PSR1212

Perfect Sound Results

WHITE PAPER

Gentner®

Table of Contents

PSR1212 White Paper
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1 Introduction

Overview

Audio is critical to human communication. Media such as voice mail, the Internet, conference calling, videoconferencing, and electronic presentations are driving the demand for better audio-communication technologies. The sound reinforcement arena also demands higher-quality sound to more faithfully reproduce source audio. At the same time, organizations using audio-communication technologies are looking for ways to decrease costs and complexity while increasing efficiency.

The PSR1212 Digital Matrix Mixer with Audio Processing meets the demands of a wide variety of sound reinforcement and conferencing requirements with 12 x 12 digital matrix mixing, parametric equalizers, filters, and 32 customizable presets. These features enable the PSR1212 to create a quality audio experience in many venues—from auditoriums and arenas to training rooms and boardrooms.

A quality audio experience is one where the audio source material is the message, not the audible inadequacies of a poorly designed or configured audio system. With a properly configured PSR1212, participants and observers do not become fatigued by reverberated audio, which reduces sound clarity.

For ease of use, the PSR1212 facilitates local and remote PC setup and diagnostics, logic outputs, and gated microphone operation. Microphone inputs can be individually customized to gate on and off as you wish, while automatic gain control keeps the overall sound level consistent. Input channels 1-8 can be configured as an automatic microphone mixer.

All microphone mixing parameters can be customized. 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 an RS-232 serial interface.

The key benefits the PSR1212 provides are:

- Outstanding audio clarity and intelligibility
- Seamless integration to external control devices
- Reduced number of separate audio devices required
- Adaptability to a variety of sound applications
- Expandability
- Ease of design and installation

For more information about installing and configuring the PSR1212, refer to the PSR1212 Installation and Operation Manual.

This document outlines the features, functions, applications and technical details of the PSR1212. Some of the diagrams included here depict other Gentner products, such as the PA870 Power Amplifier. You can find out the latest information on all Gentner products by calling us 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.

2 Product Description

Introduction

The PSR1212 is a highly-advanced twelve-by-twelve digital matrix mixer with 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 auditoriums, arenas, theaters, gymnasiums, hotel/convention centers, conference rooms, training rooms, boardrooms, and paging systems.

Features and benefits:

- 12 x 12 matrix mixer with level control at the cross points.
- Twelve line output channels. All output levels are adjustable and can be muted.
- Eight audio processing buses, each with 15 filters, can be placed anywhere within the matrix mixer audio path.
- Eight-channel automatic microphone mixer with four line inputs. The mixer operates across linked units.
- Input gain, configurable audio processing, muting, and automatic mixer are programmable per input channel (inputs 1-8 only).
- Configurable audio processor with four filters on inputs 1-8.
- All interconnected devices can be accessed, controlled, and programmed via a single RS-232 connection.
- 32 programmable presets for instant configuration changes.
- Allows grouping of mics across four automatic mic mixers within a single PSR1212.
- Internal room combining capabilities.
- Logic outputs.
- Configurable microphone gating.
- Input channels 1-8 are configurable as an automatic mic mixer.
- Any combination of inputs can be routed to any combination of outputs.
- Network-based interconnectivity allows up to eight PSR1212s to be connected and controlled as a single unit, allowing 96 inputs and 96 outputs.
- Remote and local PC set up and diagnostics.
- Macro Pro software allows the unit to function without an external control system.
- 100 percent digital signal processing.
- Gentner service and support.
- Worldwide certifications for safety and emissions: CE, FCC, CSA, C-TICK registered.
- One-year limited warranty.

Note: The PSR1212 is not an Audio Perfect product.

