	**On**, Off, adapt reference, non-linear process												
AEC Reference	Output 1-12, AEC/Expansion Bus Ref E1-E4												
Noise Cancellation	On, **Off**, 6dB to 15dB cancellation depth												
Speech Leveler	**Off**,On												

Output Channel		1	2	3	4	5	6	7	8	9	10	11	12
Program Parameter	**Selection Range**												
Output Gain Adjust	-60dB to +20 dB (**0dB**)												
Mute	On, **Off**												
NOM	**On** or Off												

Processing Channel		A	B	C	D	E	F	G	H
Program Parameter	**Selection Range**								
Processing Filters 1-15	*See Processing Filters Worksheets*								
Delay	0-500ms (**0ms**) .02ms steps								
Compressor	On, **Off**								
Threshold	-30dB to +20dB **(0dB)**								
Ratio	**1:1** - 1:20								
Attack Time	0.5ms to 100ms in 0.5ms steps **(1ms)**								
Release Time	5ms to 2sec/Increment of 5ms **(1s)**								
Processing Attenuation	0dB to -60dB **(0dB)**								

XAP 800 Input/Output Parameters worksheet

XAP 800 Processing Filter Parameters Worksheet #1

ClearOne XAP 800

Channel			A	B	C	D	E	F	G	H	1	2	3	4	5	6	7	8
Filter #	**Filter Parameter**	**Selection Range**																
1	Filter Type	Abrev. *(See Key)*																
	Center or Knee Frequency	Hz																
	Gain/Slope	dB or dB/Octave																
	Bandwidth	Octaves																
	Filter Sub-Type	LP/HP *(See Key)*																
2	Filter Type	Abrev. *(See Key)*																
	Center or Knee Frequency	Hz																
	Gain/Slope	dB or dB/Octave																
	Bandwidth	Octaves																
	Filter Sub-Type	LP/HP *(See Key)*																
3	Filter Type	Abrev. *(See Key)*																
	Center or Knee Frequency	Hz																
	Gain/Slope	dB or dB/Octave																
	Bandwidth	Octaves																
	Filter Sub-Type	LP/HP *(See Key)*																
4	Filter Type	Abrev. *(See Key)*																
	Center or Knee Frequency	Hz																
	Gain/Slope	dB or dB/Octave																
	Bandwidth	Octaves																
	Filter Sub-Type	LP/HP *(See Key)*																
5	Filter Type	Abrev. *(See Key)*																
	Center or Knee Frequency	Hz																
	Gain/Slope	dB or dB/Octave																
	Bandwidth	Octaves																
	Filter Sub-Type	LP/HP *(See Key)*																
6	Filter Type	Abrev. *(See Key)*																
	Center or Knee Frequency	Hz																
	Gain/Slope	dB or dB/Octave																
	Bandwidth	Octaves																
	Filter Sub-Type	LP/HP *(See Key)*																
7	Filter Type	Abrev. *(See Key)*																
	Center or Knee Frequency	Hz																
	Gain/Slope	dB or dB/Octave																
	Bandwidth	Octaves																
	Filter Sub-Type	LP/HP *(See Key)*																

Note:
Input channels can select from All Pass, High Pass, Low Pass and Parametric EQ filters only.
Processing channels can select from all available filters.

Key:	Filter Type	Frequency	Gain/Slope	Bandwidth	Filter Sub-Type	Filter Type	Frequency	Gain/Slope
	Parametric EQ (PEQ)	20Hz to 20kHz	-80dB to +15dB	.05 to 5.0 Octaves		All Pass (AP)	20Hz to 20kHz	
	CD Horn EQ (CD)	20Hz to 20kHz				Low Pass (LP)	20Hz to 20kHz	
	Bessel Crossover (BC)	20Hz to 20kHz	12, 18, 24dB/oct		LP (Low Pass) or HP (High Pass)	High Pass (HP)	20Hz to 20kHz	
	Butterworth Crossover (BT)	20Hz to 20kHz	12, 18, 24dB/oct		LP (Low Pass) or HP (High Pass)	Low Shelving (LS)	20Hz to 20kHz	0 to +/-15dB (.5dB)
	Linkwitz Riley Cross. (LR)	20Hz to 20kHz	12, 18, 24dB/oct		LP (Low Pass) or HP (High Pass)	High Shelving (HS)	20Hz to 20kHz	0 to +/-15dB (.5dB)

