Audio Perfect™

WHITEPAPER

VERSION3

Gentner®

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Introduction

Audio is critical to human communication. New communication media such as voice mail, the Internet, conference calling, videoconferencing and electronic presentations are driving the demand for better audio-communication technologies. At the same time, all organizations are looking for ways to decrease costs and complexity while increasing efficiency and productivity.

The Audio Perfect™ product offering was developed to respond to these needs. Quantitative and qualitative research methodologies, along with 16 years of real-world audio experience, were employed in defining Audio Perfect™. Gentner appreciates the input received from dealers, consultants, independent representatives, master distributors and end users who helped make the Audio Perfect™ product family a reality. Our overriding goal is to develop the right audio solution for you.

Audio is critical to electronic meetings. Over 70% of the information in a videoconference meeting, for example, is contained in the audio. If the video goes out, the meeting can proceed with just audio. If you lose the audio, the meeting is over. Additionally, the effectiveness of such meetings is greatly impacted by the quality of the audio. High reverberation and noise along with half-duplex audio will fatigue the users reducing the effectiveness of the medium and will, many times, drive users away from using the system permanently. Audio Perfect™ is dedicated to just that…making the audio as perfect as possible.

The driving applications for the AP product line include audioconferencing, videoconferencing, distance learning, board rooms, conference rooms, teletraining, telemedicine, court rooms and hotel/convention centers. Many of these applications are illustrated in this document. The key objectives for Audio Perfect™ are:

- Outstanding audio clarity and intelligibility
- Plug-and-play echo cancellation
- Ease of design and installation
- Seamless integration to external control devices
- Reduced complexity
- Reduced number of separate audio devices required
- Increased reliability and ease of troubleshooting
- Reduced cost of service and maintenance
- Expandability

Today, the Audio Perfect™ product line consists of the AP800 and AP10. This document outlines the features, functions, applications and technical details of these products. We have planned other new products and versions in this product family. You can discover the latest information by either calling us directly at 801.975.7200 or 800.945.7730, exploring our web site at www.gentner.com or accessing information from our fax back service at 800.695.8100 or 801.974.3661. In addition, we invite you to try our unique conference calling service 1-800-LETS MEET.™

1. Product Description - Features & Functions

The AP800 performs the functions of several audio devices in a single rack space, including an eight-channel automatic microphone mixer, a 12X12 matrix mixer, Distributed Echo Cancellation™, audio processing, equalization and audio control. The AP10 is a digital telephone hybrid that allows connection to a telephone line for audioconferencing, and is directly connected and controlled via the AP800.

AP800 Features and Functions

- Wide number of applications including audioconferencing, videoconferencing, distance learning, board rooms, conference rooms, teletraining, telemedicine, court rooms and hotel/convention centers
- Plug-and-play echo cancellation
- Each mic input has its own echo canceller (Distributed Echo Cancellation™) for a total of eight echo cancellers
- Distributed Echo Cancellation™ technology vastly superior to single echo canceller solutions
- 100% digital signal processor (DSP) implementation
- Simultaneous direct connection to several video CODECs and telephone lines (using an AP10 digital telephone hybrid)
- Simultaneous two-wire/four-wire and integrated dialing functionality through an AP10 digital telephone hybrid
- 12X12 matrix mixer, expandable to eight units for a total of 96 inputs and 96 outputs
- Two internal sub-mixing buses used for mixing and level control in sound reinforcement systems
- Twelve line output channels expandable to 96; any of the 12 input channels can be mixed to any of the 12 outputs; all output levels are adjustable and can be instantly muted
- Eight-channel automatic microphone mixer with four line inputs expandable to 64 mic inputs and 32 line inputs
- All automatic microphone functions and operating modes operate across expanded units
- Input gain, audio processing, equalization, muting, automatic mixer (and several other functions) programmable per input channel
- Expandable using a high-speed digital network bus (G-Link); a total of eight AP800s and 16 AP10s can be connected
- All G-Linked devices can be accessed, controlled and programmed via a single RS232 connection
- Program, operate and perform diagnostics from the front panel, a connected PC (direct or via modem) or any other type of serial remote control device
- Seamless integration to a custom remote controller and other remote control devices via a single RS232, even with multiple AP800 and AP10 units connected
- All functions can be remotely controlled using closures to ground including muting, audio volume and presets; status pins show status of key functions
- Instantly change complete configurations with presets
- Lock out of front panel access for security
- Easy to use using push-on/push-off terminal block connectors (Phoenix™ connectors)
- One rack unit

AP10 Digital Telephone Hybrid Features and Functions

- Direct connection to a telephone line
- Integrated dialing
- Provides simultaneous two wire/four wire operation within Audio Perfect™ system
- Conference up to 16 callers in an Audio Perfect™ system (using 16 AP10s)
- Auto-answer, auto-disconnect
- Controlled via the G-Link, allowing seamless integration with a custom remote controller or other similar remote control device

Features and Functions Common to All Audio Perfect™ Solutions

- Gentner service and support
- Meets and exceeds worldwide certifications for safety and emissions: CE, FCC, CSA, BABT registered
- One-year limited warranty

2. Applications

The following pages describe in detail several applications utilizing the AP800 and AP10 units.

Videoconferencing

In videoconferencing applications that require an outboard audio system, the AP800 executes the task with ease and simplicity. The AP800 connects directly to all microphones. One hundred percent digital technology means seamless activation of microphones, reducing reverberation and noise while the echo canceller cancels unwanted audio picked up by the microphones from the speakers. In this example, each microphone input is equipped with a mute button. This button can be programmed to mute a single microphone or all of the microphones on the system. VCR and other audio sources can be routed to and from the AP800 as shown. Audio outputs are delivered to a power amp, then to speakers in the room. Audio from the videoconferencing CODEC, the AP10 digital telephone hybrid and VCR audio can be heard through the speakers.

Connection to the video CODEC's four-wire circuit and an AP10 digital telephone hybrid allows for a simultaneous two wire/four wire connection (telephone and videoconferencing). Since the AP10 is G-Linked to the AP800, connection to the telephone line and dialing can be accomplished using a custom remote controller. In fact, all audio functions of the Audio Perfect™ system can be controlled from the remote control touch panel.

Need more CODEC connections? Each AP800 can easily accommodate up to four CODEC connections. Need more telephone connections? Simply add more AP10 digital telephone hybrids. Need more microphone channels? Simply add additional AP800 units. No matter how many components you add, the custom remote controller still only needs to make one serial connection.

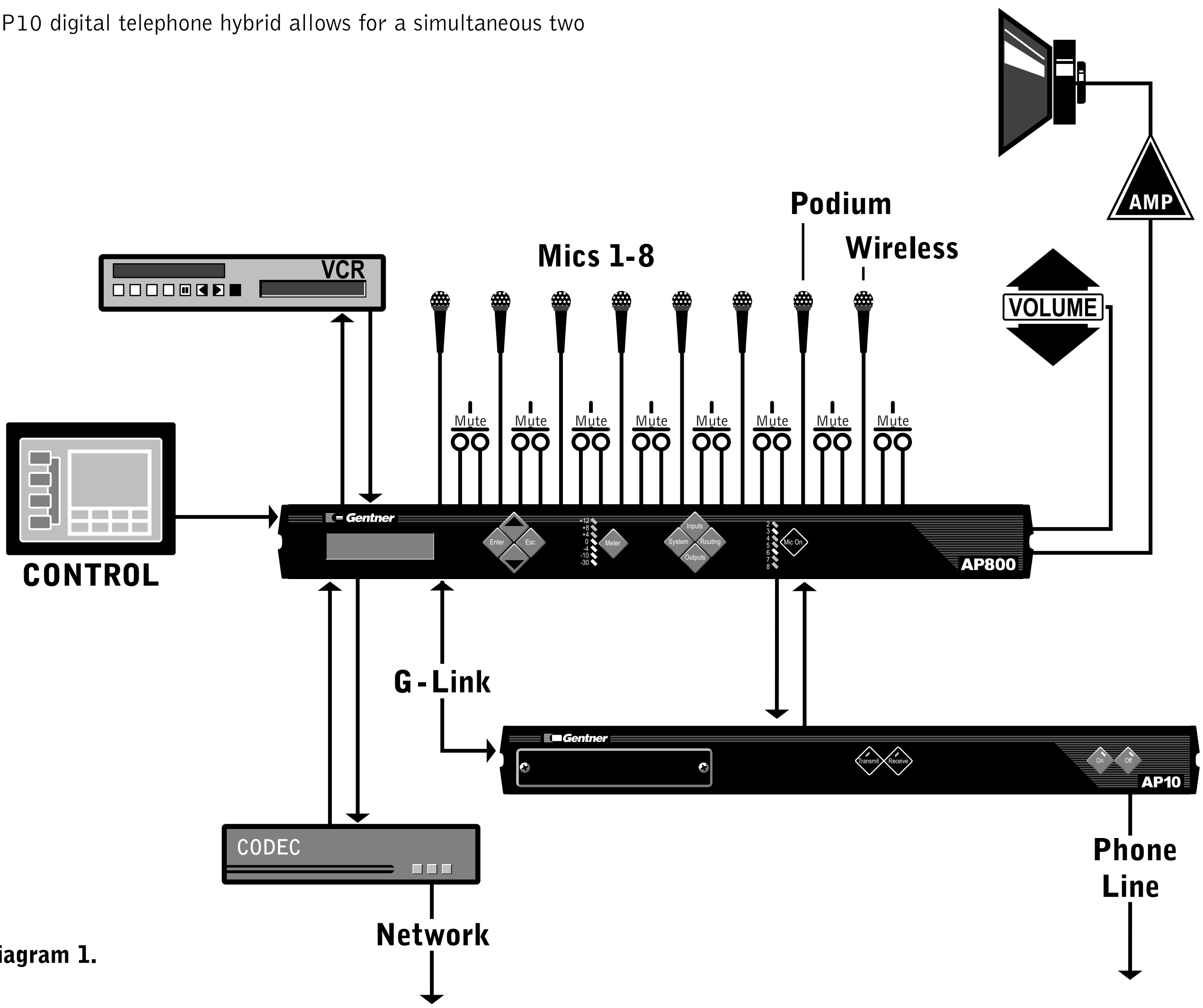


Diagram 1.

This example shows a fairly large board room with 13 table microphones, a podium mic and two wireless microphones. The AP800 will adapt well when using three different types of microphones because, with the built-in equalizer, you'll be able to equalize the microphones to sound similar. As a wireless mic moves around the room, its individual echo canceller will be able to converge quite nicely with the echo cancellers because the other channels (Distributed Echo Cancellation™) will remain converged, therefore, not affecting the system's full-duplex operation. The system uses sound reinforcement because of the large size. Audio from mic channels one through eight are heard in zone two and vice-versa.

This system has a CODEC and connection to three telephone lines using three AP10s. This allows for a three-person conference call. For larger conferences, more AP10 units could be added or you could call Gentner's conference calling service 1-800-LETS MEET™ to bridge the additional callers for you.

A touch screen is supplied for remote control. In addition, a volume control panel has been supplied, along with mute buttons, for direct user control of these functions.

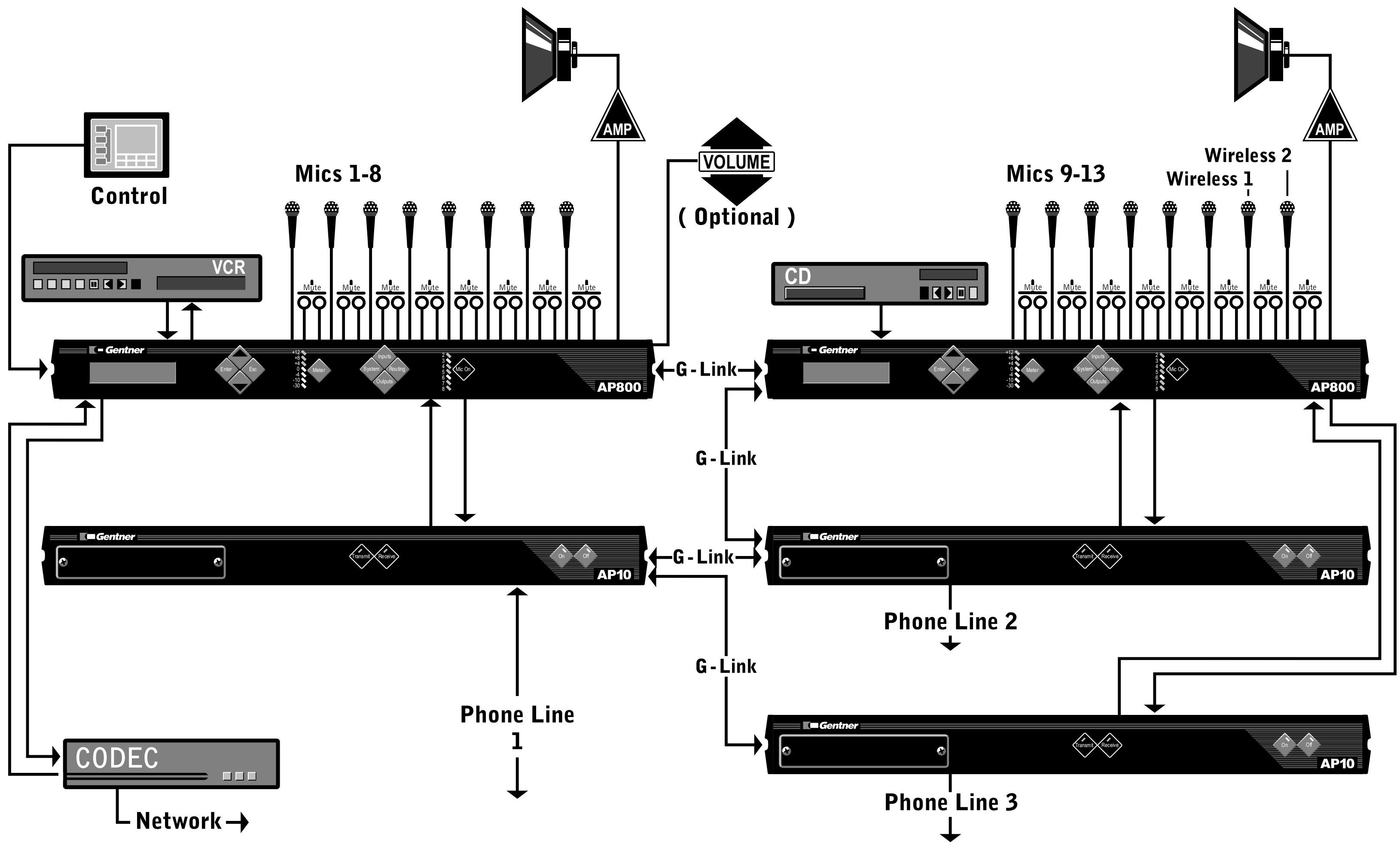


Diagram 2.

Courtroom

One box does it all ... automatic microphone mixing, direct outputs (gating or non-gating) for interfacing to the eight-channel court recorder, zoned sound reinforcement system, touch screen remote control, videoconferencing and telephone conferencing.

Case closed.

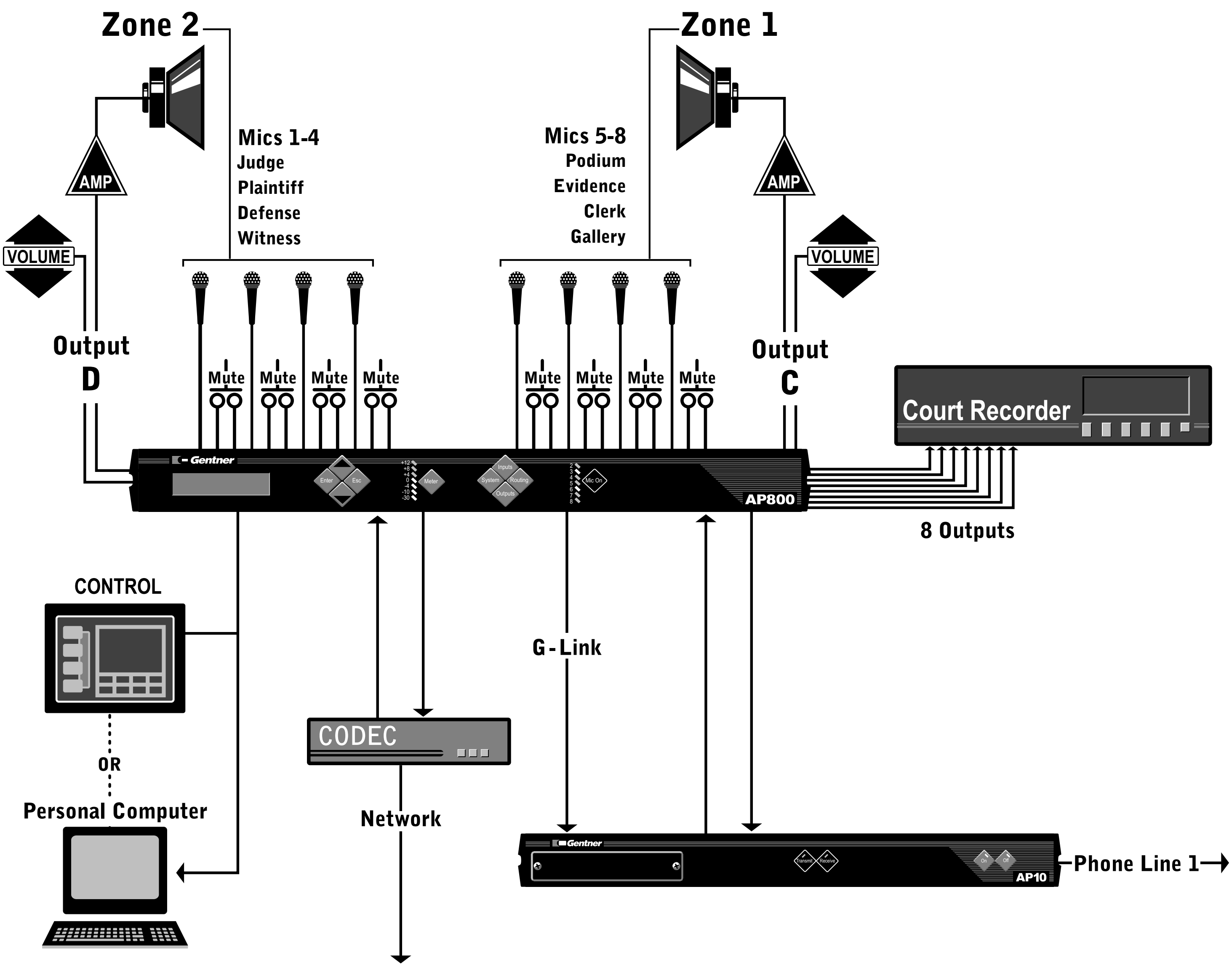


Diagram 3.