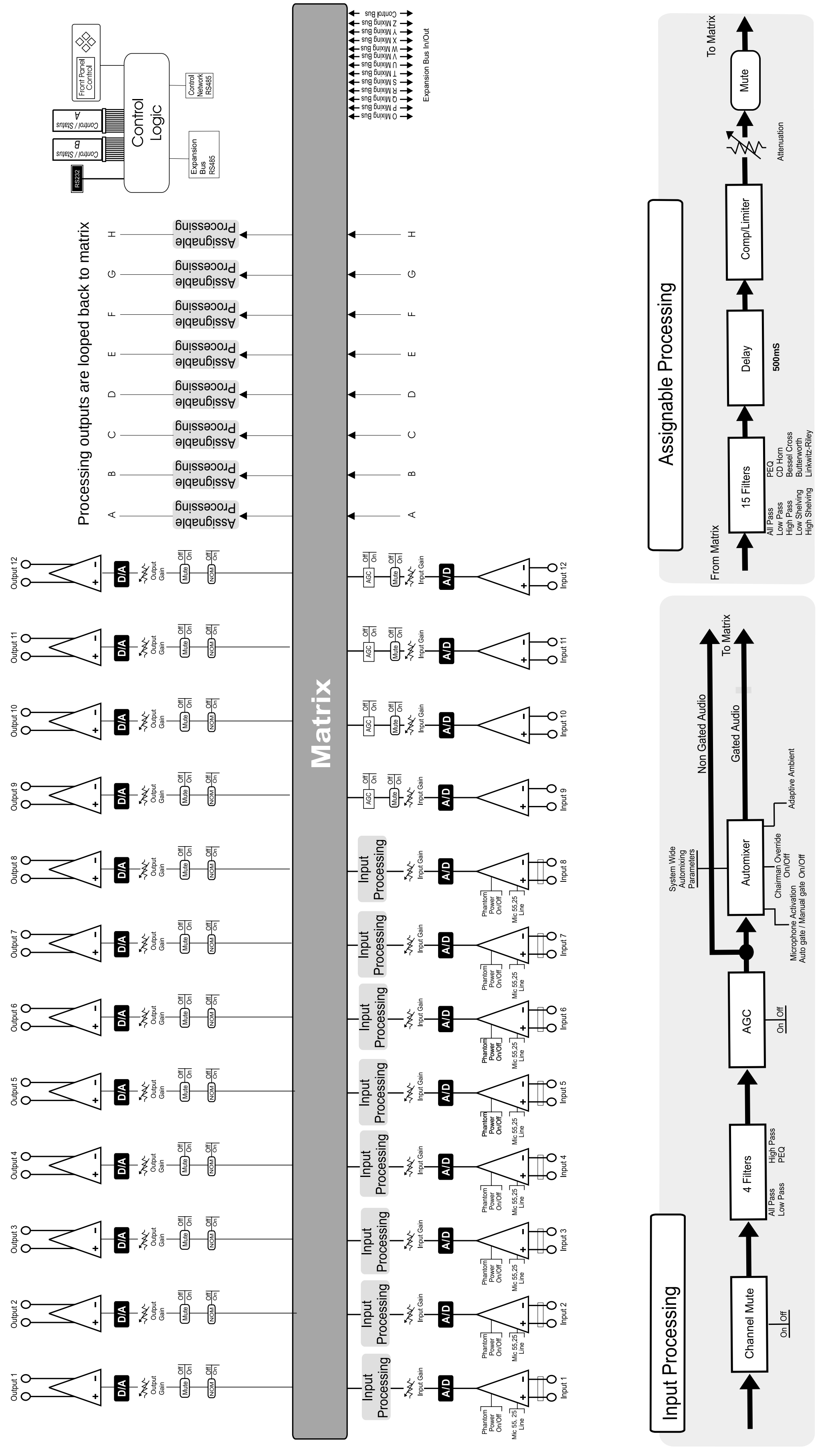

Figure 1. PSR1212 Block Diagram

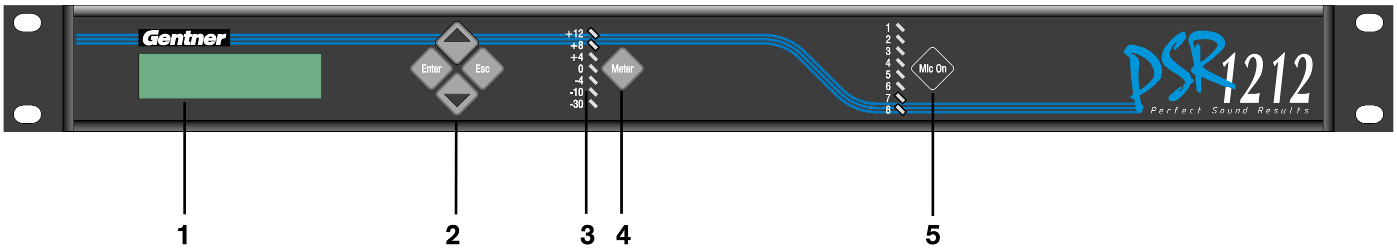

Figure 2. PSR1212 Front Panel

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

2. Enter/▲▼/ESC - Used to navigate the PSR1212's menu system.

3. LED Meter - This LED bar meter is used to display the audio level of an input, output, or processing channel of the PSR1212.

4. Meter - Takes you directly to the Meter branch of the PSR1212's LCD programming tree.

5. Mic On LED - These LEDs indicate microphone gate status.

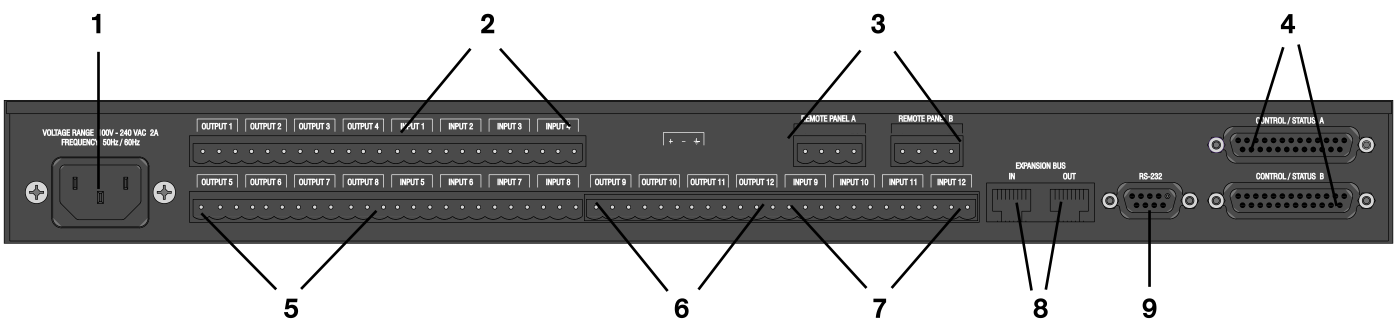

Figure 3. PSR1212 Rear Panel Connections

Rear Panel Connections

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

2. Inputs 1-8 - For mic- and/or line-level inputs.

3. RS-485 Remote Control Ports - These four-pin Phoenix™ ports allow you to control the PSR1212 with a Gentner remote controller.

4. Control/Status Ports A and B - These DB25 connectors are for connecting control devices. The control devices have access to the command set for the PSR1212 and can be used for functions such as volume, muting, preset change, room combining, etc. Devices can be connected to either port.