XAP 800 Processing Filter Parameters worksheet #1

XAP 800 Processing Filter Parameters Worksheet #2

ClearOne XAP 800

Channel			A	B	C	D	E	F	G	H	1	2	3	4	5	6	7	8
Filter #	**Filter Parameter**	**Selection Range**																
8	Filter Type	Abrev. *(See Key)*																
	Center or Knee Frequency	Hz																
	Gain/Slope	dB or dB/Octave																
	Bandwidth	Octaves																
	Filter Sub-Type	LP/HP *(See Key)*																
9	Filter Type	Abrev. *(See Key)*																
	Center or Knee Frequency	Hz																
	Gain/Slope	dB or dB/Octave																
	Bandwidth	Octaves																
	Filter Sub-Type	LP/HP *(See Key)*																
10	Filter Type	Abrev. *(See Key)*																
	Center or Knee Frequency	Hz																
	Gain/Slope	dB or dB/Octave																
	Bandwidth	Octaves																
	Filter Sub-Type	LP/HP *(See Key)*																
11	Filter Type	Abrev. *(See Key)*																
	Center or Knee Frequency	Hz																
	Gain/Slope	dB or dB/Octave																
	Bandwidth	Octaves																
	Filter Sub-Type	LP/HP *(See Key)*																
12	Filter Type	Abrev. *(See Key)*																
	Center or Knee Frequency	Hz																
	Gain/Slope	dB or dB/Octave																
	Bandwidth	Octaves																
	Filter Sub-Type	LP/HP *(See Key)*																
13	Filter Type	Abrev. *(See Key)*																
	Center or Knee Frequency	Hz																
	Gain/Slope	dB or dB/Octave																
	Bandwidth	Octaves																
	Filter Sub-Type	LP/HP *(See Key)*																
14	Filter Type	Abrev. *(See Key)*																
	Center or Knee Frequency	Hz																
	Gain/Slope	dB or dB/Octave																
	Bandwidth	Octaves																
	Filter Sub-Type	LP/HP *(See Key)*																
15	Filter Type	Abrev. *(See Key)*																
	Center or Knee Frequency	Hz																
	Gain/Slope	dB or dB/Octave																
	Bandwidth	Octaves																
	Filter Sub-Type	LP/HP *(See Key)*																

XAP 800 Processing Filter Parameters worksheet #2

XAP 800 Room Preset/Configuration Worksheet

Preset Setup

Preset #						
Description:						
Command List:						
For extra large presets, continue commands in next column. Refer to macro worksheet for details on all listed macros.						
DB25 Remote Control Port "B" Setup						
DB25 Port "B" Pin #	1 3 5 7 9 11 13 15 17 19	1 3 5 7 9 11 13 15 17 19	1 3 5 7 9 11 13 15 17 19	1 3 5 7 9 11 13 15 17 19	1 3 5 7 9 11 13 15 17 19	1 3 5 7 9 11 13 15 17 19
Preset Mask =						

Key: **H = Pin Active/High State** **L = Pin Inactive/Low State** **X = Pin state is irrelevant**

Room Configuration Setup

Room Configuration No./Name and Description:							
Configuration's Presets:							
Each configuration is recalled by activating its listed presets.							

Room Preset Configuration worksheet

Glossary

Acoustic Echo Cancellation (AEC) A process in which acoustical echo is removed from a signal. AEC can be used to remove unwanted signals from mic audio if the unwanted acoustic signal is available separately as an electronic signal.

Adaptive Ambient This portion of the mixer monitors the varying ambient noise level in the room and changes the threshold level at which a microphone gates on.

Ambient Level The manually-set background noise level upon which the XAP 800/400 bases gating protocols. Used only if the Adaptive Ambient feature isn't used.

Ambient Noise The existing room-level noise, such as that caused by ventilation systems, paper shuffling, and background chatter.

Amplitude Plot A plot of amplitude (-18 to 18dB) vs. frequency (20Hz to 20kHz) on a logarithmic scale.

ASCII The American Standard Code (for) Information Interchange. Standard code for transmitting alphanumeric characters electronically.

Attack This signal parameter determines how quickly compression is enabled. It is calibrated in milliseconds.

Attenuation A reduction of signal amplitude.