Here's an application where we've G-Linked three AP800 units and one AP10 unit together. The system uses 24 microphones, three zones of sound reinforcement, and connection to a video CODEC and a telephone line. Connect to the unit via modem to perform remote diagnostics.

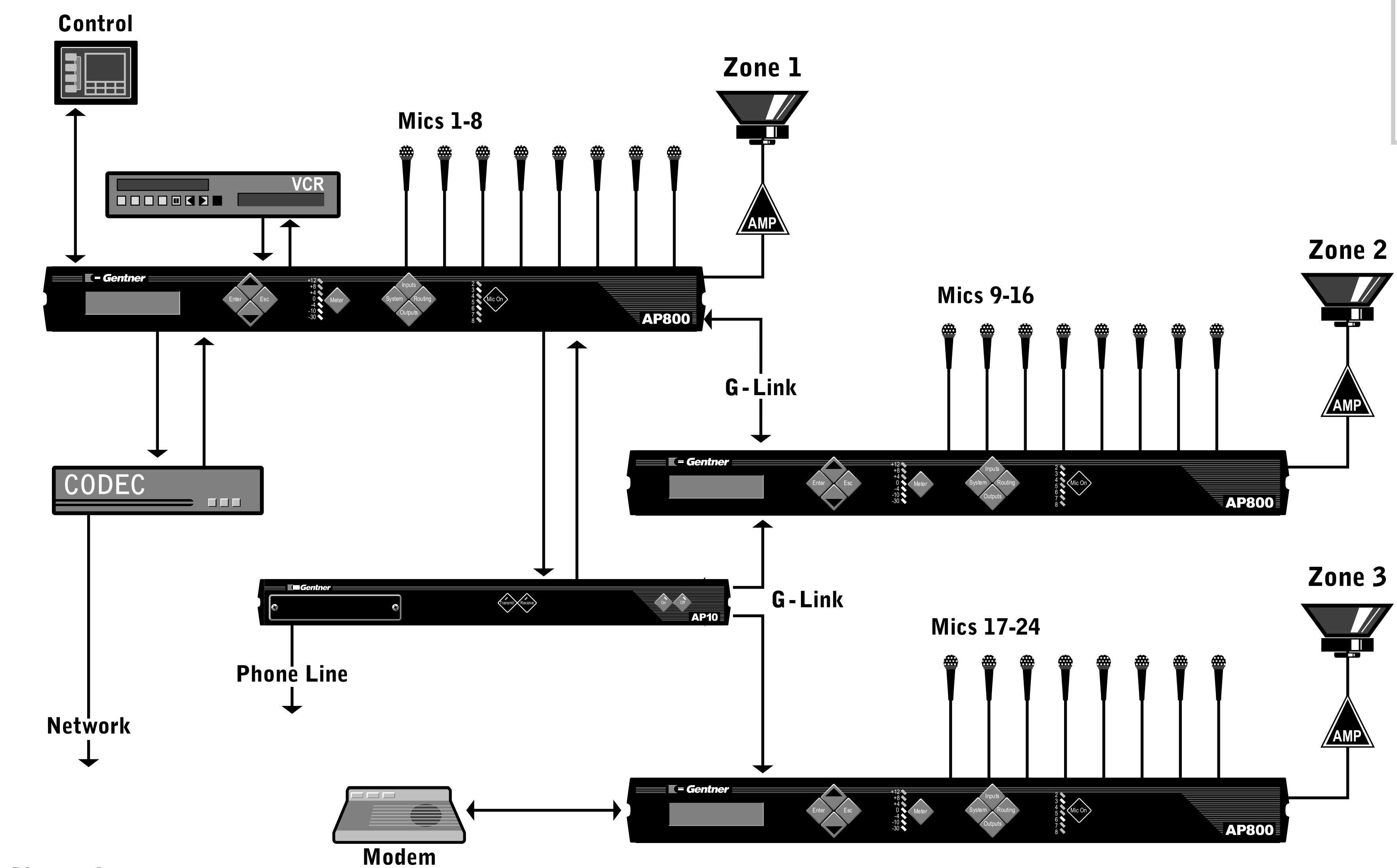


Diagram 4.

This is a room combining application. There are four different rooms in this application. Each room has eight microphones. The preset panel selects how the rooms are configured. When the room function changes, the user simply presses the control button and – presto!—new configuration. This is done using presets with the AP800 system. All the AP800 units and the AP10 are G-Linked together allowing control of the entire system. In this application, we're using a combination of custom remote controller and direct interface control for presets and volume.

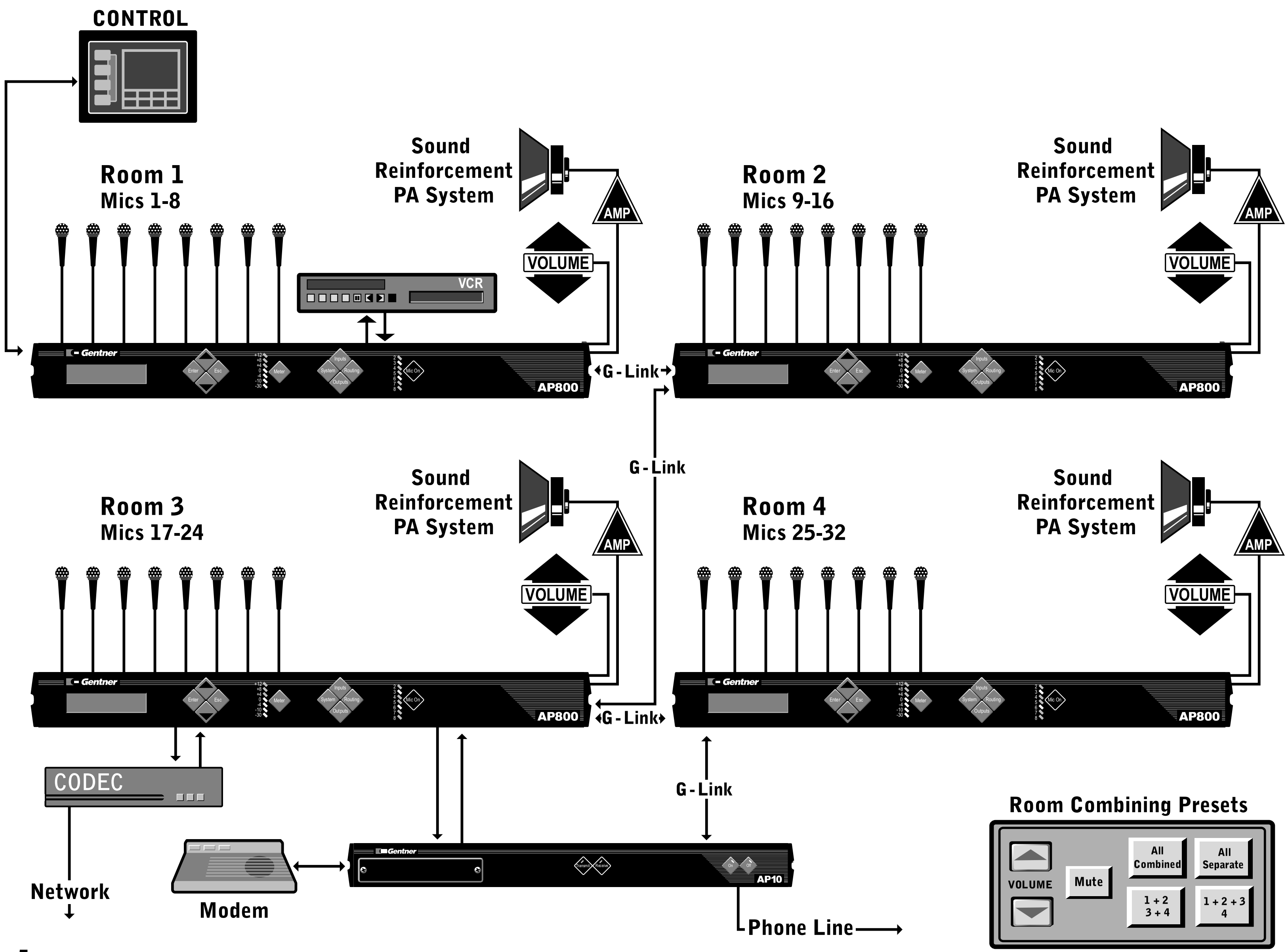


Diagram 5.

3. AP800 & AP10 System Block Diagrams, Description & Front and Rear Diagrams

The AP800 performs a variety of complex, integrated audio functions, while remaining surprisingly simple. This simplicity comes from implementing all functions digitally using digital signal processors (DSPs). The power of DSPs has allowed our design engineers to widen audio functionality for the AP800 and put it in a much smaller package. This product line was not designed as a generic solution for a wide variety of uses. Instead, it was designed to meet the specific audio needs of a limited number of applications (as shown in the previous section). By pursuing these vertical applications for the product and carefully listening to customers, the AP800 has been stripped of features and functions that would result in higher cost, complexity and reliability problems for those involved in design, installation, operation and ongoing product service.

In its most simple form, the AP800 is a microphone mixing matrix. As such, all microphone mixing parameters can be customized and any input or combination of inputs can be routed to any output, allowing flexibility in fitting different applications and customer requirements. Adjustments in routing, level and all other functions can be made at any time by one of three ways: front panel programming, presets (activated through a closure on the rear panel) and/or through a RS232 serial interface. This allows direct user control of functions such as mute, volume control, room combining, etc.

However, the AP800 is much more than a simple microphone mixing matrix. The following system block diagram (Diagram 6) shows the complete product functionality. The diagram following the system block diagram shows front/rear diagrams with indications of the function of each section (Diagrams 7 and 8). The system block diagram (Diagram 9) and front/rear diagrams of the AP10 (Diagrams 10 and 11) are also provided in the following pages.

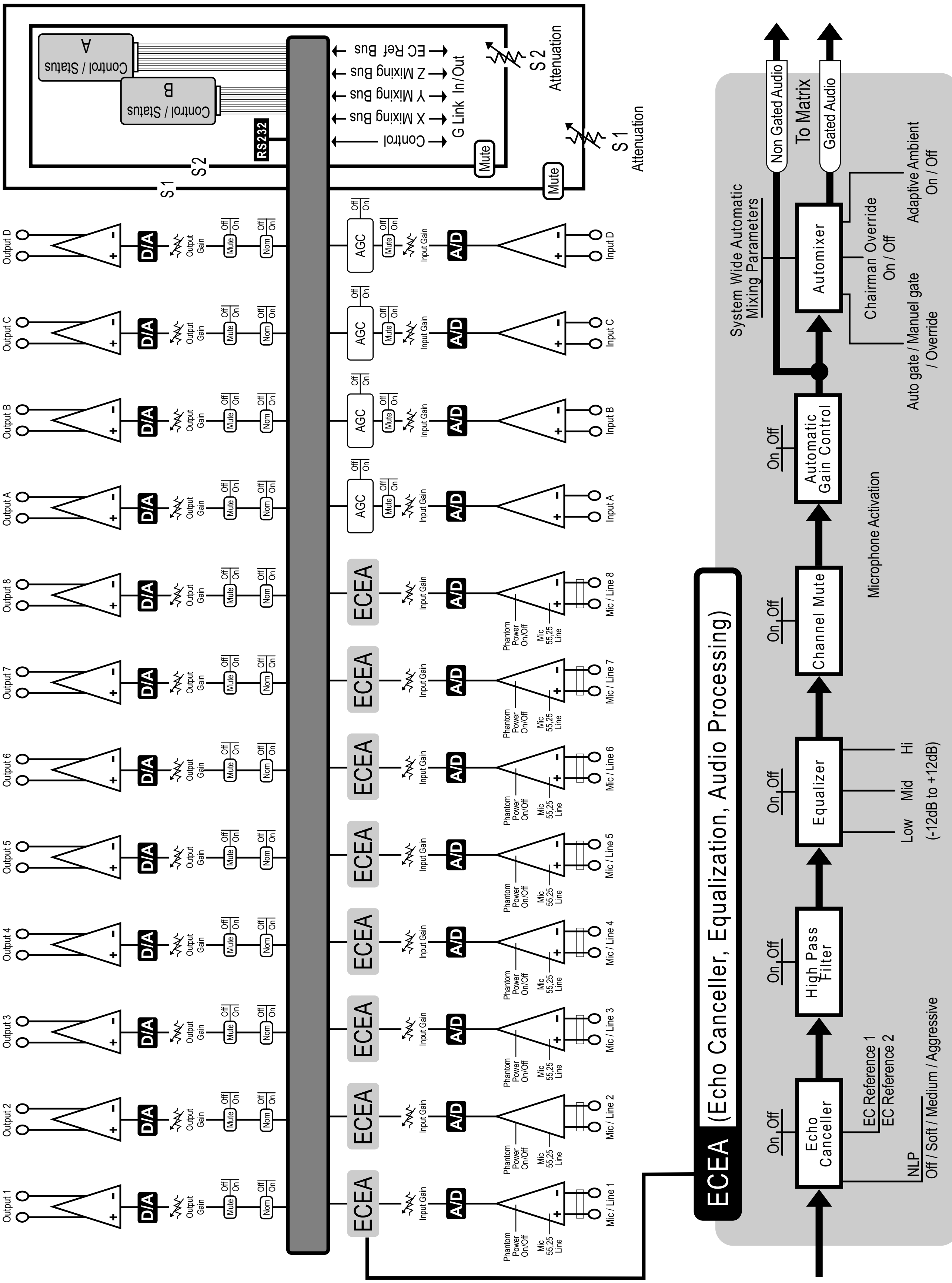

Control / Status A
Control / Status B
RS232
S 2
S 1
EC Ref Bus
Z Mixing Bus
Y Mixing Bus
X Mixing Bus
Control
G Link In/Out
Mute
S 2 Attenuation
S 1 Attenuation
Output 1
Output 2
Output 3
Output 4
Output 5
Output 6
Output 7
Output 8
Output A
Output B
Output C
Output D
D/A
Output Gain
Mute
Nom
Off On
AGC
ECEA
Input Gain
A/D
Phantom Power On/Off
Mic 55,25 Line
Mic / Line 1
Mic / Line 2
Mic / Line 3
Mic / Line 4
Mic / Line 5
Mic / Line 6
Mic / Line 7
Mic / Line 8
Input A
Input B
Input C
Input D
ECEA (Echo Canceller, Equalization, Audio Processing)
Echo Canceller
High Pass Filter
Equalizer
Channel Mute
Automatic Gain Control
Automixer
On Off
EC Reference 1
EC Reference 2
NLP
Off / Soft / Medium / Aggressive
Low Mid Hi
(-12dB to +12dB)
Microphone Activation
Auto gate / Manuel gate / Override
Chairman Override On / Off
Adaptive Ambient On / Off
System Wide Automatic Mixing Parameters
Non Gated Audio
Gated Audio
To Matrix

Diagram 6.

AP800 Front Panel

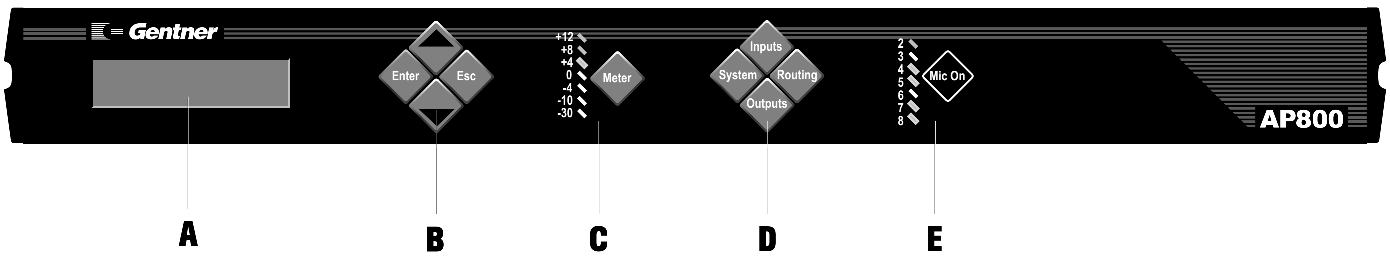


Diagram 7.

A. LCD Display. LCD display is used for setup, programming, troubleshooting and numeric audio level and gain readouts.

B. Enter/▲/▼/ESC Keys. These keys are used to navigate the AP800's easy-to-use menu system.

C. LED Meter. The LED meter displays the audio level of any input or output of the AP800, as well as echo return loss (ERL) and echo return loss enhancement (ERLE).