5. Outputs 1-8 - 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.

6. Inputs 9-12 - For line-level inputs.

7. Outputs 9-12 - Same functions as Outputs 1-8.

8. Expansion Bus In/Out - Used for daisy-chaining PSR1212 units in a network.

9. RS-232 - This DB9 serial port is for interconnection between the PSR1212 and a PC, modem, or other custom remote controller.

Expansion Bus Connections

The Expansion Bus network architecture allows up to eight PSR1212s and up to 96 inputs, 96 outputs, and 64 microphones 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 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 PSR1212s. 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 or VCR, 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 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 PSR1212s. 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.

Network Requirements

The Expansion Bus (RS-485 LAN) lets you link multiple PSR1212s. The maximum distance allowed between any two PSR1212 units on an Expansion Bus network is 80 feet. Gentner recommends category five twisted-pair (10BaseT LAN) cable be used.

Equipment Placement

The PSR1212 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

The PSR1212 can safely operate in temperature environments between 32°—110°F.

PSR1212 Typical Installation

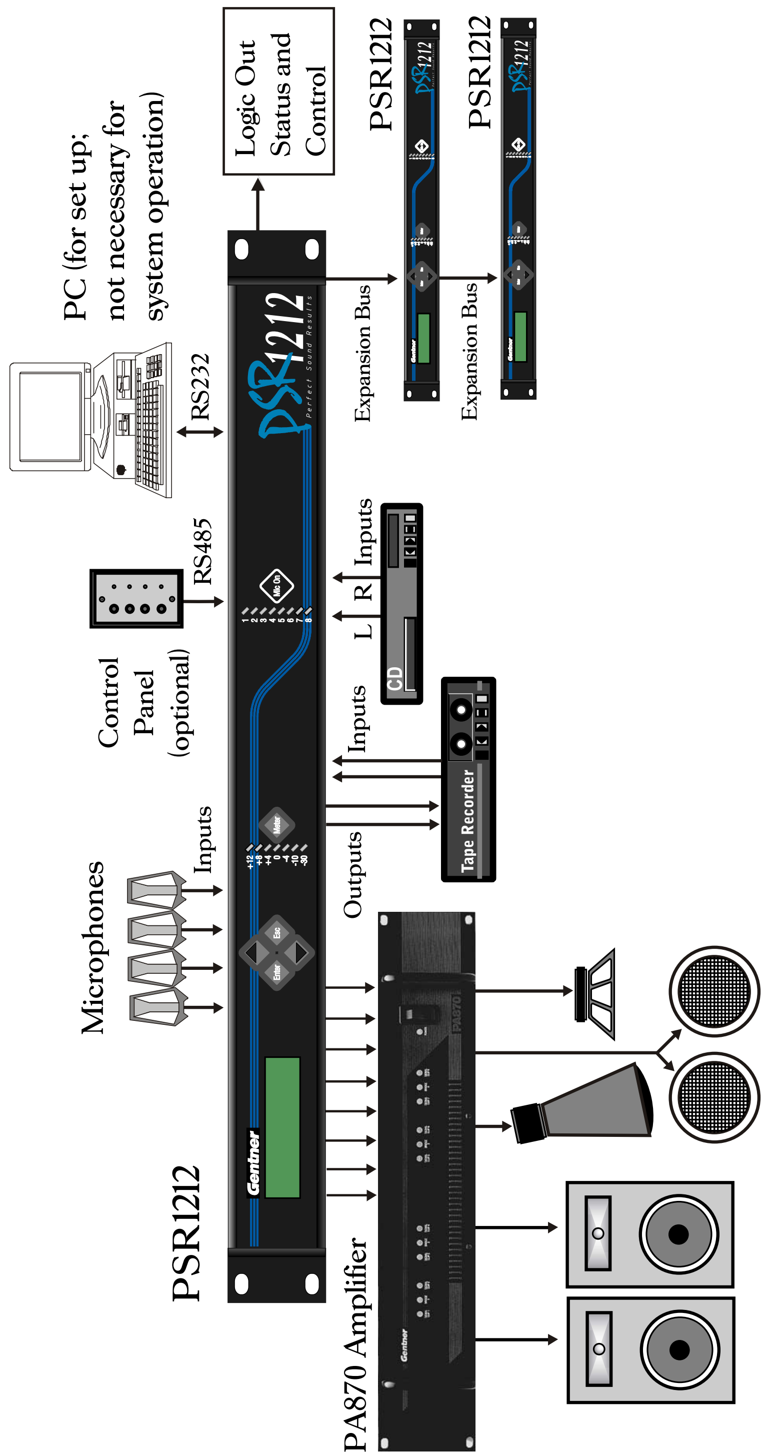

Figure 4. Typical PSR1212 Installation (simplified for illustrative purposes)

3 Applications

Introduction

The sophistication and adaptability of the PSR1212 allow it to control and enhance many sound applications. Following are nine applications where the PSR1212 forms the centerpiece of a high-quality sound reinforcement or room-combining system.

There are numerous other applications where the PSR1212 can control and enhance the audio experience. The principles used in the applications outlined here carry over into other applications.

Auditorium Installation

A typical auditorium application requires the use of multiple inputs and outputs, as well as audio equalization and signal delay, to provide a pleasing audio experience. The PSR1212 handles these tasks perfectly.