Audio Processor A device that modifies an audio signal in response to certain requirements.

Automatic Gain Control (AGC) Automatically increases or decreases audio gain to maintain a consistent audio level.

Automatic Gating Automatically gates microphones on or off based on input levels and other parameters programmed into the XAP 800/400.

Bandwidth The difference between the lower and upper 3dB endpoints of an audio band. Also, the range or differences between the limiting frequencies of a continuous frequency band.

Baud Rate The number of signal transitions per second, or the clock rate of the serial bit stream in hertz. Given 7 or 8 bits for data plus start and stop, the approximate ASCII character transmission rate is one-tenth the baud rate.

Chairman Override Provides gating priority for all microphones selected for the chairman override group. When a mic in this group gates on, all microphones not included in this group gate off.

Clipping A condition in which a signal level exceeds the maximum level a circuit can handle. This is usually caused by overdriving an input. It always causes distortion and typically leads to listener fatigue and accelerated failure of loudspeaker drivers.

Compression An induced reduction in the dynamic range of part or all of an audio signal. Compression is usually used to protect individual loudspeaker components from the damaging effects of transients.

Constant Directivity Horn Equalizer (CD Horn EQ) Horn drivers commonly used in arrays in arenas and auditoriums have an inherent 6dB/octave high frequency rolloff. The XAP 800/400 CD Horn EQ compensates for this characteristic.

Crossover A device that passes designated frequency segments of an audio signal to various loudspeaker elements in a sound system.

Crossover, Bessel A crossover using a low-pass filter design characterized by a linear phase response. This results in a constant time delay throughout the passband.

Crossover, Butterworth A crossover using a low-pass filter design characterized by a maximally flat magnitude response. This results in no amplitude ripple in the passband.

Crossover, Linkwitz-Riley A fourth-order crossover consisting of a cascaded second-order Butterworth low-pass filter. Offers a vast improvement over the Butterworth crossover and is the de facto standard for professional audio active crossovers.

Decay Rate (slow, medium, fast) Programs how quickly the audio level is attenuated once a channel has been gated off.

DSP Digital signal processor.

Echo Canceller Reference The signal point which contains an electronic copy of all signals which will be removed from the signal the mics pick up. This is always the far-end audio and, optionally, local program material.

Expansion bus Consists of two RJ-45 connectors on the rear panel of the XAP 800/400. An expansion bus allows multiple XAP units to be networked together using category five twisted-pair (10BaseT) cable.

Filter A device that passes and blocks audio signals based on user-definable requirements of the system.

Filter, All Pass A filter that provides only phase shift or phase delay without appreciably changing the magnitude characteristic. The filter produces a flat amplitude response. It is useful for matching the delay of two processing channels with different delays.

Filter Display A group of nodes plotted on a logarithmic scale. The XAP 800/400 filter display can be accessed through the Inputs 1–8/1–4, From Processing, or To Processing windows.

Filter, High Pass A filter that passes high signal frequencies while attenuating low frequencies. The gain or loss

Filter, High Shelving Provides boosting or attenuation of frequencies above a designated frequency. The transition between the spectrum above and below the designated frequency occurs at a fixed 6dB/octave rate. The gain or loss above the corner frequency is adjustable to +/- 15dB.

Filter, Low Pass A filter that passes low frequencies while attenuating high frequencies.

Filter, Low Shelving Provides boosting or attenuation of frequencies below a designated frequency. The transition between the spectrum above and below the designated frequency occurs at a fixed 6dB/octave rate. The gain or loss below the corner frequency is adjustable to +/- 15dB.

First Mic Priority Increases the audio level required to gate on additional microphones after the first mic is on. This helps ensure that only one mic gates on when a person speaks.

Gain The amount a signal is increased over a given reference, typically 0. Normally specified in dB (decibels). On the XAP 800/400, gain is adjustable from -65 to 20dB (85dB range) in .5dB increments.

Gain Structure The configuration of parameters which define gain adjustment of a signal. The optimal input gain setting is one which provides both an adequate signal-to-noise ratio and reasonable headroom.

Gate Ratio Specifies how much louder the microphone audio level must be above the ambient sound level before a microphone gates on.

GPIO (general purpose input/output) The Control/Status Ports A and B on the rear of the unit.

G-Ware Software The XAP setup and configuration software.