D. Quick Keys. Quick keys instantly access a programming menu.

E. Input LEDs: These LEDs indicate when a mic/line input channel is active.

AP800 Rear Panel

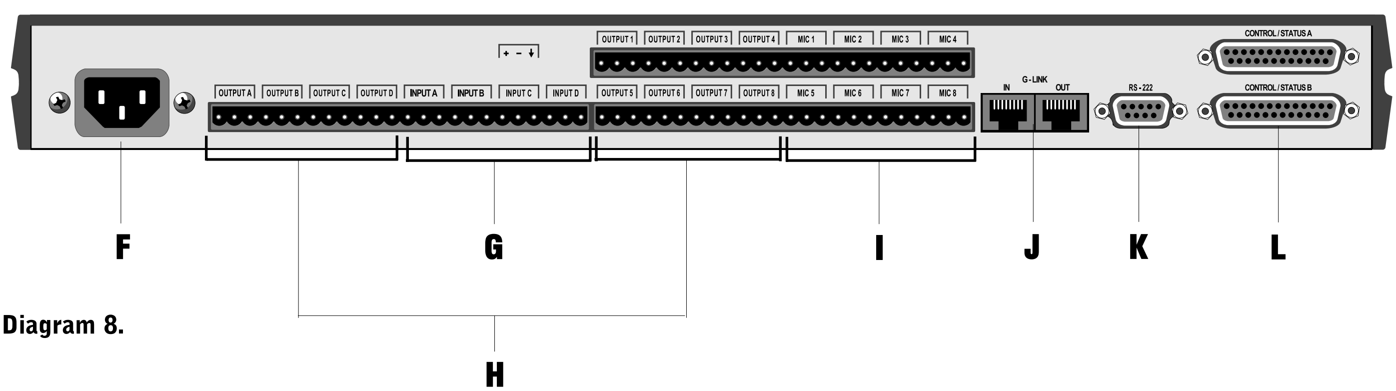


Diagram 8.

F. Power Supply. Internal switching power supply operates all worldwide power and frequency standards.

G. Line Inputs. Four line level only inputs.

H. Outputs. Twelve line level outputs. Any input and any combination of inputs can be routed to any output.

I. Mic/Line Inputs. The AP800 has eight balanced mic or line inputs. Phantom power can be activated for each input.

J. G-Link. With this high-speed network bus, you can connect up to eight AP800 units and 16 AP10 units.

K. RS232. This serial output can be used for programming, trouble shooting or control. Connect a modem and you can do it all remotely.

L. Control/Status A and B. These two connectors are used to interface parallel control to the AP800. A closure to ground can control any command on the unit such as volume, mute, etc. In addition, the AP800 has six presets that can be activated at any time. Status pins provide status of certain conditions such as on/off, mute, etc.

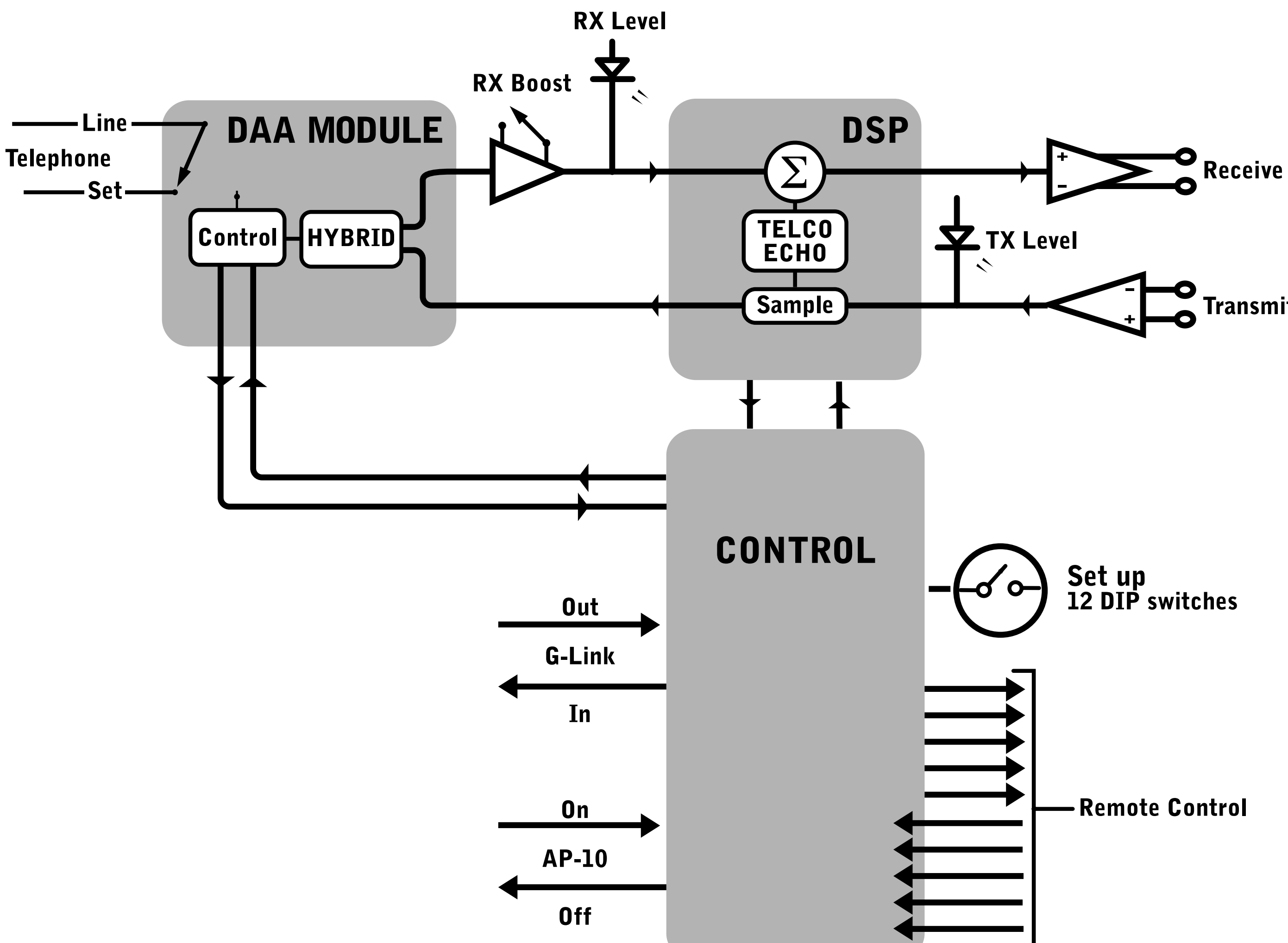

RX Level
RX Boost
Line
Telephone
Set
DAA MODULE
Control
HYBRID
DSP
Σ
TELCO ECHO
Sample
Receive
TX Level
Transmit
CONTROL
Set up
12 DIP switches
Out
G-Link
In
On
AP-10
Off
Remote Control

Diagram 9.

AP10 Front Panel

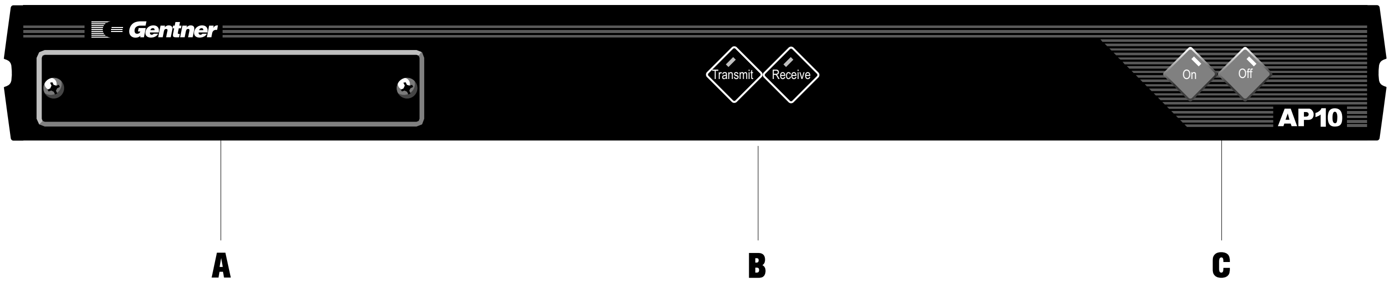


Diagram 10.

A. Operational features can be enabled or disabled via dip switches behind this panel. These features include auto-answer, auto-disconnect, momentary/latching mode, device identification, caller AGC, caller boost, noise burst adapt/self-adapt.

B. Transmit/Receive LEDs indicate audio presence on the telephone hybrid.

C. On/Off control and indication of status

AP10 Rear Panel

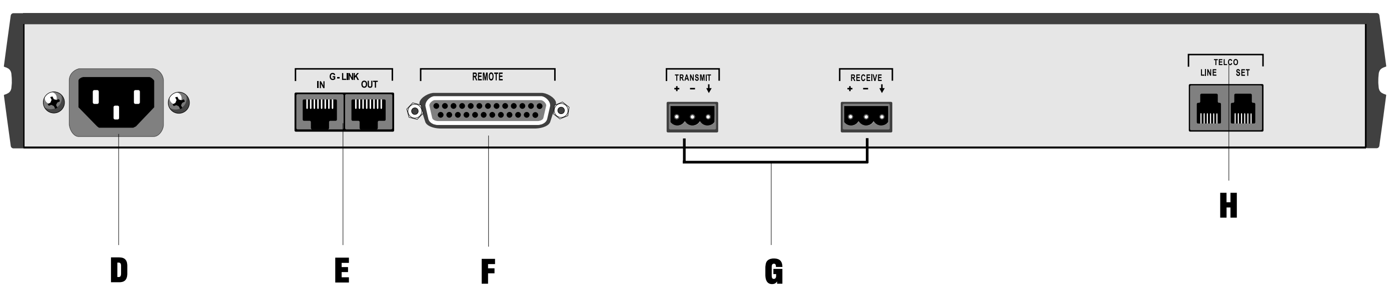


Diagram 11.

D. Power supply

E. G-Link in and out

F. Parallel Remote Control

G. Audio transmit input and audio receive output

H. Telco line and set connections

4. Distributed Echo Cancellation™

Echo can be a four-letter word. Audio picked up from the speakers and returned to the distant site in a teleconferencing application will destroy the participants' ability to effectively communicate. System-wide echo cancellers perform an adequate job of canceling unwanted echo in many applications. However, these products are often asked to perform echo cancellation in problematic acoustical environments where the acoustic echo canceller cannot fully cancel all echo.

There are a number of factors that contribute to poor echo canceller performance, including

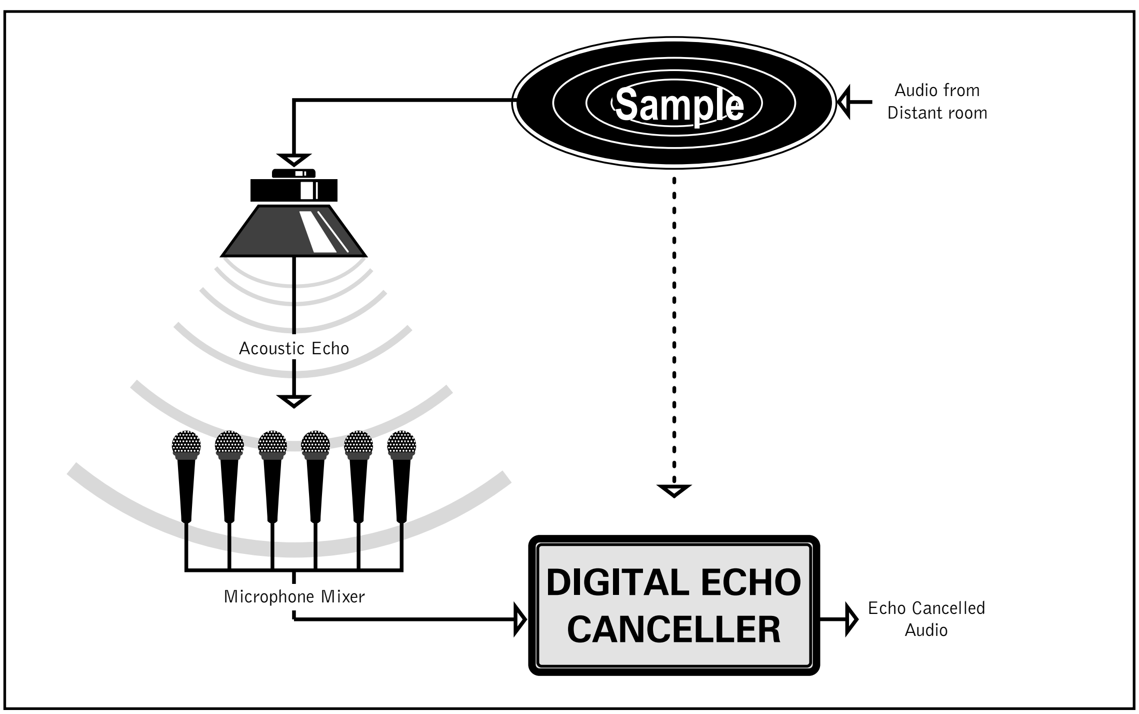
- Poor room acoustics
- High reverberation
- High noise
- Rapidly changing acoustical environment
- Wireless or other moving microphones
- Close proximity of microphone to speaker placement
- Automatic microphone mixers that are not properly set up to work with an echo canceller
- Other audio devices (such as audio processors, user gain controls, etc.) that produce changes in the acoustic gain that the echo canceller must adapt to

At first glance, it would seem that some returning echo would not be such a big problem. So what if you hear some of your own audio returning from the distant site? This is a true statement, unless there is audio delay in the system. In many cases, audio delay is caused by propagation (the actual time it takes to bidirectionally transport the audio from point A to point B and back) and by digital processing delays in video and audio CODECs. Human tests show that system delays over 20ms are enough for users to become psychologically impaired. Longer delays of well over 100ms are typical. In these cases, it is almost impossible to carry on a normal conversation. Echo in teleconferencing applications is not tolerated by users.

Many echo cancellers deploy compensating techniques that reduce the negative effects of echo that cannot be fully eliminated, such as center clipping and suppression. However, these techniques produce side effects that are objectionable, such as distortion in the audio, half-duplex operation, increased gritch and reduced audio levels. Sometimes the cure is worse than the symptoms!

For system integrators, echo can be both annoying and costly.

Diagram 12. A single echo canceller attempts to cancel the 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 (this is acoustic echo), the acoustic echo canceller senses the echo and builds an adaptive filter that eliminates or "cancels" the echo.



A system that is set up and calibrated today, then starts producing echo tomorrow can lose customer confidence and reduce or eliminate the profit on each job where echo is a problem. This is especially a problem in sites that are located long distances from the integrator's office. Diagram 11 shows how echo is typically cancelled.

Is there a better solution for echo cancellation? Yes! Distributed Echo Cancellation™ (DEC) is the answer. Here's how DEC works. 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 DEC system would have eight echo cancellers. Each echo canceller only has to work 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 canceled (suppression, center clipping, etc.), the compensation effects are only heard on the single microphone channel, rather than the entire mixed audio source. This greatly improves full duplex, noise, gritch and compensating audio level reduction. Diagram 12 shows how a DEC system works.

DEC technology could not be properly used until recently, because the cost of DSPs has decreased as the performance of such processors has increased. The AP800 is the first room audio system product to deploy such echo cancellation technology.

The advantages of Distributed Echo Cancellation™ include

- Significantly better echo cancellation in a wider variety of acoustical environments
- No training noise is required
- Faster convergence time
- Better full duplex
- Reduced gritch and suppression
- Increased audio levels
- Higher tolerance to room and network audio level changes.

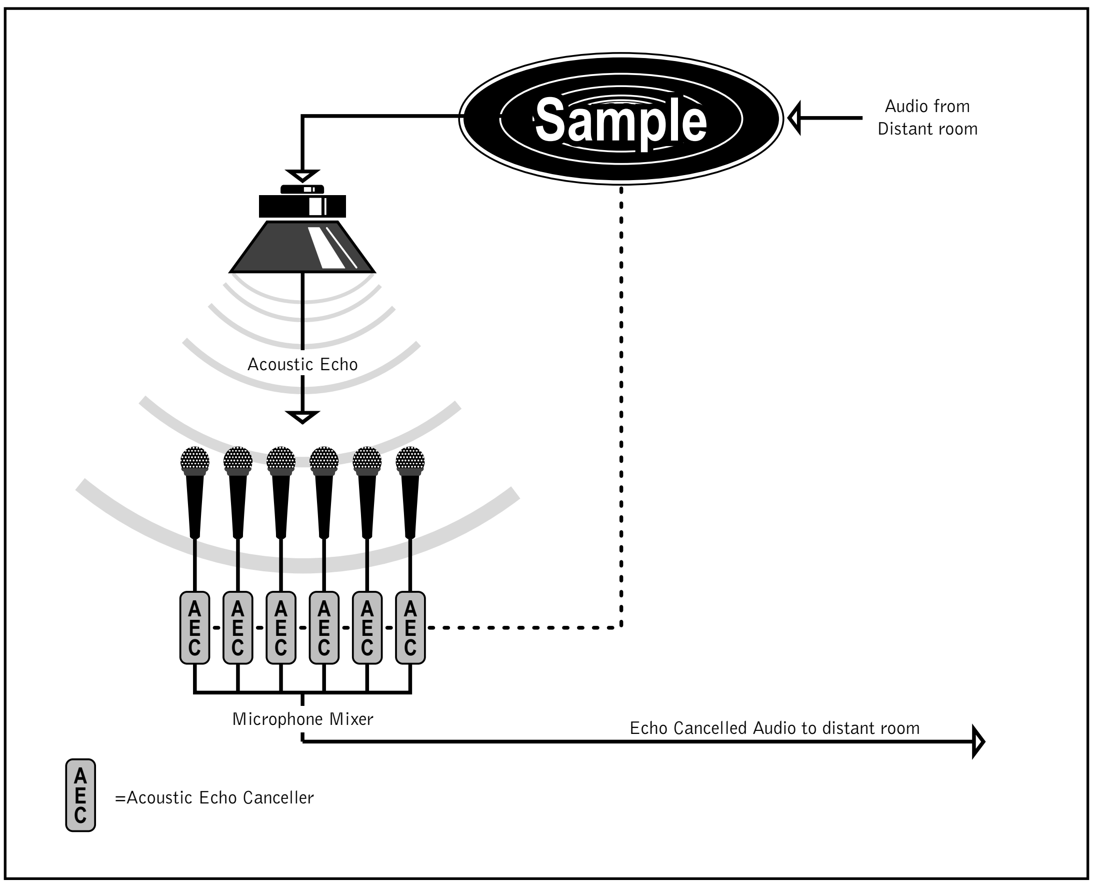


Diagram 13. Audio from a distant room is sampled. This audio is a reference for each echo canceller on every microphone. When sampled audio (acoustic echo) is detected by the echo canceller, it is eliminated or "cancelled."