The auditorium diagram (Figure 5) illustrates a typical auditorium layout, with locations of microphones, speakers, and seating areas. Typical audio scenarios include on-stage speaking; singing and/or musical instruments; and pre-recorded music sourced from a CD or tape player. You can configure a preset for each scenario, using the PSR1212's filters to tailor the sound for a natural, balanced response.

Auditorium applications using constant directivity horn speakers located high at the front of the room are enhanced with the PSR1212's CD Horn EQ. This feature compensates for the inherent 6dB/octave high frequency rolloff typical of CD horn drivers. You can program this function for each preset you use for auditorium applications.

Audience members seated underneath the balcony are shielded from some of the output from the horn speakers located at the front of the room. To compensate, fill speakers are used in the ceiling underneath the balcony overhang (see Figure 5). To eliminate the imbalance caused by sound reaching a listener's ears at different times from the fill speakers and the horn speakers, you can program the PSR1212 to introduce a delay to the fill speakers. The PSR1212's G-Ware software can calculate distances in feet and meters to help establish the amount of delay required.

Auditorium

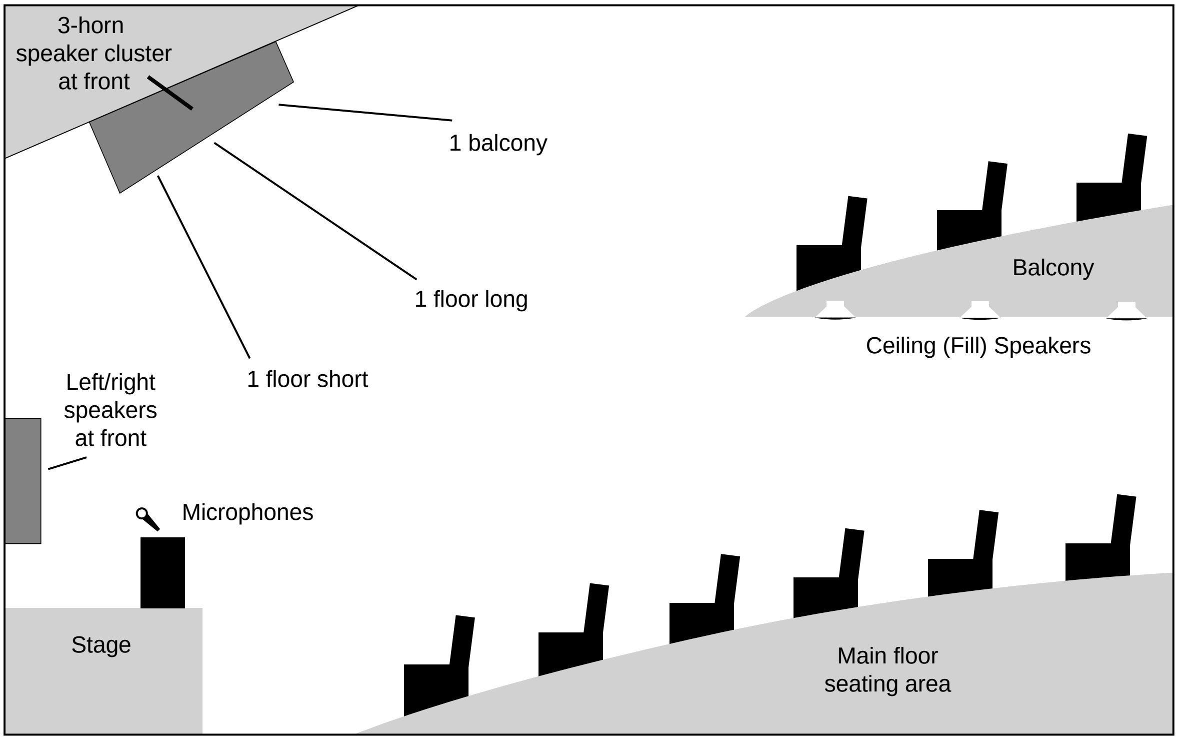

Figure 5. Auditorium

PSR1212 Auditorium Installation (basic example)