Hold Time The length of time that a microphone remains on after the voice (input) level drops below the gate ratio. This prevents the microphone from gating off during brief pauses in speech.

Last On Mode Sets the last-activated mic to Last On, Mic 1–8/1–4, or Off. The Last On setting leaves the last-activated mic gated on until another mic input gates on.

Macro A series of user-created instructions, stored within the unit, which can be executed from an RS-232 command or contact closure.

Macro Mode The section of G-Ware that allows you to customize and execute macro commands for a XAP unit or network.

Manual Gating Provide the ability to gate a microphone on or off manually.

Matrix Mixer A mixer that allows routing of any input or combination of inputs to an output or any combination of outputs. In the case of the XAP 800, the matrix mixer permits level control at each cross point in the matrix.

Maximum Number of Mics/Filibuster Sets the maximum number of mics that can be gated on simultaneously.

Microphone Activation A condition in which a microphone is gated on.

Microphone Mixing A situation in which audio from mics on different signal lines are mixed together. All mic signals can be mixed to one signal line, any or all outputs, or mixed into separate groups. Mic signals can also be processed together or individually.

Microphone 1 Mode Reverts mic assignment to a designated mic when all mics gate off.

Mute A condition in which an audio signal is attenuated below the audible threshold.

Noise Cancellation A process through which ambient noise is removed from a signal.

Number of Open Mics (NOM)/Constant Gain Mode Adjusts the output level based on the number of mics gated on and routed to an output.

Off Attenuation The amount of level reduction a microphone is given when the microphone is not gated on.

PA Adaptive Mode The XAP 800/400 recognizes how much loudspeaker audio is picked up by the microphones and then uses this level as the new ambient level when audio is present at the power amplifier. This prevents loudspeaker audio from gating ON a microphone, while still allowing people in the room to gate on microphones as they speak.

PA Adaptive Reference This G-Ware setting determines which output (typically for a loudspeaker) is used as a reference for an input.

Paging Zone A subset of a paging system. Intended to isolate paging system outputs to specific areas.

Parametric Equalizer (PEQ) A multi-band variable equalizer with control of gain, center frequency, and bandwidth. A properly configured PEQ enables the XAP to offset speaker or room acoustic deficiencies.

Phantom Power Power supplied by the XAP to power most condenser microphones. The XAP 800/400 provides 24 volts of phantom power. This feature can be switched off for devices not requiring phantom power.

Phase Plot A plot of phase angle (-180 to 180 degrees) vs frequency (20Hz to 20kHz) on a logarithmic scale.

Pink Noise An audio test signal containing all the frequencies in a given audio spectrum, with equal energy in each octave.

Preset One of 32 configurable memories in the XAP 800/400. A preset can be programmed with a variety of routing, level, gating, delay, filter, and equalizer settings to meet specific application requirements.

Preset Mask Defines whether a contact closure activates on a low or high signal on the rear panel of the unit.

Q Quality factor. It is the ratio of the center frequency divided by the bandwidth. Q reflects an inverse relationship to the bandwidth, and adjusts from .02:1 to 40:1 on the XAP 800/400.

Ratio The amount of compression applied to the output signal compared with the input signal as the signal exceeds the threshold level.

Release Release is a parameter which determines how quickly compression is released after the input signal drops below the threshold.

Reverberation A diffuse acoustic energy field fed and maintained by sound reflections from the room surfaces.

Serial Command A bit description designed to execute an instruction or command.

Signal Delay Used for:

1) Compensating for physical placement of multiple drivers in a cluster—to align points of acoustic origin in the same plane.

2) Ensure matching arrival times at listeners from multiple loudspeaker drivers.

3) Using the Haas effect to maintain localization of the source, even with a distributed system.

Signal Generator A device for generating a reference tone for sound system calibration purposes.

Speech Leveler Essentially an AGC applied to a speech signal after processing by a noise cancellation stage. Acts only on voice signals. Does not pull up the noise floor during periods of no signal.

Table View Displays the numerical values of the filter parameters for all nodes of the active filter display.

Threshold The upper or lower level at which a signal processing mechanism begins or terminates operation.

White Noise Acoustical noise with equal energy throughout a given frequency range.

ClearOne Communications ~ 1825 Research Way, Salt Lake City, UT 84119 ~ tel 1-800-945-7730 ~ fax 1-800-933-5107