5. Inputs & Outputs

The AP800 has 12 inputs, eight mic/line inputs and four line inputs as illustrated in Diagrams 14 and 15 below. The unit has 12 outputs as shown in Diagram 16.

All inputs and outputs are actively balanced. Mic/line inputs 1-8 have 4kOhms of terminating impedance while line-level inputs A,B,C,D provide >20kOhms of termination. Outputs provide a source impedance of 50Ohms. 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 +18dBu. The unit will deliver a maximum output level of +18dBm. The AP800 utilizes 18-bit A/Ds and D/As while sampling at a 32kHz rate. This results in a system wide dynamic range of 85dB and a pass band from 20Hz to 15kHz.

All input and output levels can be monitored in real time on the front-panel LCD and through the RS232 serial port. The LCD display and RS232 port provides 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 from the same LCD screen.

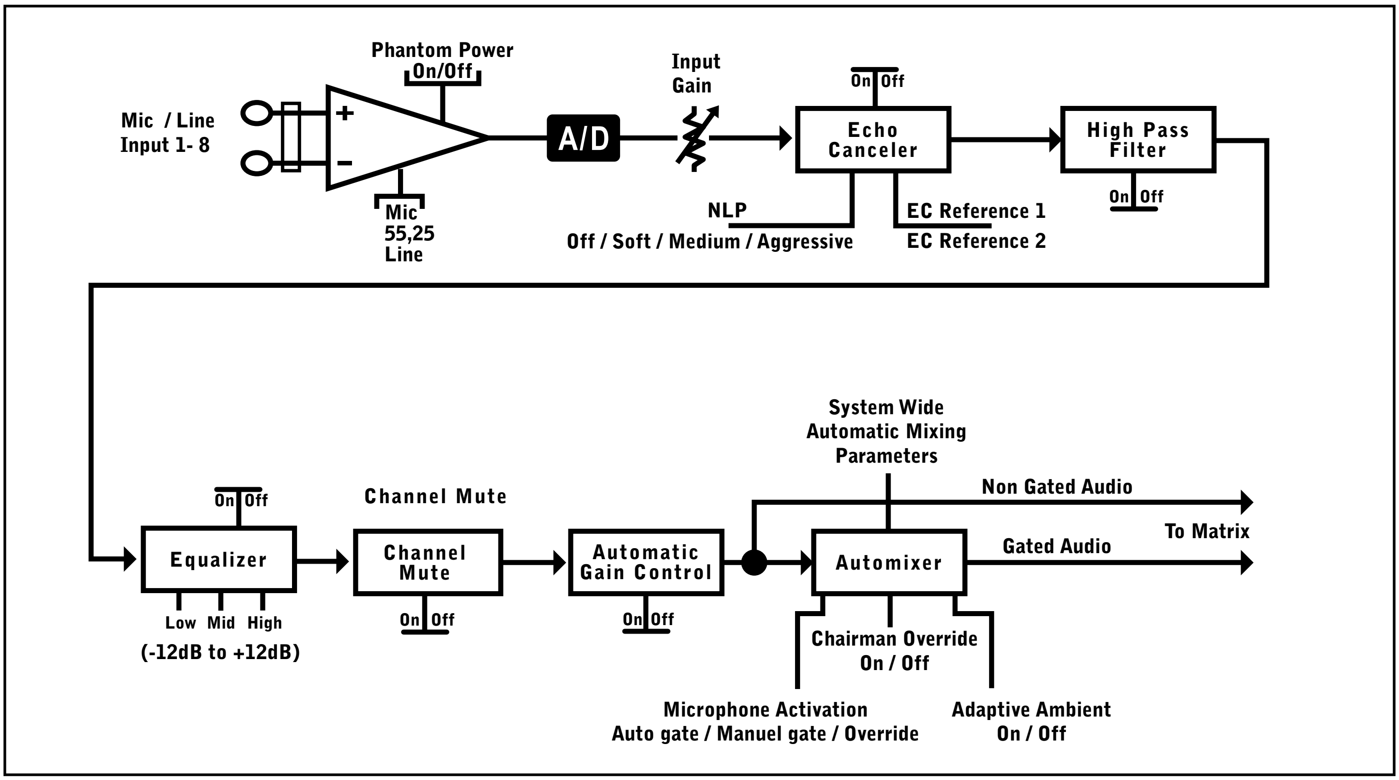


Diagram 14. This diagram illustrates all parameters of the eight mic/line inputs.

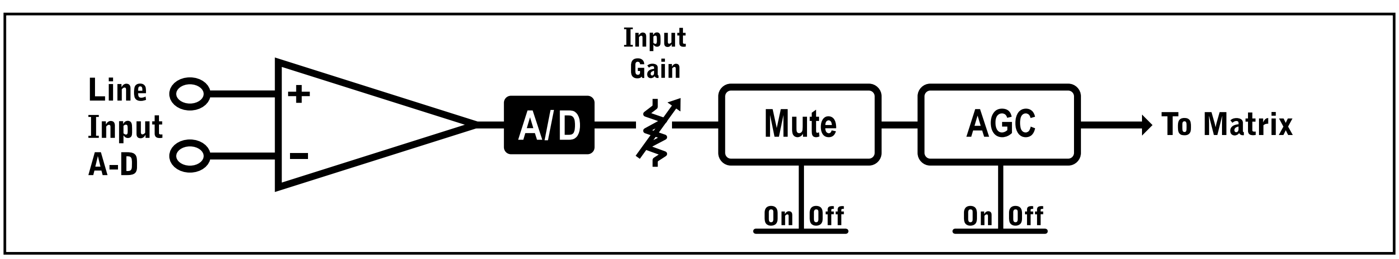


Diagram 15. Inputs A, B, C and D are shown in this diagram.

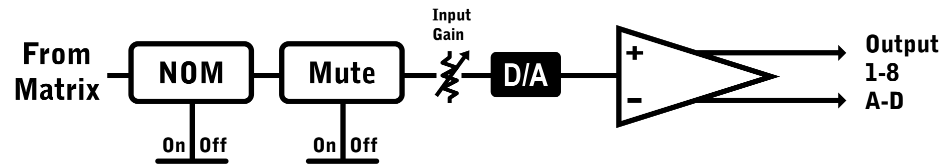


Diagram 16. Each of the 12 outputs of the AP800 are identical.

Mic/Line Inputs 1-8

Refer to Diagram 14. Balanced audio appears at the rear panel Phoenix™ connector. Mic or line level is selected and phantom power is provided (if required). The AP800 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 front panel, via the RS232 port, and/or via the control pins on the control/status connector. This provides for real-time audio volume control, muting, etc.

Next, the audio is echo cancelled if this function has been activated. If active, a reference for the echo canceller must be provided. The reference is the audio that is coming out of the speakers in the room and is being picked up by the microphone. You can choose reference 1 or 2. As previously discussed, the audio that makes up the two echo canceller references is programmed in the routing matrix (see default routing diagram). Non-linear processing (NLP) can be activated in three different levels. NLP adds additional echo cancelling "horsepower" to the echo canceller in difficult acoustical environments. Care should be taken when using NLP because of the corresponding trade-offs which potentially include suppression and half-duplex operation.

After echo cancellation, a high-pass filter may be activated to filter out unwanted hum. Next the three-band audio equalizer may be applied to the audio signal. When different microphones are used in the same room, the equalizer can be used to make all the mics sound similar. You have the option of increasing or decreasing each band up to 12dB in increments of 1dB on each mic/line input. Then the channel mute function is applied.

The next option is the automatic gain control (AGC). 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. 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. In the past, AGC could not be used in conjunction with an echo canceller because the echo canceller could not keep up in adapting to the fluctuating AGC audio levels. On the AP800, AGC is under the control of the DSPs which allow the echo cancellers to know precisely what the AGC is doing. Thus, AGC does not negatively affect the functioning of the AP800's echo cancellers.

At this point, 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 mic 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 three modes of mic activation that can be selected on a per-mic basis: auto-gate, manual gate on/off and gate override on/off. In auto-gate mode, the mic channel is 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 mic to contribute to automixing parameters. In gate override mode, the mic is forced on or off and will not contribute to the automixing parameters.
- **Chairman override (On or Off).** Each gated input may be selected as a chairman override microphone. This feature, when selected, adds this input to the chairman override group and, when gated on, all other gated inputs that are not in the chairman override group will gate off.
- **Adaptive Ambient (On or Off).** In the ON mode, the ambient level used to calculate microphone gating will be based on the room's actual noise floor, integrated over time, as measured by the microphone in the room. In the OFF mode, the manual ambient level set by the integrator will be used to calculate microphone gating.

Line Inputs

Diagram 15 shows a line only input channel. These inputs can be level controlled, gain controlled through the AGC and muted as indicated. All of these functions operate identically to the mic/line inputs.

Line Outputs

All 12 of the line outputs are identical and are shown in Diagram 16. Three functions are associated with each output: gain control, mute and NOM. Gain control allows you to set the output level. The mute function essentially turns the volume off. Again, all of these functions can be controlled via the front panel, via the RS232 port and via the control/status connector. Thus if you want to control the volume of the speakers, you use two control pins on the control/status connector for volume up and volume down. Another pin could be used for mute.

Activation on NOM places this output ONLY in the constant gain mode. In this mode, as more microphones are gated on (either by auto gate or manual gate), the total gain remains the same. An exclusive feature of the AP800 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 reinforced applications.

6. Automatic Microphone Mixing

The best audio systems are those designed with the user in mind. Audio room systems are in constant use in board rooms, class rooms, court rooms and many other applications. Audio is the critical component of effective communication. The following objectives are key to end users:

- The audio must be transparent. Users should not have to even 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 and will fatigue the user producing a less-than-effective outcome. We all have enough things in our lives that drain our resources and fatigue our minds and bodies. A poorly executed audio system shouldn't be one of them.
- 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 audio. When used with directional microphones, an automatic microphone mixer will reduce reverberation and noise — the two major culprits in making it difficult to understand voice communications. In Diagram 17 (on the next page), direct audio from a person's voice is picked up by several microphones and a microphone mixer that has all microphones on at all times.

In addition, reflected audio (reverberation) is picked up by all microphones. Thus, what you hear is a combination of audio sources: direct audio and reverberated audio. 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, our brain has a difficult time understanding the audio.

We have all experienced trying to speak in a room that has a lot of reverberation – it's darned 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.

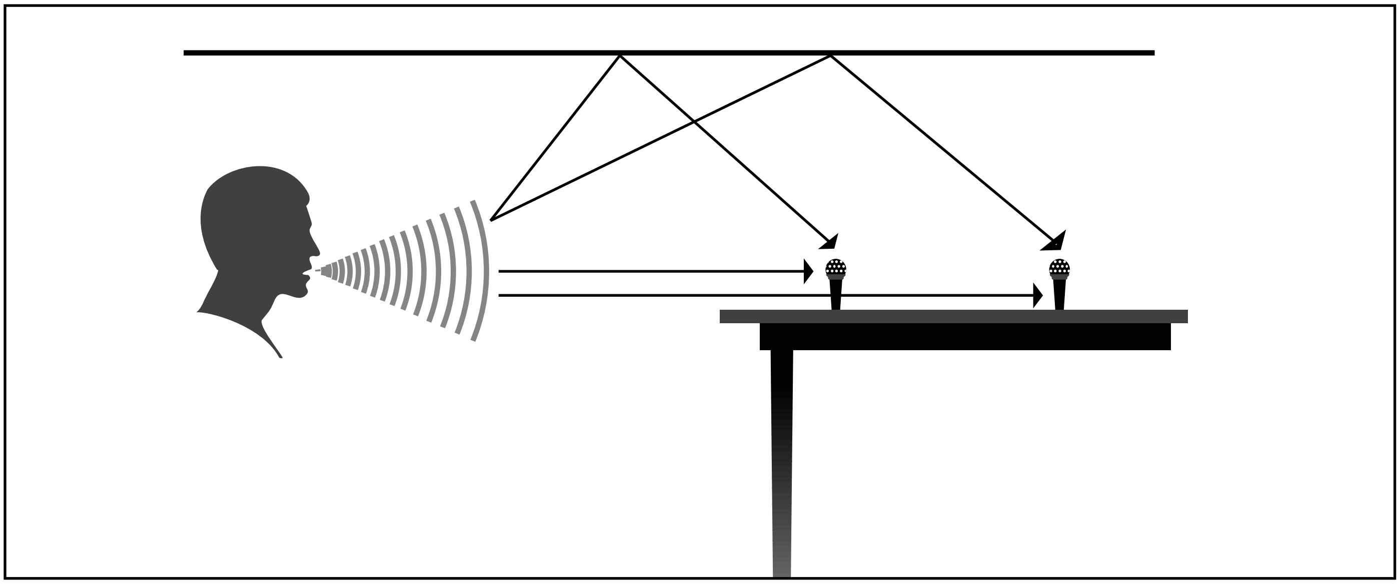


Diagram 17. Direct audio from a participant's voice, as well as the reverberation, is picked up by all the microphones in the room.

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

- Keep microphones close to the participants.
- Only activate those microphones where voice audio is present.
- Use directional microphones that only pick up the audio where someone is actually speaking.
- Acoustically treat the room to reduce reverberation and noise.
- Eliminate or reduce the source of the noise.

The AP800 was designed to implement automatic microphone mixing that increases audio intelligibility by reducing reverberation and noise. Unlike most automixers, the AP800 implements its mixing function 100 percent in the digital domain. This greatly increases precision in making automixing decisions.

One of the biggest problems in using an automixer in room audio systems is in degrading the performance of the echo canceller. Since automixers, by their very nature, radically and continually change the acoustic gain of the audio system, the echo canceller must attempt to track those acoustic gain changes. Some automatic microphone mixers have been modified to work well with echo cancellers, such as Gentner's MPAII.

The AP800 takes a completely different approach. Since AP800 functions are implemented in DSP, audio information regarding automixing functions is available to the acoustic echo canceller in the device. Thus, the echo canceller no longer has to attempt to track the automixer, because it already knows precisely what it's doing! In addition, other automixing technologies previously implemented in the analog domain can be executed in the digital domain with far better accuracy.

Another important point about the AP800's automixing functions is that, since all audio is routed through the AP800 (both microphone and speaker audio), the AP800 can more accurately make microphone activation decisions. For example, audio from another source (such as music or audio from another room) is amplified through the speakers in the room. In a typical automixer, the mixer would activate at least one microphone, thinking that audio is a voice in the room. This "false activation" will not occur with the AP800, (as shown in Diagram 17) because the unit "knows" that this audio is not voice audio. A few milliseconds before this audio hit the microphone, the AP800 sensed it.

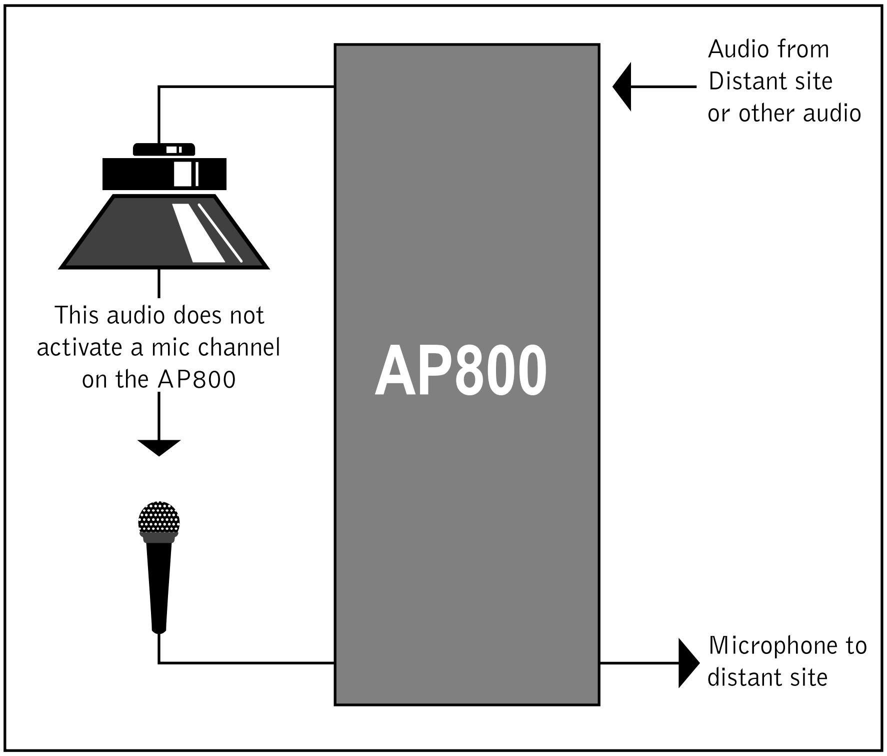


Diagram 18. Most automixers will mistakenly activate microphone channels when audio from the distant site or other audio sources is heard through the speaker. Since this audio information is available to the AP800, it will not make this mistake.