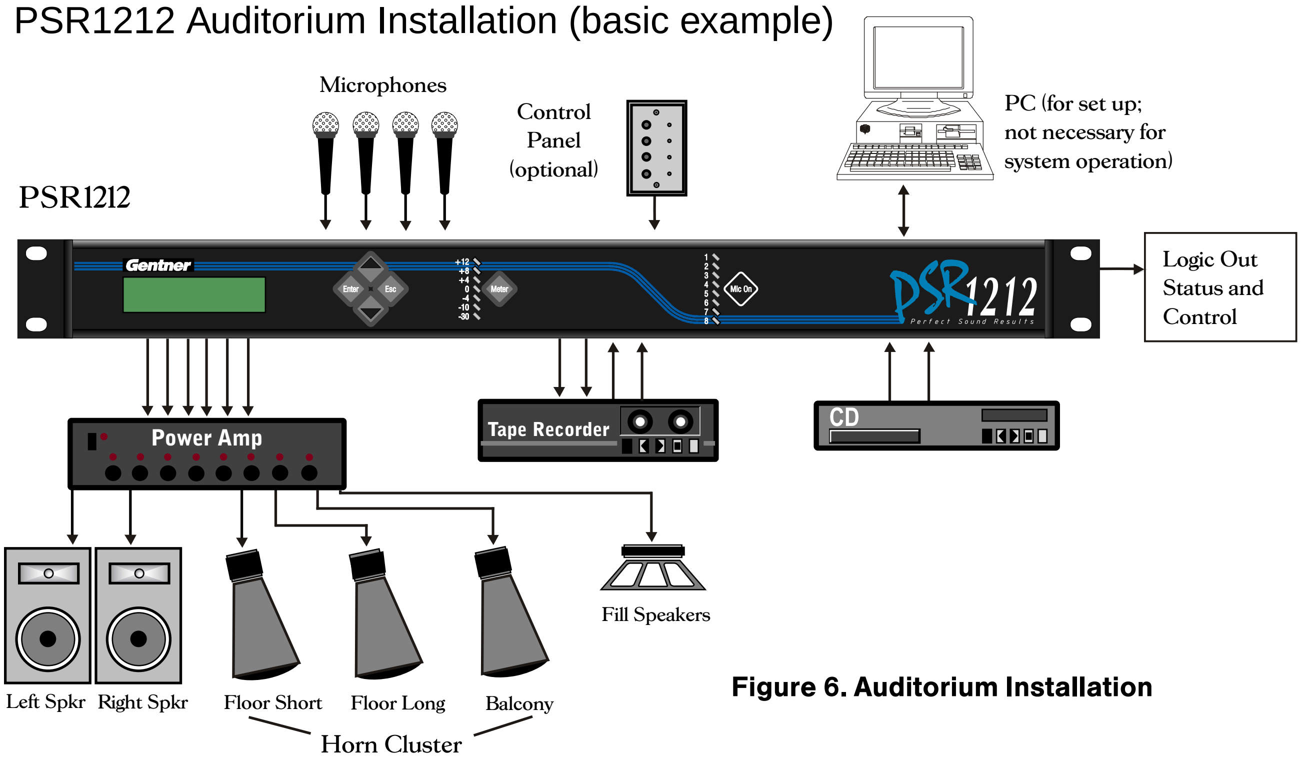

Figure 6. Auditorium Installation

Arena Installation

Arenas (see Figure 7) present numerous acoustical challenges, including high ambient sound levels and generally poor acoustical characteristics. The PSR1212 can help adapt the osund system to challenging acoustical environments.

The multitude of sound requirements in an arena is perfect for taking advantage of the PSR1212's Expansion Bus technology, which allows multiple PSR1212s to be linked and operated as a single unit. The arena diagram (Figure 7) and the arena installation diagram (Figure 8) illustrate the PSR1212's ability to distribute balanced audio to the main arena area, the fill locations underneath balcony or upper bowl areas, and concession areas located outside the arena.

During a typical sports event, you might want to use a PSR1212 preset configured to send the announcer's voice to the central speaker array (with the CD Horn EQ enabled), with a delayed signal sent to the fill speakers. Rally music, either pre-recorded or from an instrument such as an organ, can be routed to the same speakers, with various filters configured to enhance the audio quality of the source. The concession area speakers are used to announce countdown to game time and other information to people outside the main arena area.

Configure a preset for halftime entertainment that routes several microphones and live or pre-recorded music to the central speaker cluster and fill speakers, with microphone mix, delay, filter, and EQ settings configured for balanced audio response.

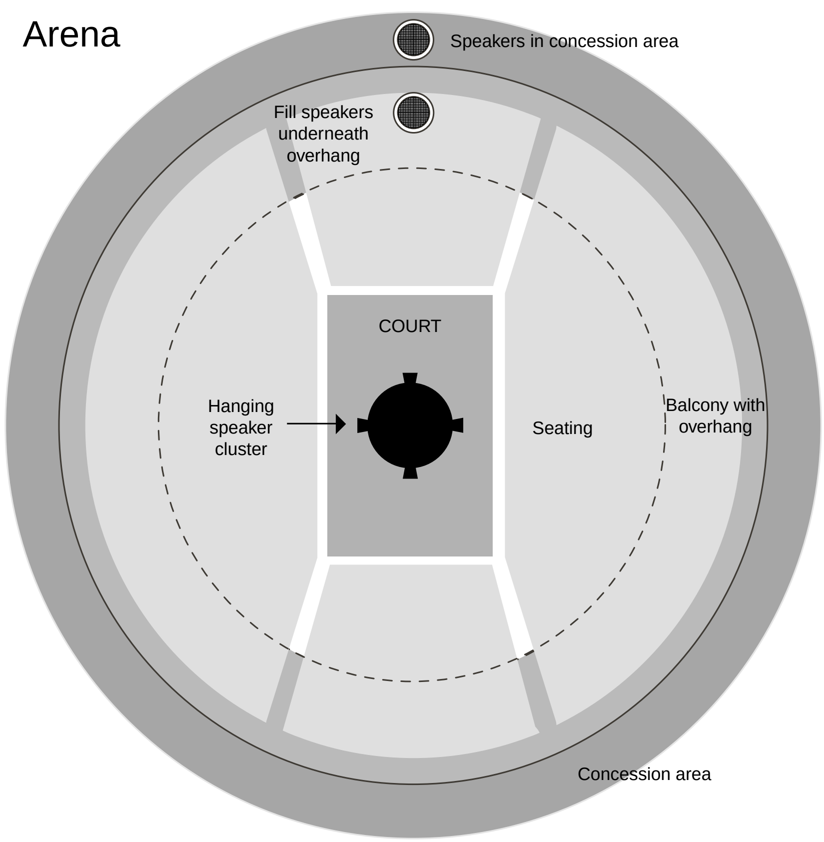

Figure 7. Arena

PSR1212 Arena Installation (basic example)

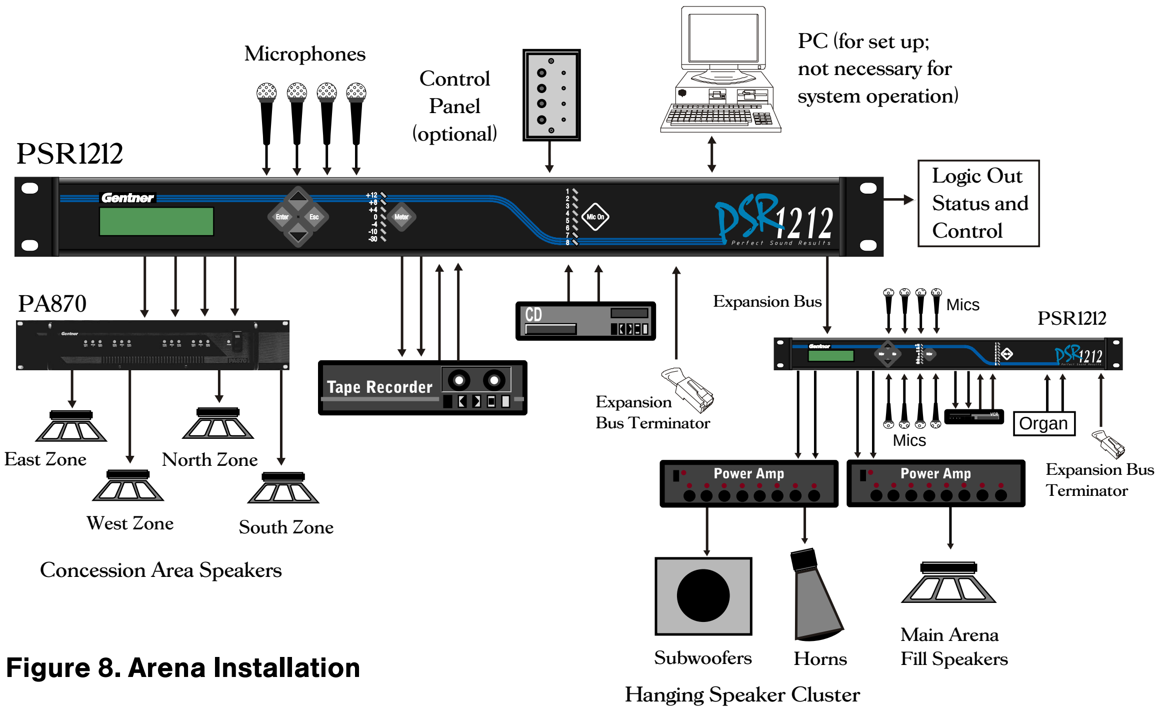

Figure 8. Arena Installation

Movie Theater Installation

A properly designed and calibrated sound system makes all the difference in the enjoyment of a film or movie presentation. Unlike with many other audio application scenarios, movie theaters are designed to exhibit good acoustic properties. Most sound system calibrations center around creating a broad, flat, high-fidelity audio response throughout the seating area.

A multiple-channel theater surround sound system is shown in Figure 9, and a connection diagram is shown in Figure 10. The PSR1212 can customize the response of each loudspeaker through the use of parametric equalizers, filters, and compressors designed to compensate for speaker and room characteristics and deficiencies. A complete description of these adjustable parameters (and how to configure them) is found in the PSR1212 Installation and Operation Manual.

A CD player and tape deck can be connected to play background music between movies – over both the theater and lobby speaker systems, and a microphone can be used to facilitate the occasional announcement.

Movie Theater Surround System

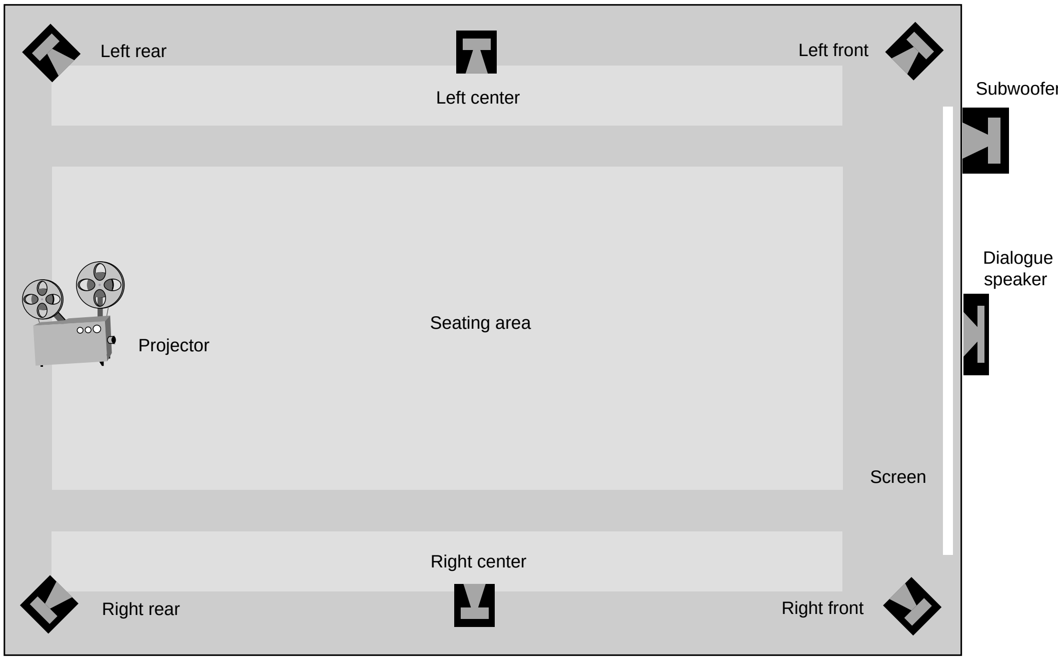

Figure 9. Movie Theater

PSR1212 Movie Theater Installation (basic example)