The AP800 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 page. Each AP800 can have two separate automatic mixers working independently or as a single unit. In addition, more microphone channels can be added by linking AP800 units via the G-Link, the digital network bus. Unlike other "expandable" automatic microphone mixers, the AP800 works as a single unit for up to eight units networked together for a total of 64 microphones. Expanded analog automixers can offer only limited functionality such as NOM (number of open microphones). Multiple AP800 units can operate as a single unit because all functions are implemented digitally and all units are connected

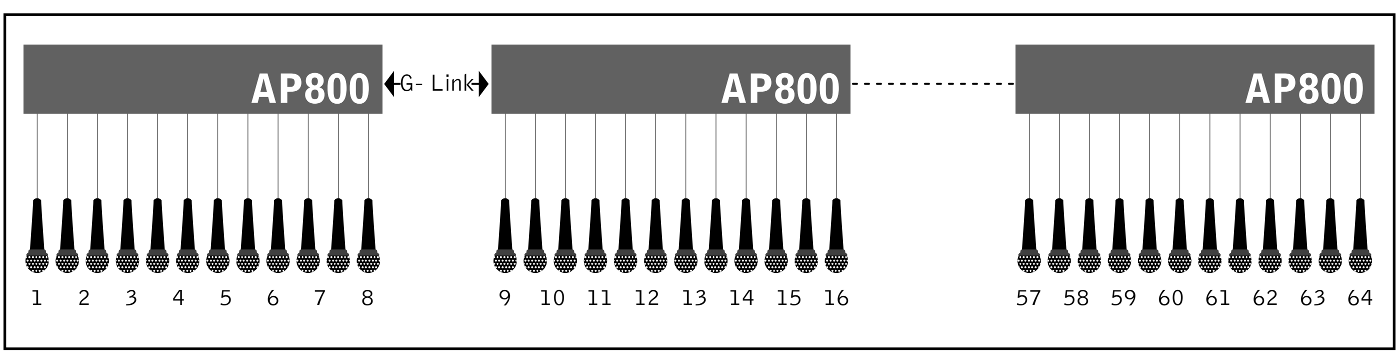


Diagram 19. The AP800 can automatically mix up to 64 microphones and works virtually as a single automatic microphone mixer.

The following parameters and modes are used on the AP800 to provide high precision and reliability in microphone mixing:

- **Mixer Mode.** The AP800 can be set in four different mixer modes to accommodate a variety of installation needs. When placed in the *Master-Single* mode, the unit acts as a stand alone eight channel automatic microphone mixer. In the Dual mode, it becomes two four-channel automatic microphone mixers with mics 1-4 on unit 1 and 5-8 on unit 2. When using multiple AP800 units G-Linked together, the *Master-Linked* mode is used for the first AP800 and the *Slave* mode is used for all other units.
- **PA Adaptive (On or Off).** First the problem: speaker audio gates on microphones when it shouldn't. Now the solution: PA adaptive knows when speaker channels are activated and prevents the speakers from gating the mics on. This is a Gentner exclusive. As illustrated in Diagram 18, the reason the AP800 can accurately determine when speaker audio is present because that audio actual goes through the AP800 a few milliseconds before it reaches the microphone. This is because audio travels through air at the approximate speed of one foot per millisecond. Say the speaker to microphone distance is three feet, it will take at least three milliseconds for the audio to travel from the speaker to the microphone input. That's more than enough time for the AP800 to determine that the audio is not voice audio and to make the decision <u>not</u> to activate the microphone channel.
- **Maximum Number of Microphones On (1-8 or Off).** If you dropped a book on the table in a typical automixing configuration, all of the microphones would turn on. If you think about it, there is really no reason for more than two people to speak at the same time ... OK maybe three in heated conversation. So why turn on more than three microphones? This mode allows you to program how many microphones (maximum) can be activated simultaneously.
- **First Mic Priority Mode (On or Off).** This is a useful feature for even better precision in microphone activation. Here's how this mode works: let's say someone is speaking into microphone 3, activating that channel. Typically, when someone is speaking and someone wants to say something, he or she usually waits for the other person to stop talking before speaking. However, if someone really wants to "make a point" or "get a word in edgewise," he or she will increase the volume of their voice to prompt the other person to give a chance to speak. When this mode is on, it requires more audio level for the second microphone to activate. So, if someone is trying to speak on microphone 7, it will take a higher level to activate that channel. The AP800 takes advantage of this higher level when the first mic priority mode is activated.
- **Last Mic On/Mic1/Off.** This leaves the last activated mic on until a new one is activated. Mic 1 mode reverts back to Mic 1 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. Optionally, you set this parameter to Off which disables this function altogether.

  (See Diagram 20)

- **Gate Ratio Adjust (0 to 50dB).** 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 more than 35dB of audio above the ambient level in the room to activate the microphone. The ambient audio level can be specified or the adaptive ambient mode can be turned on (see below). In this case, the ambient room level changes or adapts as the noise floor changes.
- **Off Attenuation (0 to 50dB).** Sets how much the microphone will be attenuated when it is not activated.
- **Hold Time (.1 to 8.0 seconds).** Programs the amount of time it takes until the microphone input starts the off attenuation process.
- **Decay Rate (slow, medium, fast).** Programs how quickly the audio level is attenuated once a channel has been gated off.
- **Manual Ambient Level (0 to -70dB).** 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 AP800 solution set. Because all decisions regarding acoustic echo cancellation and automixing are made by the same digital engine, better decisions in echo cancellation and automixing can be made. In addition, since a single DSP is dedicated to each input (this is required in DEC) this processing horsepower can also be utilized to make automixing decisions.

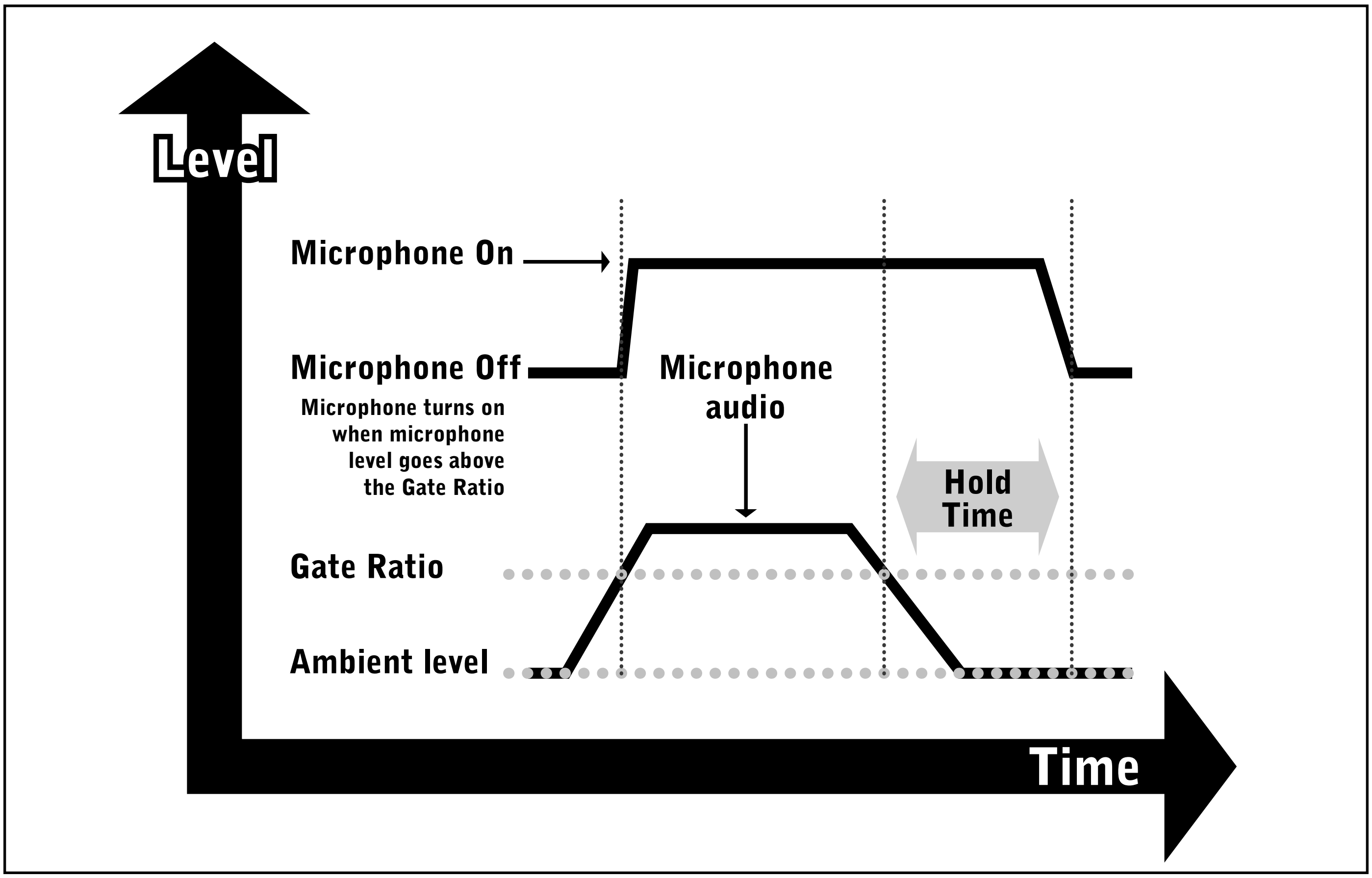


Diagram 20. AP800 Automixing Gate Functions

The following matrix summarizes all programming automixing functions of the AP800.

Mixing Parameter	Channel or System-wide Functionality	Range	Description
Mixer Mode	System-wide	Master-single, master-linked, slave, dual	Selects mixer mode of operation.
Microphone Activation	Input mic/line channel	Auto Gate, Manual Gate On/Off, Gate Over-ride On/Off	Sets the method of microphone gating.
Chairman Over-ride	Input mic/line channel	On, Off	When a chairman over-ride channel is gated on, all non-chairman channels are gated off.
Adaptive Ambient Mode	Input mic/line channel	On, Off	Automatically sets the ambient audio level of the room averaged over time.
PA Adaptive Mode	System-wide	On, Off	This Gentner exclusive, prevents mic channels from gating when distant audio or other non-microphone audio is heard through the speakers.
Maximum number of Microphones On	System-wide	1 through 8 or Off	Sets the maximum number of micro phones allowed to be gated on at a time.
First Mic Priority Mode	System-wide	On, Off	Increases the audio level required to gate on microphones after the first mic is on.
Last Mic Mode	System-wide	Last, Mic 1, Off	Keeps the last gated microphone or Mic 1 on when no mics are providing a gating input.
Gate Ratio Adjust	System-wide	0 to 50dB	Specifies how much louder above the ambient level the audio level must be to gate on.
Off Attenuation Adjust	System-wide	0 to 50dB	Sets how much the microphone will be attenuated when it is not gated.
Hold Time	System-wide	.1 to 8.0 Seconds	Programs the amount of time it takes until the microphone input starts the attenuation process.
Decay Rate	System-wide	Slow, Medium, Fast	Programs how quickly the audio level is attenuated once a channel has been gated off.
Manual Ambient Level	System-wide	0 to -70dB	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 select ed output. As more mics gate on, each mic is appropriately attenuated.

7. Audio Routing

One of the more important functions of the AP800 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 from the front panel, via the RS232 port and/or via presets on the control/status connectors.

The AP800 audio matrix has 25 possible input sources and 17 output destinations. The routing chart on the next page describes the default AP800 routing. Inputs and outputs are labeled for this default routing diagram, but any input and output scheme could be used. To ensure understanding, inputs and outputs to the matrix are described on the following page.

Inputs

Mic/Line Gated and Non-gated Inputs - Mic or line inputs appear 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 court room), direct, non-gated outputs are required. Default routing for gated microphone inputs are to outputs A, B, C and to the X-Bus. Non-gated outputs are routed by default to their corresponding output number (i.e. mic 1 is routed to output 1).

Inputs A, B, C, D - These are line-level inputs that appear on the rear-panel terminal blocks. This is typically audio that comes from a video CODEC, AP10 digital hybrid, VCR and other auxiliary audio sources. In typical applications, this audio must be heard in the local PA system (as well as networked AP800 units) and must be a reference for each distributed echo canceller, including those of other AP800 units on the network (see section 6). In the default routing, audio is routed to every other device except itself. Example: Video CODEC audio is routed to the input of an AP10 digital hybrid and VCR, but NOT itself.

Outputs

Outputs 1 through 8 - These are exactly the same as outputs A, B, C and D. Their default routing is for each non-gated mic/line input to go directly to these outputs.

Outputs A, B, C, D - These are line-level outputs that appear on the rear-panel terminal blocks. This is typically audio that goes to a video CODEC, AP10 digital hybrid, VCR and other auxiliary audio sources. Typically, this audio contains a mix of the microphones, auxiliary audio and audio from other networked AP800/AP10 units. In the default routing, inputs A-D (minus your channel input) are contained in this audio along with all microphones, master microphone mix (all microphone audio from other AP units) and master auxiliary mix (all auxiliary audio from other AP units).

G-Link

G-Link Buses - This is a digital bus that appears at every AP800 and AP10 networked on the system. This 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 AP800 can be placed on a bus or audio can be taken off a bus and routed to any destination within the unit. The AP system has three such digital mix-minus buses with the following default programming:

X-Bus - This bus is defaulted as the MASTER MICROPHONE MIX. All gated microphones are routed to this bus.

Y-Bus - This is defaulted as the AUXILIARY MIX. All auxiliary audio such as video CODEC, AP10 hybrid, VCR and other devices.

Z-Bus - This is a user-defined auxiliary mix-minus bus.

G-Link EC Reference Bus - This bus works identically to the X, Y and Z buses. This bus provides a system-wide bus for mic channels to receive a reference input for their echo canceller. Here's an example to clarify: Let's say you have four AP800 units G-Linked together. Audio on output D of unit 1 is audio routed to the PA system in the room. This audio is needed as a reference to the echo cancellers for mics on units 2, 3 and 4. This is accomplished by routing output D on unit 1 to the G-Link EC Reference Bus and then routing the output of the G-Link EC Reference Bus to the input of EC Reference 1 on units 2, 3 and 4.

Sub-bus

Sub-bus S1 and S2 - These are internal sub-mixing buses. There are a total of two sub-buses allowing two zoned outputs per AP800. The two sub-buses are used to select and mix various audio sources together. Next, these combined audio sources are attenuated in level (Diagram 21) and delivered back to the matrix for further routing. Typically, this is used in sound reinforcement applications where separate level control is needed. Example: Microphones 1-4 are routed to sub-bus 1 and microphones 5-8 are routed sub-bus 2. Audio levels on the two buses are attenuated for optimum sound reinforcement. Next, the sub-bus is routed to the output that drives the PA system. Default routing and level control for the sub-buses will depend upon the preset programming selected.

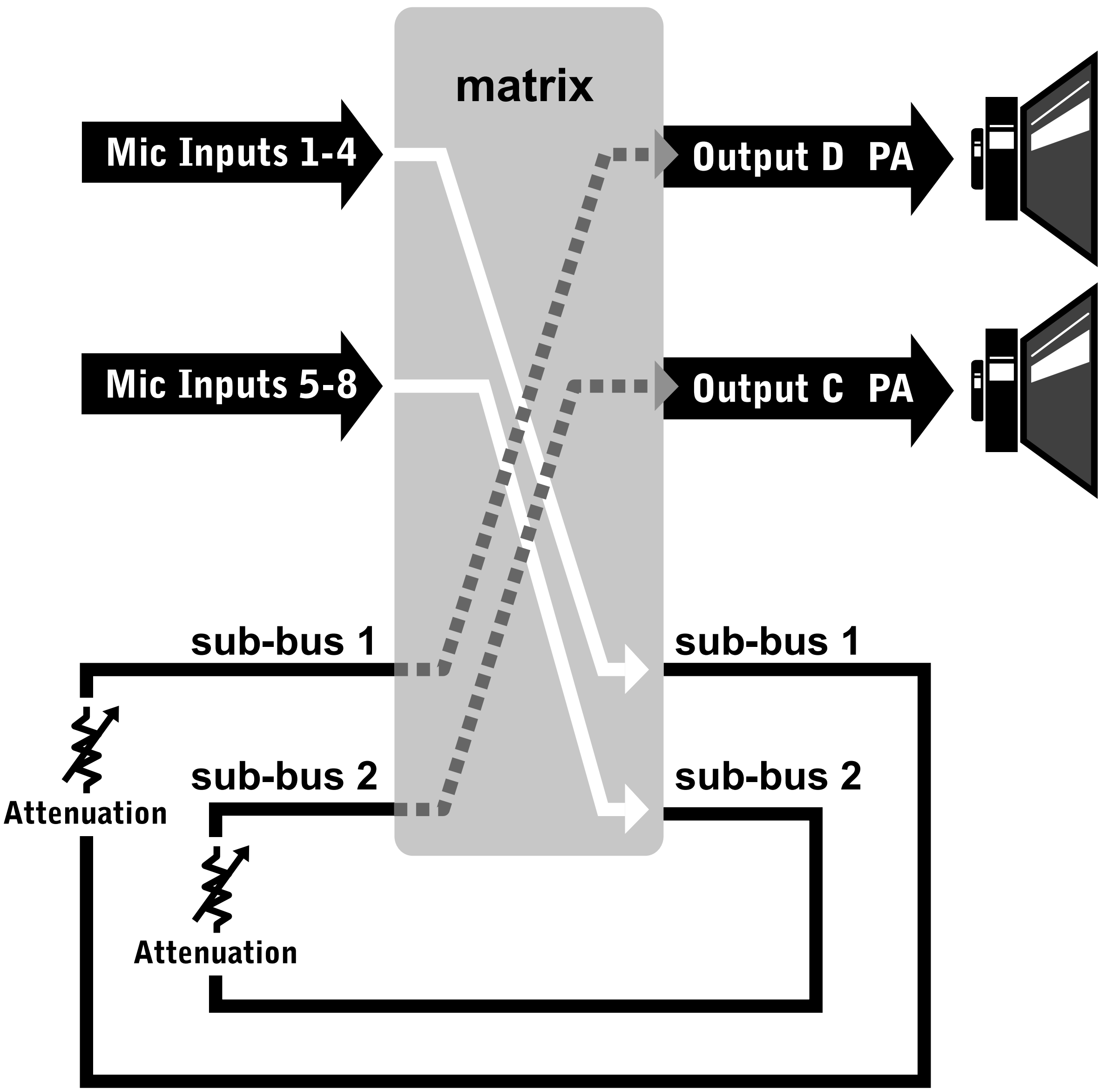


Diagram 21. An example of how the sub-bus mixing system works.