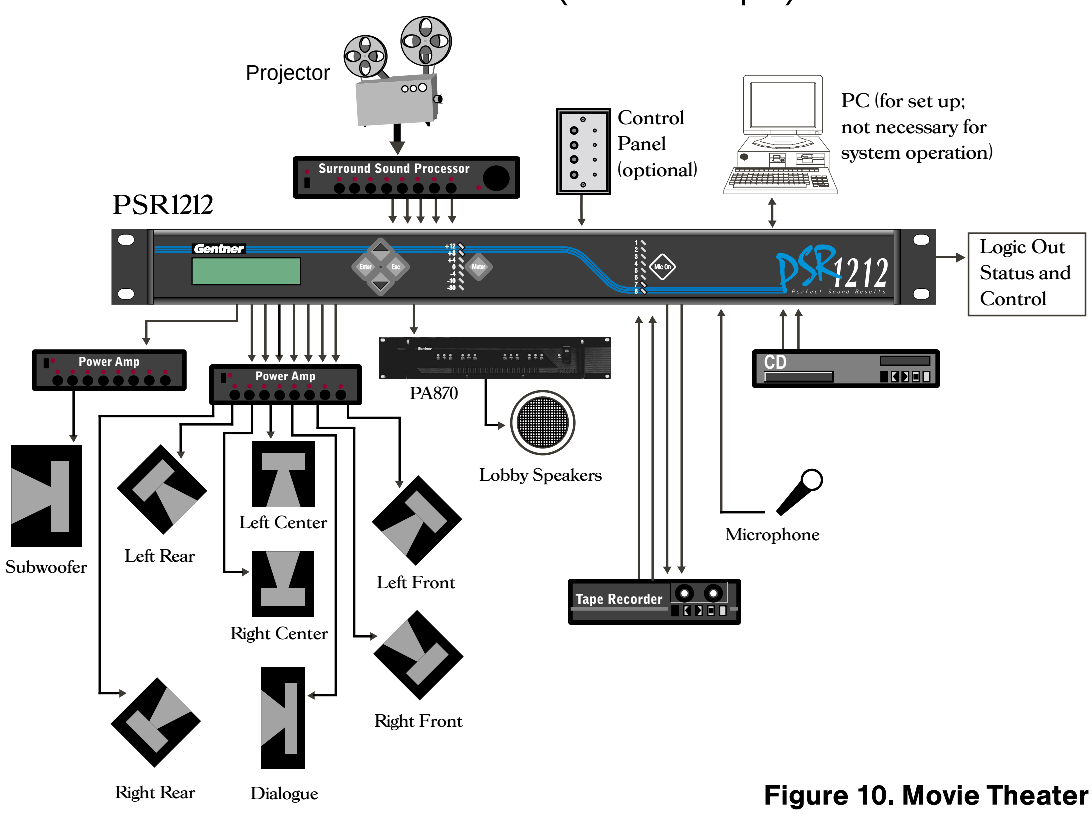

Figure 10. Movie Theater Installation

Gymnasium Installation

Gymnasium sound systems are called upon to meet the needs of a variety of scenarios, from sports announcing to meetings to student skits and productions. In all cases, the PSR1212 can adapt the sound system to accommodate the demands required.

For general announcing or presentations, such as for games or assemblies, presets can be configured for one or more microphones, with mixing and gating, if desired.

More elaborate configurations would use the PSR1212's parametric equalizers and filters to create better sound fidelity from live or pre-recorded music played over horn speakers typically used in gymnasiums. An example would be establishing a preset for halftime performances, where the equalizers and filters are configured to provide stronger low-frequency response from music sources.