Routing Matrix Setup

		Outputs A: To CODEC Transmit	Outputs B: To AP10 Transmit	Outputs C: To Tape Record	Outputs D: To PA System (Amp/Speakers)	G-Link X: Mics 1-8 to X Bus	G-Link Y: Inputs A, B, C, D to Y Bus	G-Link Z	Sub-bus S1	Sub-bus S2	Outputs 1: Direct Input 1	Outputs 2: Direct Input 2	Outputs 3: Direct Input 3	Outputs 4: Direct Input 4	Outputs 5: Direct Input 5	Outputs 6: Direct Input 6	Outputs 7: Direct Input 7	Outputs 8: Direct Input 8
From CODEC Receive	Input A		X	X	X		X											
From AP10 Receive	Input B	X		X	X		X											
Tape Play	Input C	X	X		X		X											
	Input D	X	X	X	X		X											
Mics inputs from other AP800s	G-Link X Bus	X	X	X														
Inputs a,b,c,d, from other AP800s	G-Link Y Bus	X	X	X	X													
	G-Link Z Bus																	
	Sub-bus S1 In																	
	Sub-bus S2 In																	
Mic 1	Mic / Line 1*	G	G	G		G					N							
Mic 2	Mic / Line 2*	G	G	G		G						N						
Mic 3	Mic / Line 3*	G	G	G		G							N					
Mic 4	Mic / Line 4*	G	G	G		G								N				
Mic 5	Mic / Line 5*	G	G	G		G									N			
Mic 6	Mic / Line 6*	G	G	G		G										N		
Mic 7	Mic / Line 7*	G	G	G		G											N	
Mic 8	Mic / Line 8*	G	G	G		G												N

*Inputs 1-8 G = Gated N = Non gated

Echo Cancellation Setup

	Outputs A	Outputs B	Outputs C	Outputs D	Outputs 1	Outputs 2	Outputs 3	Outputs 4	Outputs 5	Outputs 6	Outputs 7	Outputs 8	G-Link EC Ref
Define EC Ref #1				X									
Define EC Ref #2													
Define G-Link EC Ref Bus													

Diagram 22. Default Routing

Programming Worksheet.

Diagram 23 is a programming worksheet that indicates all of the programming functions of the AP800 (except for control/status pin configurations). This diagram is provided to help you fully understand all AP800 functions and as a tool when designing a system. Both the default routing diagram and the programming worksheet are provided as loose worksheets for designing, programming and troubleshooting Audio Perfect™ systems.

When you begin programming the AP800 using the front panel, you will notice that the worksheet follows the same menu structure as the unit. Values in bold on the worksheet indicates the default value. Each unit has the capability of six presets. For this reason, you will need a programming worksheet and a default routing diagram for each preset required.

The top half of the worksheet indicates all system programmed parameters for the AP800. The upper left shows system-wide parameters while the upper right side sets the automatic mixing parameters. For more information on these parameters see "Automatic Microphone Mixing."

The bottom half of the worksheet indicates the input and output channel parameters. The "Inputs and Outputs" section of this document details these programming options.

System Wide Parameters

Program Parameter	Selection Range
Preset	**1**-6
Lock Front Panel	On, **Off**
Set Passcode	Any Front Panel Keys
Device ID No.	**0**-7
Unit ID No.	Factory Programmed
Mixer Mode	**Master-Single**, Master-Linked, Slave, Dual
RS232 Baud Rate	**9.6**, 19.2, 38.4 kbps
RS232 Flow Control	On, **Off**
Modem Mode	On, **Off**
Clear Modem Password	Enter
Timeout	0-15 (**10**)

Auto Mix Parameters

Program Parameter	Selection Range	
PA Adaptive	**On**, Off	
Maximum No. of Mics.	1-8 (**4**)	
First Mic Priority	**On**, Off	
Last Mic Mode	Last On, **Off** or Mic 1	
Gate Ratio	0-50dB (**15**)	
Off Attenuation	0-50dB (**12**)	
Hold Time	.1-8.0 seconds (.**3**)	
Decay Rate	**Slow**, Medium, Fast	
Manual Ambient	0dB to -70dB (**-30**)	

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Input Channel

Program Parameter	Selection Range	1	2	3	4	5	6	7	8	A	B	C	D	S1	S2
Input Gain Adjust	-20dB to 20dB (**0**)														
Subbus Attenuation	0dB to -20dB (**-6dB**)														
Input Type	**Mic 55dB**, Mic 25dB, Line														
Microphone Activation	**Auto**, Override On/Off, Manual On/Off														
Mute	On,**Off**														
Phantom Power	**On**, Off														
AGC	On, **Off**														
High-Pass Filter	On, **Off**														
EQ	On, **Off**														
Low Band EQ	-12dB to 12dB (**0**)														
Mid Band EQ	-12dB to 12dB (**0**)														
High Band EQ	-12dB to 12dB (**0**)														
Chairman Mic	On, **Off**														
Adaptive Ambient	**On**, Off														
EC Reference	**1**, 2														
Echo Cancellation	**On**, Off														
NLP	Off, **Soft**, Medium, Aggressive														

Output Channel

Program Parameter	Selection Range	1	2	3	4	5	6	7	8	A	B	C	D		
Output Gain Adjust	-20dB to 20dB (**0**)														
Mute	On, **Off**														
NOM	On (**A-D**) **or Off** (**1-8**)														

Diagram 23. Default Programming

8. Application Examples

Videoconferencing Application Example

- The eight microphones are input to mic/line channels 1-8. These audio sources are routed to the video CODEC input, the AP10 input and to the VCR input.
- Input A audio from the video CODEC has been routed to the AP10 hybrid, the VCR input and to the PA system.
- Input B, AP10 digital hybrid audio, has been routed to the video CODEC, to the VCR input and to the PA system.
- Input C, the VCR output is routed to the video CODEC, to the AP10 and to the PA system.
- EC Ref # 1 is defined as output D. This is because output D drives the PA system and it contains all the audio that needs to be cancelled at each mic input.
- Each mic input has selected EC Ref # 1 to be used as its reference audio for echo cancellation.

This application fits the default routing and default programming parameters. No changes in the factory defaults are required to install the system.

Videoconferencing Routing Diagram

Routing Matrix Setup

		Outputs A	Outputs B	Outputs C	Outputs D	G-Link X	G-Link Y	G-Link Z	Sub-bus S1	Sub-bus S2	Outputs 1	2	3	4	5	6	7	8
		To CODEC Transmit	To AP10 Transmit	To VCR Record	To PA System													
From CODEC Receive	Input A		X	X	X		X											
From AP10 Receive	Input B	X		X	X		X											
From VCR Play	Input C	X	X		X		X											
	Input D	X	X	X	X		X											
	G-Link X Bus	X	X	X														
	G-Link Y Bus	X	X	X	X													
	G-Link Z Bus																	
	Sub-bus S1 In																	
	Sub-bus S2 In																	
Mic 1	Mic / Line 1*	G	G	G		G					N							
Mic 2	Mic / Line 2*	G	G	G		G						N						
Mic 3	Mic / Line 3*	G	G	G		G							N					
Mic 4	Mic / Line 4*	G	G	G		G								N				
Mic 5	Mic / Line 5*	G	G	G		G									N			
Mic 6	Mic / Line 6*	G	G	G		G										N		
Podium, Mic 7	Mic / Line 7*	G	G	G		G											N	
Wireless, Mic 8	Mic / Line 8*	G	G	G		G												N

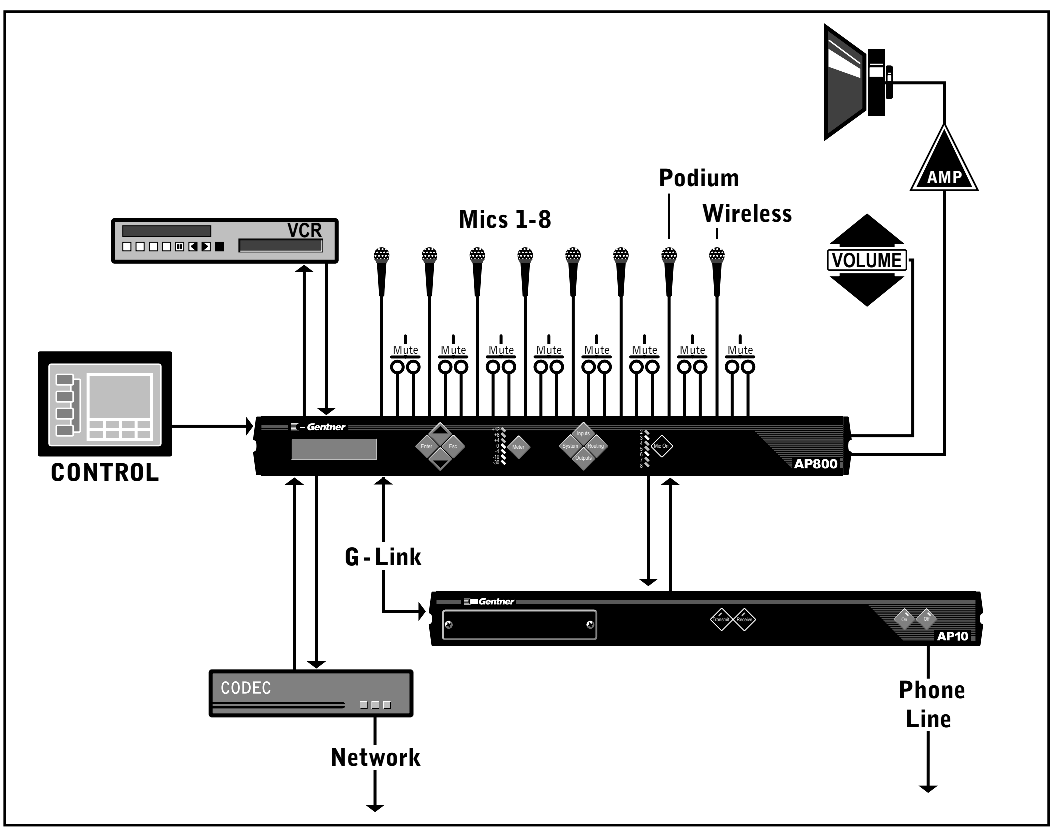
No Changes Required

*Inputs 1-8 G = Gated N = Non gated

Echo Cancellation Setup

	Outputs A	B	C	D	1	2	3	4	5	6	7	8	G-Link EC Ref
Define EC Ref #1				X									
Define EC Ref #2													
Define G-Link EC Ref Bus													

Diagram 24.



System Wide Parameters

Program Parameter	Selection Range
Preset	**1**-6
Lock Front Panel	On, **Off**
Set Passcode	Any Front Panel Keys
Device ID No.	**0**-7
Unit ID No.	Factory Programmed
Mixer Mode	**Master-Single**, Master-Linked, Slave, Dual
RS232 Baud Rate	**9.6**, 19.2, 38.4 kbps
RS232 Flow Control	On, **Off**
Modem Mode	On, **Off**
Clear Modem Password	Enter
Timeout	0-15 (**10**)

Auto Mix Parameters

Program Parameter	Selection Range	
PA Adaptive	**On**, Off	
Maximum No. of Mics.	1-8 (**4**)	
First Mic Priority	**On**, Off	
Last Mic Mode	Last On, **Off** or Mic 1	
Gate Ratio	0-50dB (**15**)	
Off Attenuation	0-50dB (**12**)	
Hold Time	.1-8.0 seconds (**.3**)	
Decay Rate	**Slow**, Medium, Fast	
Manual Ambient	0dB to -70dB (**-30**)	

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Input Channel

Program Parameter	Selection Range	1	2	3	4	5	6	7	8	A	B	C	D	S1	S2
Input Gain Adjust	-20dB to 20dB (**0**)														
Subbus Attenuation	0dB to -20dB (**-6dB**)														
Input Type	**Mic 55dB**, Mic 25dB, Line														
Microphone Activation	**Auto**, Override On/Off, Manual On/Off														
Mute	On,**Off**														
Phantom Power	**On**, Off														
AGC	On, **Off**														
High-Pass Filter	On, **Off**														
EQ	On, **Off**														
Low Band EQ	-12dB to 12dB (**0**)														
Mid Band EQ	-12dB to 12dB (**0**)														
High Band EQ	-12dB to 12dB (**0**)														
Chairman Mic	On, **Off**														
Adaptive Ambient	**On**, Off														
EC Reference	**1**, 2														
Echo Cancellation	**On**, Off														
NLP	Off, **Soft**, Medium, Aggressive														

No Changes Required

Output Channel

Program Parameter	Selection Range	1	2	3	4	5	6	7	8	A	B	C	D
Output Gain Adjust	-20dB to 20dB (**0**)												
Mute	On, **Off**												
NOM	On (**A-D**) or Off (**1-8**)												

Diagram 25. Videoconferencing Programming

Court Room Application Example

- The eight microphones are inputs to mic/line channels 1-8. These audio sources are routed and summed to the video CODEC input and the AP10 inputs. Each non-gated microphone is routed to a different input channel of the court recording system. Mics 1-4 are routed to sub-bus 1 and mics 5-8 are routed to sub-bus 2. This is used to level control each bus for sound reinforcement.
- Input A audio from the video CODEC has been routed to the AP10 hybrid, to the two zones of the PA system and to channel 8 of the court recording system.
- Input B audio from the AP10 hybrid has been routed to the video CODEC, to the two zones of the PA system and to channel 7 of the court recording system.
- Sub-bus 1 and 2 are actually now inputs to the matrix. Sub-bus 1 makes a cross-point connection to the output C bus. Thus, output C contains microphones 1-4, video CODEC audio and telephone audio from the AP10. Sub-bus 2 makes a cross-point connection to the output D bus. Thus, output D contains microphones 5-8, video CODEC audio and telephone audio from the AP10.
- EC Ref # 1 is defined as output D, while EC Ref # 2 is defined as output C. This is because outputs C and D drive the PA system and they contain all the audio that needs to be cancelled at the mic inputs.
- Mic input channels 1- 4 select EC Ref # 1, while mic input channels 5 - 8 select EC Ref # 2 as their reference audio for echo cancellation.

Routing Matrix Setup

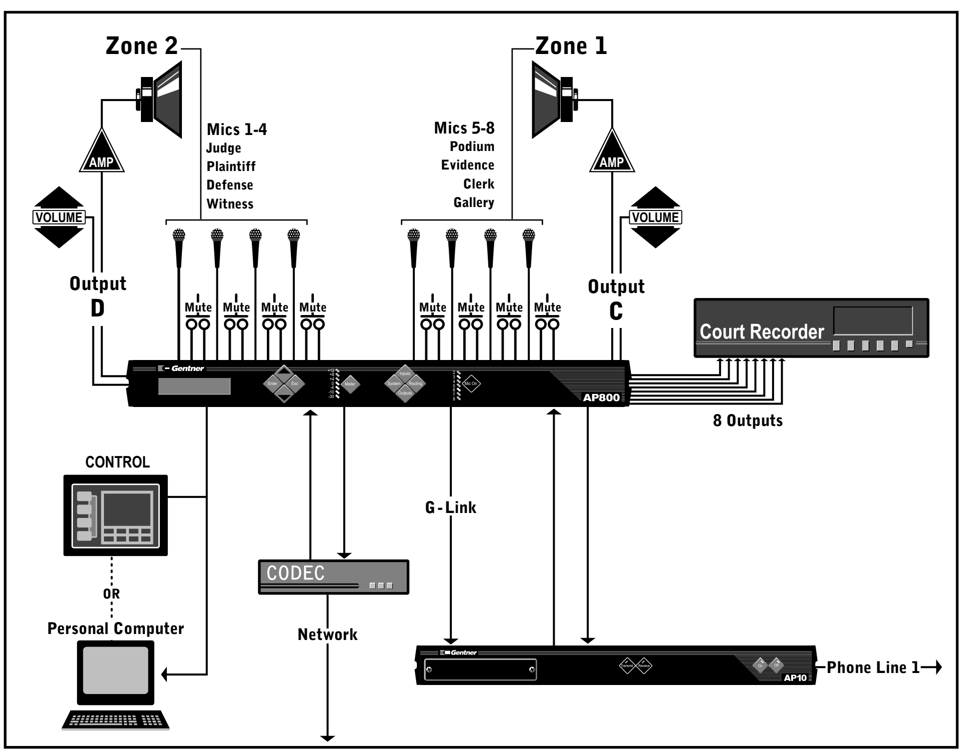
		Outputs A: To CODEC Transmit	Outputs B: To AP10 Transmit	Outputs C: To Tape Record	Outputs D: To PA System (Amp/Speakers)	G-Link X: Mics 1-8 to X Bus	G-Link Y: Inputs A, B, C, D to Y Bus	G-Link Z	Sub-bus S1	Sub-bus S2	Outputs 1: Direct Input 1	Outputs 2: Direct Input 2	Outputs 3: Direct Input 3	Outputs 4: Direct Input 4	Outputs 5: Direct Input 5	Outputs 6: Direct Input 6	Outputs 7: Direct Input 7	Outputs 8: Direct Input 8
From CODEC Receive	Input A		X	X	X		X											
From AP10 Receive	Input B	X		X	X		X											
Tape Play	Input C	X	X		X		X											
	Input D	X	X	X	X		X											
Mics inputs from other AP800s	G-Link X Bus	X	X	X														
Inputs a,b,c,d, from other AP800s	G-Link Y Bus	X	X	X	X													
	G-Link Z Bus																	
	Sub-bus S1 In																	
	Sub-bus S2 In																	
Mic 1	Mic / Line 1*	G	G	G		G					N							
Mic 2	Mic / Line 2*	G	G	G		G						N						
Mic 3	Mic / Line 3*	G	G	G		G							N					
Mic 4	Mic / Line 4*	G	G	G		G								N				
Mic 5	Mic / Line 5*	G	G	G		G									N			
Mic 6	Mic / Line 6*	G	G	G		G										N		
Mic 7	Mic / Line 7*	G	G	G		G											N	
Mic 8	Mic / Line 8*	G	G	G		G												N

*Inputs 1-8 G = Gated N = Non gated

Echo Cancellation Setup

	Outputs A	B	C	D	1	2	3	4	5	6	7	8	G-Link EC Ref
Define EC Ref #1				X									
Define EC Ref #2													
Define G-Link EC Ref Bus													

Diagram 26.



System Wide Parameters

Program Parameter	Selection Range
Preset	**1**-6
Lock Front Panel	On, **Off**
Set Passcode	Any Front Panel Keys
Device ID No.	**0**-7
Unit ID No.	Factory Programmed
Mixer Mode	**Master-Single**, Master-Linked, Slave, Dual
RS232 Baud Rate	**9.6**, 19.2, 38.4 kbps
RS232 Flow Control	On, **Off**
Modem Mode	On, **Off**
Clear Modem Password	Enter
Timeout	0-15 (**10**)

Auto Mix Parameters

Program Parameter	Selection Range	
PA Adaptive	**On**, Off	
Maximum No. of Mics.	1-8 (**4**)	
First Mic Priority	**On**, Off	
Last Mic Mode	Last On, **Off** or Mic 1	
Gate Ratio	0-50dB (**15**)	
Off Attenuation	0-50dB (**12**)	
Hold Time	.1-8.0 seconds (.**3**)	
Decay Rate	**Slow**, Medium, Fast	
Manual Ambient	0dB to -70dB (**-30**)	

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Input Channel

Program Parameter	Selection Range	1	2	3	4	5	6	7	8	A	B	C	D	S1	S2
Input Gain Adjust	-20dB to 20dB (**0**)														
Subbus Attenuation	0dB to -20dB (**-6dB**)													-6	-6
Input Type	**Mic 55dB**, Mic 25dB, Line														
Microphone Activation	**Auto**, Override On/Off, Manual On/Off														
Mute	On,**Off**														
Phantom Power	**On**, Off														
AGC	On, **Off**														
High-Pass Filter	On, **Off**														
EQ	On, **Off**														
Low Band EQ	-12dB to 12dB (**0**)														
Mid Band EQ	-12dB to 12dB (**0**)														
High Band EQ	-12dB to 12dB (**0**)														
Chairman Mic	On, **Off**														
Adaptive Ambient	**On**, Off														
EC Reference	**1**, 2					2	2	2	2						
Echo Cancellation	**On**, Off														
NLP	Off, **Soft**, Medium, Aggressive														

Output Channel

Program Parameter	Selection Range	1	2	3	4	5	6	7	8	A	B	C	D		
Output Gain Adjust	-20dB to 20dB (**0**)														
Mute	On, **Off**														
NOM	On (**A-D**) or Off (**1-8**)														

Diagram 27. Courtroom Programming

9. Direct Remote Control and Status

To allow many different control configurations, the AP800 can be controlled three ways: 1) using the front panel – information is displayed on the LCD display and programming keys are used to program; 2) serially via the RS232 port; 3) directly through the two control/status A or B labeled connectors on the rear panel or through the any combination of the above. This section discusses direct remote control.

There are two functions available: control and status. To activate a control function, pins on the control/status connectors (labeled A and B) must go low either in a momentary or sustained action depending on the setup of the unit. A status pin shows the status of a particular parameter. Status outputs sink 40mA of current through an open collector.

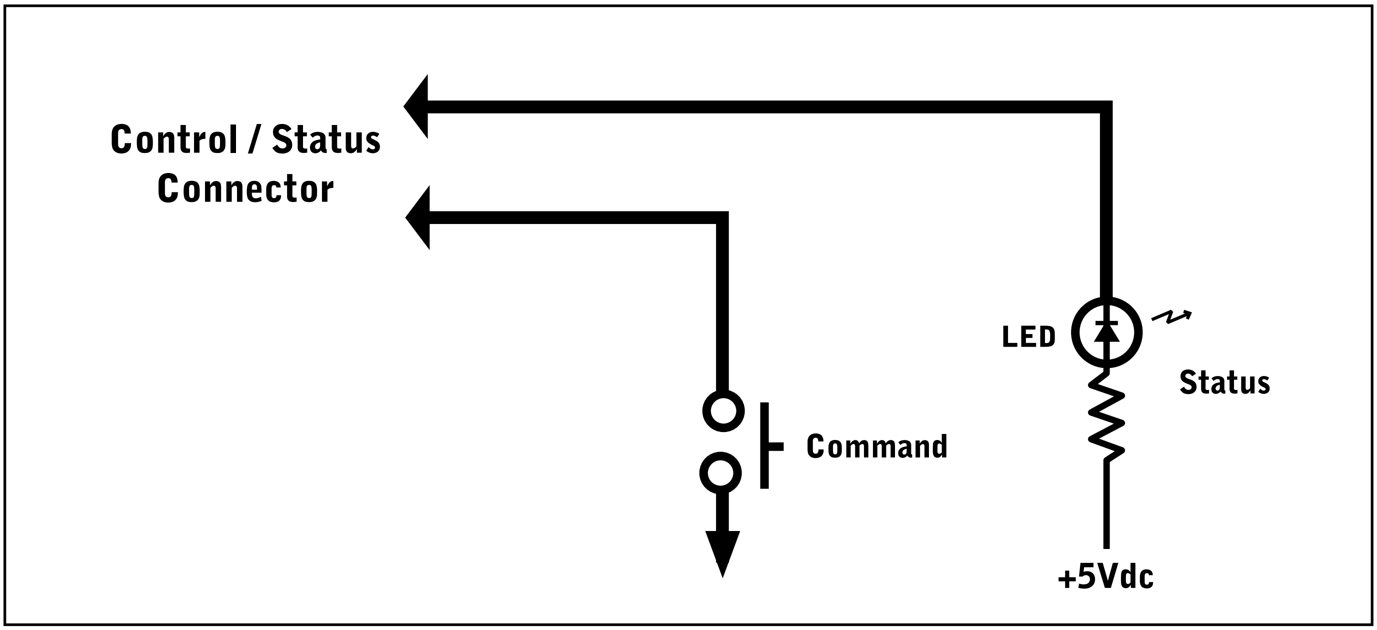

Diagram 28. Direct control of the AP800 is accomplished by sustaining or momentary closure to ground. Status provides open collector outputs capable of sinking up to 40mA of current.

Any valid function of all AP800s connected to the G-Link is capable of being controlled from the control/status pins of any connected AP800. These functions include but are not limited to volume up, volume down, mute, etc. In addition, pins can be programmed to call up to six preprogrammed presets (represent a programmed configuration). Thus, routing, level control, AGC, equalization, etc. can instantly be changed or "reconfigured" by presets. An example of a useful preset would be in room combining where the room configuration needs to change on the fly. Wall buttons can be used to activate presets. Refer to the Hotel/Convention Application Diagram (Diagram 5) in section 2.

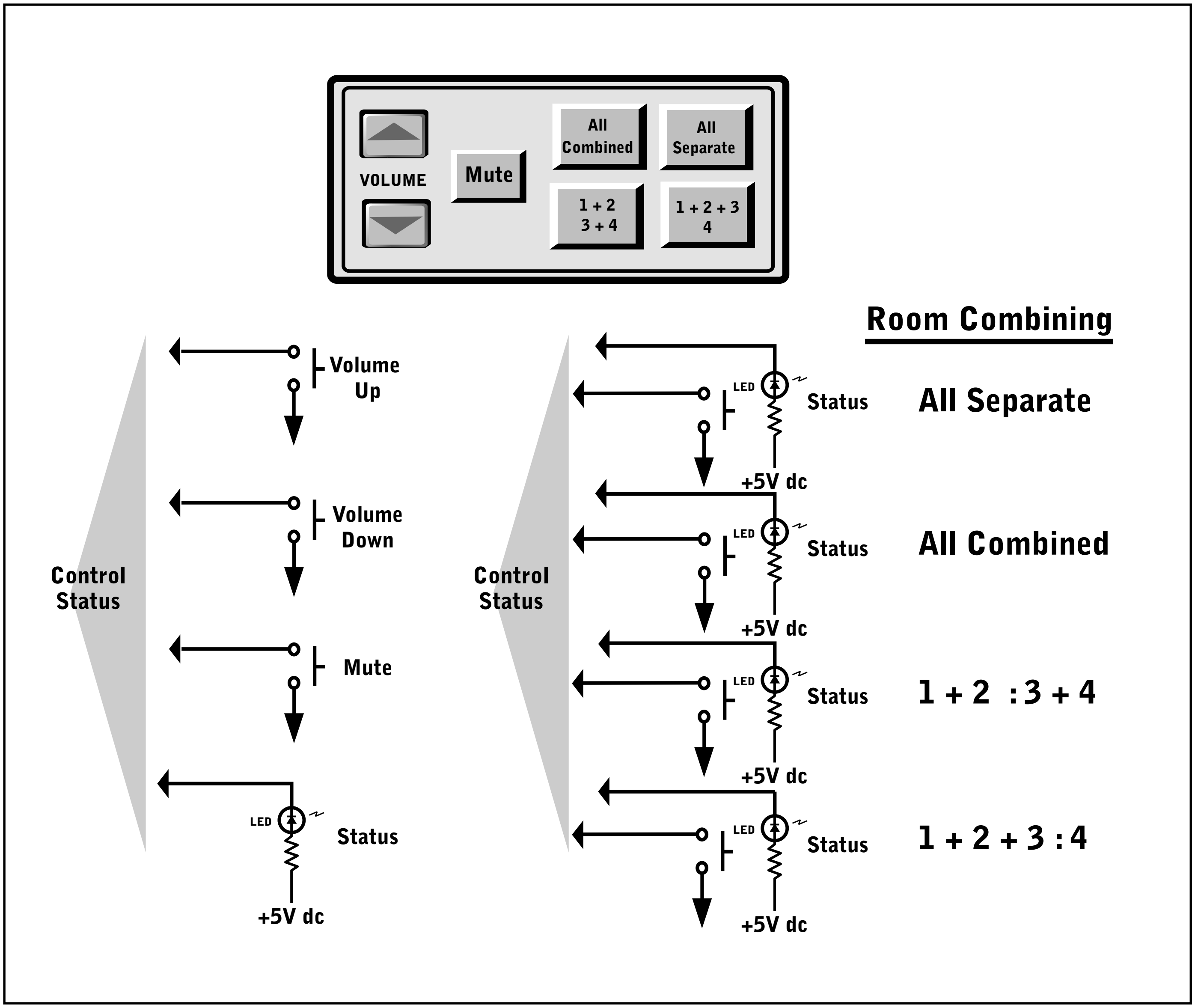

Diagram 29. A combination of command and status pins are used to create a direct remote control panel for the room combining application.

User Definable Control and Status Pins

Most of the pins on the control/status connectors can be programmed to perform any function – they are user definable. However, these pins come preprogrammed with defaults. There are some pins that are not user definable. The table on the following page outlines the pin configurations.

Control Status Connector A

PIN#	USER DEFINABLE YES OR NO	STATUS (S) OR COMMAND (C)	MOMENTARY (MA) OR LATCHING (L)	DEFAULT DESCRIPTION
1	Yes	C	M	Lock front panel toggle
2	Yes	S		Status of front panel lock
3	Yes	C	M	Mute all mics toggle
4	Yes	S		Status of mute all mics
5	Yes	C	M	Mute A output toggle
6	Yes	S		Status of A output mute
7	Yes	C	M	Mute B output toggle
8	Yes	S		Status of B output mute
9	Yes	C	M	Mute C output toggle
10	Yes	S		Status of C output mute
11	Yes	C	M	Mute D output toggle
12	Yes	S		Status of D output mute
13	Yes	C	M	Volume Up D Output (1dB)
14	Yes	S		Not used
15	Yes	C	M	Volume Down D Output (1dB)
16	Yes	S		Not used
17	No	S		Mic #1 Gate Status
18	No	S		Mic #2 Gate Status
19	No	S		Mic #3 Gate Status
20	No	S		Mic #4 Gate Status
21	No	S		Mic #5 Gate Status
22	No	S		Mic #6 Gate Status
23	No	S		Mic #7 Gate Status
24	No	S		Mic #8 Gate Status
25	No	Ground		Ground

Control Status Connector B

PIN#	USER DEFINABLE YES OR NO	STATUS (S) OR COMMAND (C)	MOMENTARY (MA) OR LATCHING (L)	DEFAULT DESCRIPTION
1	Yes	C	M	Preset 1
2	Yes	S		Status of Preset 1
3	Yes	C	M	Preset 2
4	Yes	S		Status of Preset 2
5	Yes	C	M	Preset 3
6	Yes	S		Status of Preset 3
7	Yes	C	M	Preset 4
8	Yes	S		Status of Preset 4
9	Yes	C	M	Mute Input A Toggle
10	Yes	S		Status of Input A Mute
11	Yes	C	M	Mute Input B Toggle
12	Yes	S		Status of Input B Mute
13	Yes	C	M	Mute Input C Toggle
14	Yes	S		Status of Input C Mute
15	Yes	C	M	Mute Input D Toggle
16	Yes	S		Status of Input D Mute
17	No	S		Input A Presence Status
18	No	S		Input B Presence Status
19	No	S		Input c Presence Status
20	No	S		Input D Presence Status
21	No connection			
22	No connection			
23	No	-		+5Vdc
24	No	-		+5Vdc
25	No	-		Ground

Many times, a combination of serial and direct control might be used in an installation. For example, in the room combining example outlined on Page 8, it may be useful to use a custom remote controller touch screen for controlling the bulk of the A/V and other equipment in the room. However, functions such as volume, mute and room combining configuration may be controlled directly from the unit.

Example of Programming an AP800 and Using Direct Control

To further clarify how one might use the control and status pins on the AP800, here is a step-by-step example:

1. Design the entire audio chain and document.
2. Determine which pins on the control/status connector will be used for control and status. (See table above.)
3. If using a custom remote controller, program these units to communicate via RS232 with the AP800. Connect to the AP800 and verify proper operation.
4. Using the Audio Perfect™ Tools Windows® software, configure the desired number of presets for the application. Items such as routing, AGC, equalization, levels, etc. are programmed and saved as a preset. Presets can be called by the custom remote controller, via the front panel, from the control/status connector or any combination of the above.
5. Define the pin number on the control/status connector as the presets using the software. In addition, program any status outputs.
6. Connect the user panel(s) and verify direct control/status is operational.
7. All of the above operations can be performed and tested on the work bench.
8. Now install the audio system and complete the audio calibration procedure by either using the front-panel LCD display or the PC software or combination thereof. All gains and actual levels in dBu are provided at both locations as well as test tones and test noise.

10. Serial Remote Control

An Audio Perfect™ system is most easily set up and operated via the RS232 serial port. When multiple AP800 and AP10 units are G-Linked together, operation of the entire system can be accomplished with a single serial connection. Audio level control, muting, audio signal routing, telephone dialing, remote diagnostics and a host of functions can be addressed using the RS232 serial port.

While any external device with an RS232 serial connection can communicate with the Audio Perfect™, the system was designed primarily to be programmed and set up using a personal computer, and operated using a custom remote controller. While the front-panel controls and LCD panel provide a majority of supported functionality, there are a number of higher level functions that can only be accessed via RS232.

The AP800 provides real-time control and status via the RS232 serial port of all functions of the system, 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
- Echo canceller status and performance including echo return loss (ERL) and other performance measures.
- Routing
- Automixing functions and modes
- Control/Status connector configurations
- Preset configurations
- Password protection
- G-Link setup
- System setup

The RS232 serial port was generally intended to be used as follows

1. Setup and programming. At the design and installation level, a PC is connected to the AP system and the system is programmed for the application gain structure, routing, equalization, etc.
2. Calibration. Using either a connected PC or the front-panel LCD display and/or meter, precise input and output levels can be directly read from a PC or front panel LCD display in dBu. Additionally, the AP system will tell you precisely how much gain is programmed into the system. Example: you place a -6dBu input tone into a mic/line channel in the line mode. The input reading will read -6dBu. If you apply say 10dB of gain, the input will read +4dBu at 10 as a gain reading.
3. Operation. A custom remote controller is then used to operate the functions of the Audio Perfect™ system.

11. Front Panel Programming

Using a PC to program an AP system is a fast and simple way to set up, calibrate and test the AP800 system. However, what happens when you don't have a PC and you need to make a change now? The AP800 front-panel LCD and programming keys are designed just for this reason. In fact, most of the functions of the AP800 can be programmed from the front panel. And with the default routing (See section 8, routing.), there may be several installations where only minimal changes will need to be made.

Refer to the AP800 front panel diagram: the front panel consists of a LCD display, nine programming buttons, a LED VU meter and eight mic LEDs.

To protect from unauthorized changes, the AP800 is pass coded. Navigation of the menus is allowed without a pass code; however, changes to programming require a valid pass code.

There are nine programming keys on the AP800. The first four keys (▲,▼, ENTER and ESC) are used for navigation in menus selected by the other five keys. The METER, INPUTS, OUTPUTS, ROUTING and SYSTEM keys select the appropriate menu as follows:

METER - This menu determines what audio is selected on the front panel LED VU meter. In addition, numeric audio levels are displayed on the LCD panel. Echo return loss (ERL) and echo return loss enhancement (ERLE) is also metered.

INPUT - This menu is used to set up and program the input channels for microphone inputs 1-8, line inputs A-D and sub-bus 1-2. Refer to the system block diagram (Diagram 6 in section 3). On this menu, you can select

- Input Audio Level - When this screen is displayed, both the actual level in dBu is displayed on screen and the gain of the system in dB. You can change the gain of the input using the ▲/▼ keys.
- Mic 55/Mic 25/Line Input
- Mic Activation, Auto Gate/Manual Gate On,Off/ Gate Override On,Off
- Mute On/Off
- Phantom Power On/Off
- AGC On/Off
- High Pass Filter On/Off
- EQ On/Off
- Low Band Equalizer -12dB to +12dB
- Mid Band Equalizer -12dB to +12dB
- High Band Equalizer -12dB to +12dB
- Chairman Override On/Off
- Adaptive Ambient On/Off
- AEC Reference Ref 1/ Ref 2
- Acoustic Echo Cancellation On/Off
- Non-Linear Processing Soft/Medium/Aggressive/Off

OUTPUT - This menu is used to set up and program the line output channels 1-8 and line outputs A-D. Refer to the system block diagram in section 3. On this menu, you can select

- Output Audio Level - When this screen is displayed, both the actual level in dBu is displayed on screen and the gain of the system in dB. You can change the gain of the output using the ▲/▼ keys.
- Mute On/Off
- NOM Mode On/Off

ROUTING - This menu determines what inputs to the matrix go to which outputs of the matrix. Since the AP800 has 25 possible inputs and 17 possible outputs, this process can be complex. Section 8 describes the possibilities. This menu takes you through the process of setting up the routing matrix for your application and loading preset values into the six available presets.

SYSTEM - This is where everything else goes. You can program the following:

- Preset selection
- Lock front panel On/Off
- Set pass code
- Device ID number (0-7)
- Unit identification
- Master-single/linked, slave, dual
- RS-232 baud rate and flow control
- Modem mode On/Off
- Front panel time-out (0 to 15 min.)
- PA adaptive mode On/Off
- Maximum number of microphones (1-8, Off)
- 1st Mic Priority On/Off
- Last mic mode: Last, Mic 1, Off
- Gate Ratio (0 to 50dB)
- Off Attenuation (0 to 50dB)
- Hold Time (0.1 to 8.0 second)
- Decay Rate (second, medium, fast)
- Manual Ambient Level (0 to -70dB)

The eight mic on LEDs indicate the gate status of a microphone channel. If the LED is on, it indicates that the microphone is gated on.

The following items cannot be programmed from the front panel and require connection to PC:

- Loading of new software to the AP800
- Programming of pins on the control/status connectors

12. Programming Using a Personal Computer

One of the most useful features of the AP800 is its ability to be programmed via a PC. All AP800 functions can be programmed via a PC and new software releases can be uploaded from a PC to the AP800.

The software is a Windows® 95/NT application. Configurations can be programmed and saved on disk. When connected to the unit, configurations can be loaded into the AP800. Additionally, configurations from the AP800 can be loaded into the PC to see how the AP800 has been programmed.

Real-time analysis of audio levels in dBu, gate status, ERL (echo return loss) and other measurements are available.

The AP800 can be connected via a modem for complete remote diagnostics, programming and control.

For more information, obtain a copy of AP Tools software from your dealer or representative.

13. Expanding the System - Using the G-Link

One of the most important features of the Audio Perfect™ system is its expandability. Analog audio products such as automatic microphone mixers offer limited expandability. Using analog methodologies, such mixers provide only a few functions that operate across all expanded units. The most common mode that is expanded is NOM (number of open microphones). Unlike these devices, all functions are available to expanded units making Audio Perfect™ capable of automixing 64 microphones and 32 line inputs.

Audio Perfect™ takes advantage of its DSP infrastructure in accomplishing this task. The G-Link is a high-speed network protocol that allows up to eight AP800 units and 16 AP10 units to be networked together. The G-Link provides two primary system functions: 1) communications among units and 2) audio linking.

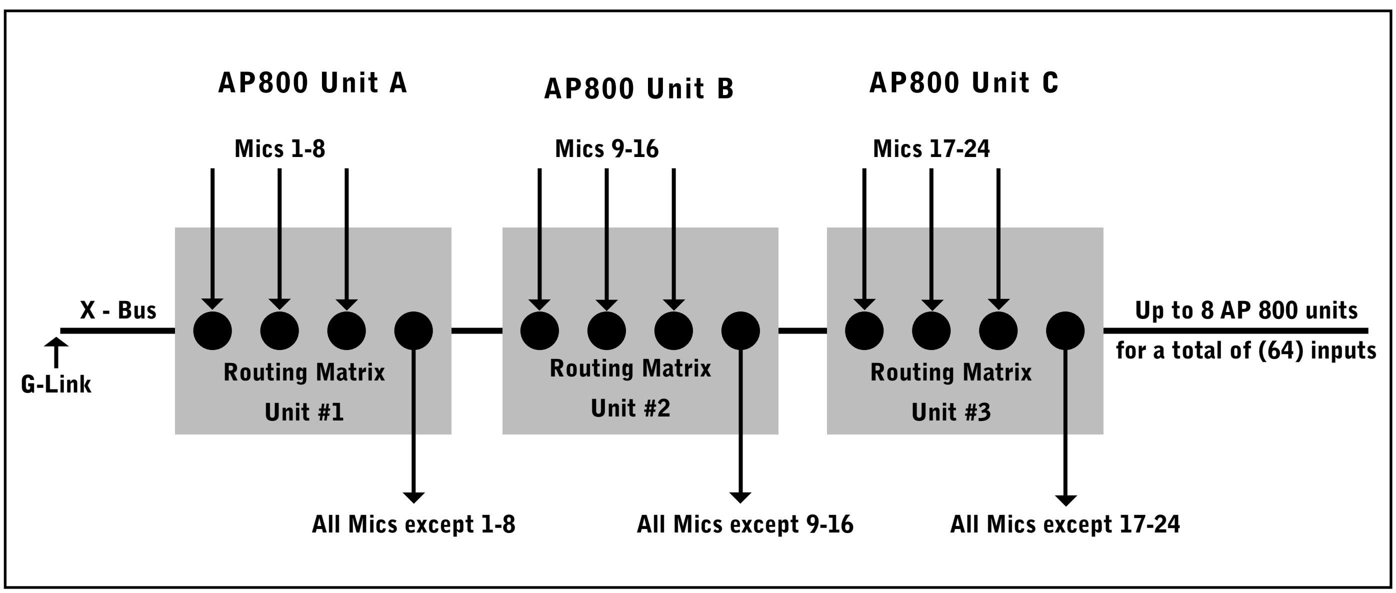

Diagram 30. The network audio buses (X, Y, Z, EC Reference) provide a method of mix-minus bus for distributing audio to other AP units.

Communication Functions of the G-Link

Networked Audio Perfect™ units communicate to one another via the G-Link. Control, status and addressing functions are performed via the network bus. To accomplish this, the installer determines the MASTER unit and programs this unit as such. All additional units are then programmed as SLAVEs. The master unit then provides communication supervision for all other units on the network.

Serial connection to the master AP800 permits programming, operation and diagnostics to all AP units networked together. This permits a single connection for the installer and user, decreasing costs and complexity.

G-Link Audio Functions

The Audio Perfect™ line allows for four digital audio mixing buses on the G-Link. These buses are referred to as the X, Y, Z and G-Link EC Reference buses. Audio on any networked unit can be placed on a bus or audio can be taken off a bus and routed to any destination within the unit. Three buses are mix-minus buses. That is to say that any audio that is placed on the bus for a particular unit, is not fed back to that unit when audio is taken off that bus.

Diagram 30 shows an example of how these virtual buses operate. In this example, microphones 1 through 8 are routed to the X-Bus on units A, B and C. As you can see, on the outputs of each unit, only the inputs from the other units are available on the bus.

The X, Y and Z-Bus can have any audio source on the unit routed to them. However, they are programmed for the following defaults:

X-Bus Default - Master Microphone Mix. All microphone input channels are routed to the X-Bus. This is how microphone audio is delivered to video CODECs, AP10 digital hybrid inputs, sound reinforcement devices and other audio devices.

Y-Bus Default - Auxiliary Mix. All line input audio (inputs A, B, C and D) is mixed to this bus. This is how audio from video CODECs, AP10 digital hybrid outputs and other audio devices is delivered to other destinations on the AP system.

Z-Bus Default - This is a user defined auxiliary mix-minus bus.

G-Link EC Reference Bus -This bus works identically to the X, Y and Z-buses. This bus provides a system wide bus for mic channels to receive a reference input for their echo canceller. Here's an example to clarify: Let's say you have four AP800 units G-Linked together. Audio on output D of unit 1 is audio routed to the PA system in the room. This audio is needed as a reference to the echo cancellers for mics on units 2, 3 and 4. This is accomplished by routing output D on unit 1 to the G-Link EC Reference Bus and then routing the output of the G-Link EC Reference Bus to the input of EC Reference 1 on units 2, 3, and 4.

Connecting to the G-Link

The AP products come standard with one G-Link cable. The G-Link can be extended to a total distance not to exceed 20 feet.

14. System Connections

Audio Connections

Audio Perfect™ products utilize 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 were chosen to maximize reliability and ease of us.

Control/Status Connections

Direct remote control and status outputs are provide on two DB25 connectors on the rear of the AP800.

G-Link Connection

The G-Link consists of two RJ45 connectors. A cable of 18″ in length is provided with every Audio Perfect™ product. Additional G-Link cables are available.

Serial RS232

The serial RS232 communications port is connected via a standard DB9 connector.

RS232 baud rate can be programmed for 9600, 19.2K and 38.4K baud rate and flow control can be set for either hardware or none.

Pass Codes

To prevent unwanted access to the front panel or via the modem, the unit can be programmed to require an access code. The RS232 pass word is set from a PC. Should the RS232 pass word be forgotten, it can be reset from the front panel.

Meters

The AP800 has both an LED meter and an LCD display. Whenever the input or output menus are accessed, the meter displays the level for the input or output access. When not in the input or output menus, the default meter is shown. The default meter can be changed to any input, output or ERL (echo return loss) by pressing the Meter button.

Power

A universal power connector is provided. All Audio Perfect™ products will operate on all global voltage and cycles.

Telephone

The AP10 connects to the telephone network and to a telephone set via a standard RJ11 connector. The product has been approved for connection to telephone networks in the United States, Canada, Europe (CE compliant), the United Kingdom (BABT compliant), and compliance is pending for other countries.

15. Software Upgrades

To accommodate future releases of software that provide additional capabilities, improved performance and custom features, the AP800 is capable of having its software upgraded. This is accomplished by connection to a PC either directly or through a modem. Software upgrades will be made available on disk or via the Worldwide Web.

16. Product Specifications

AP800 Specifications

Dimensions	19" W x 1.75" H x 10" D 48.3 W x 4.45 H x 25.4 D cm
Weight	10 lb/4.5 kg dry weight 13 lb/5.9 kg shipping weight
Power	Auto-adjusting power module 100-240 Vac; 50/60Hz; (fuse) 2 amp 250 Vac, 30W, typical
RS232	DB9 female, selectable 9600/19,200/38,400 baud
Control/Status I/O	DB25 female; (2) + 5 Vdc pins, 100mA each
Control Inputs	Input activation selectable; momentary or latching ground
Status Outputs	DB25 female; open collector, 20V Max, 40MA each
Acoustic Echo Cancellation Tail Time	> 120 milliseconds, each gated input
Operating Temperature	32-100 degrees F / 0-38 degrees C
Government Approvals	FCC Part 15, CSA, NRTL/C, CE
Mic/Line Inputs	Removable Terminal Block Connector, -55 or 0dBu nominal adjustable, balanced, bridging Impedance 4k ohms
Frequency Response	20 to 15kHz ± 2dB
Phantom Power	24Vdc input selectable
Noise	EIN 20Hz to 15kHz - 125dBu
THD	< 0.1%
Line Inputs	Removable Terminal Block Connector, 0 dBu nominal adjustable, balanced, bridging
Impedance	20kOhms
Frequency Response	20 to 15kHz ± 2dB
SNR	>65dB ref. 0dBu input, 0dBm output
THD	< 0.1%
Outputs	Removable Terminal Block Connector, 0 dBm nominal, level adjustable, balanced
Impedance	50 ohms (designed to drive > 600 ohms inputs)
Frequency Response	20 to 15kHz ± 2dB
SNR	>65dB ref. 0dBu input, 0dBm output
THD	< 0.1%

AP10 Specifications

Dimensions	19" W x 1.75" H x 10" D 48.3 W x 4.45 H x 25.4 D cm
Weight	10 lb/4.5 kg dry weight 13 lb/5.9 kg shipping weight
Power	Auto-adjusting power module 100-240 Vac; 50/60Hz; (fuse) 2 amp 250 Vac, 8W, typical
Remote	DB25 female connector, Momentary/Latching
Transmit Input	Removable Terminal Block Connector, 0dBu, balanced, bridging Impedance >10k ohms
Receive Outputs	Removable Terminal Block Connector, 0dBu, balanced, 50 ohms
Audio Performance Frequency Response	250Hz to 3.5kHz ± 1dB
SNR	>56dB ref. 0dBu input at -15dBm on the telephone line
THD	< .3% 250Hz to 3.5kHz (AGC disabled)
Operating Temp.	32 to 100 degrees F / 0 to 38 degrees C
Government Approvals	FCC Part 15 and Part 68, CSA, NRT L/C, CE, BABT

17. Glossary of Terms

Acoustic Echo - Audio from a speaker that is picked up by a microphone is defined as acoustic echo. When this audio is returned to a distant site in a teleconferencing application, it is difficult for the participants at the other end to effectively communicate since they are hearing their own audio (echo) coming back at them.

Acoustical Treatment - Refers to treating rooms acoustically to minimize acoustical reverberation and noise.

Adaptive Ambient Mode - An automixing function that dynamically changes the ambient noise reference level based on a moving average of the actual noise floor of the room as picked up by the microphones.

AGC - Automatic Gain Control. Increases the audio level when it is too low and decreases the audio level when it is too high.

Ambient Level - The sound pressure level (SPL) of a room when no one is speaking and no external audio is present. Also known as noise floor.

Automatic Microphone Mixer - A microphone mixer that automatically activates and deactivates microphone channels. This serves to increase intelligibility of the audio by reducing reverberation and noise picked up by the microphones.

Decay Rate - The rate in which an automatic mixer attenuates a microphone channel audio level.

Digital Hybrid - A telephone hybrid that uses DSPs to provide excellent audio separation between the send and receive audio paths.

Distributed Echo Cancellation™ - A registered trademark of Gentner Communications Corporation. Refers to providing acoustic echo cancellation on every microphone input. This technology affords more reliable acoustic echo cancellation.

DSP - Digital Signal Processor. A high speed digital numeric processor that can rapidly calculate complex mathematical equations. The DSP is the most important component used in Gentner digital products.

Echo Cancellation - The process in which acoustic echoes are canceled. This is accomplished by sampling the distant audio that is amplified through the speaker in the room. When this audio is picked up by the microphone (this is acoustic echo), an echo canceller eliminates or "cancels" the echo.

Equalization - Filtering of audio. This can include high, low and band pass filtering. Used to filter out unwanted hum, noise and other unwanted audio. Is also used to tailor the sound for the best intelligibility and ease of listening.

Four Wire - Typically refers to audio in and out of a video CODEC.

G-Link - A high speed digital network bus used to link AP800 and AP10 units.

Hold Time - The amount of time an automixer waits after a microphone channel has no stimulus before beginning the off attenuation process.

Integrated Dialing - The ability of a telephone hybrid to dial (generate touch tones) via a RS232 serial connection.

Matrix Mixer - A device that takes multiple audio inputs mixes them to any combination of outputs.

NOM - Number of open microphones. An automixing function that establishes a constant microphone gain in the room. As more microphones are activated, each microphone gain is reduced and vice versa.

Omni-directional Microphone - The pick up pattern of a microphone that picks up audio in a 360-degree pattern.

PA Adaptive Mode - An automixing function exclusive to Gentner products that keeps microphone channels from activating when distant or other audio sources are amplified through a speaker in a room.

Reverberation - Audio in a room that bounces off a wall, ceiling, table or other objects and then is picked up by microphone. High reverberation causes listener fatigue and is difficult to understand.

Simultaneous Two Wire/Four Wire - The ability for a teleconferencing device to connect to a video CODEC and a telephone line simultaneously.

SPL - Sound Pressure Level. The measure of acoustical audio in the air. Is measured in dB using a SPL meter.

Telephone Hybrid - A term used to refer to a telephone interface that separates send and receive audio. Same as a digital hybrid

Two Wire - Typically refers to a telephone line.

Uni-directional Microphone - The pick up pattern of a microphone that picks up audio in a directional (usually 180 degrees or less) pattern.

Notes

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