Gymnasium

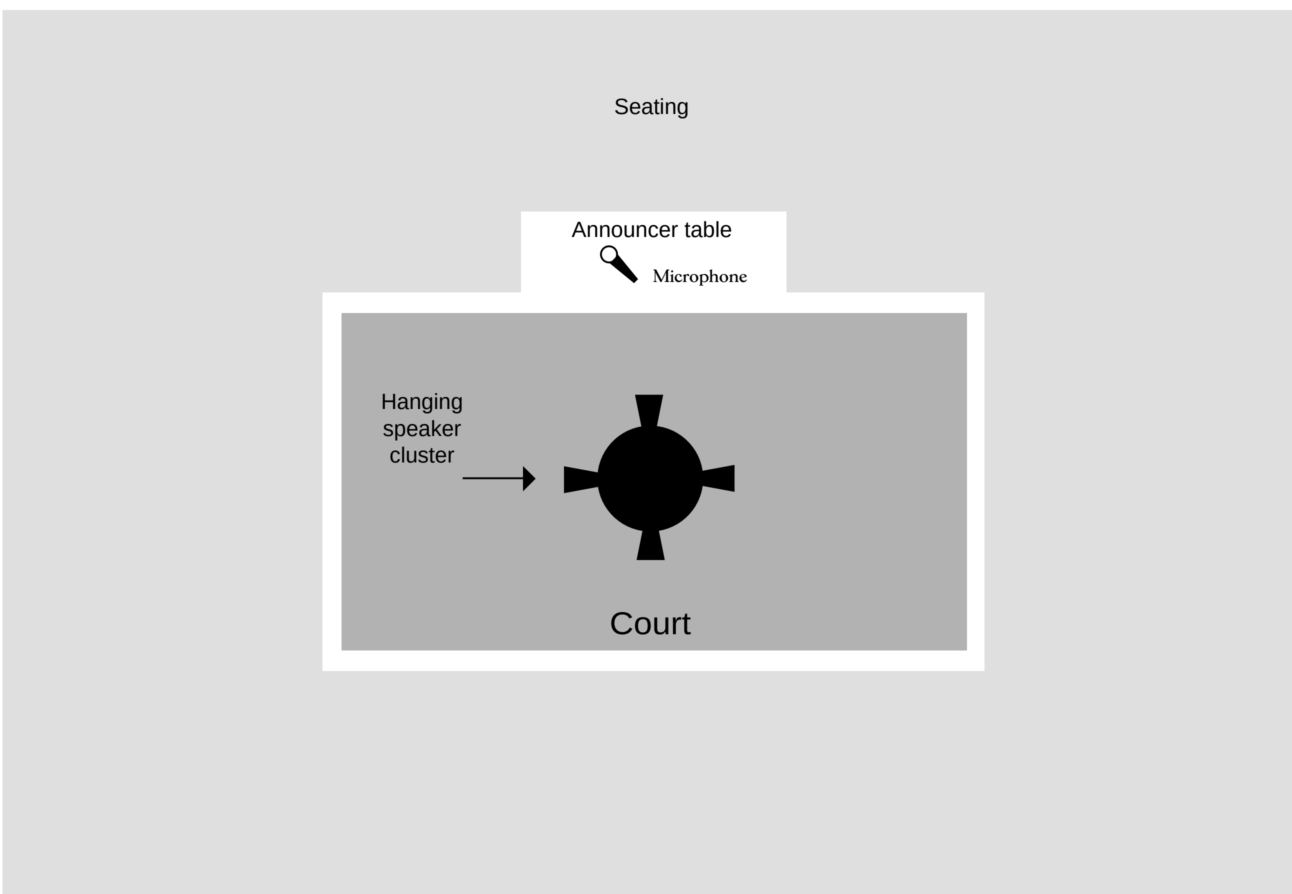

Figure 11. Gymnasium

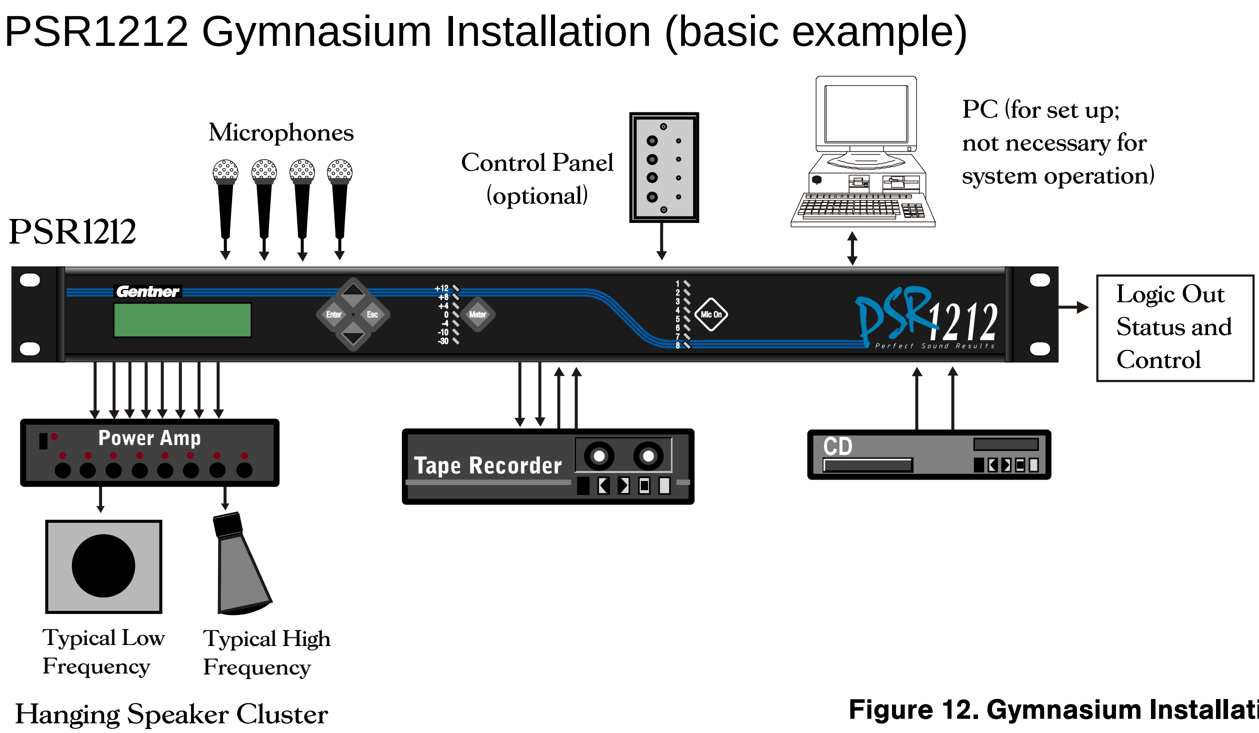

Figure 12. Gymnasium Installation

Hotel/Convention Center Installation

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 PSR1212, 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 13 shows four rooms with removable partitions. The PSR1212 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.

The use of other audio sources can be configured using the PSR1212's parametric equalizers and filters to enhance audio quality.

Hotel/Convention Center

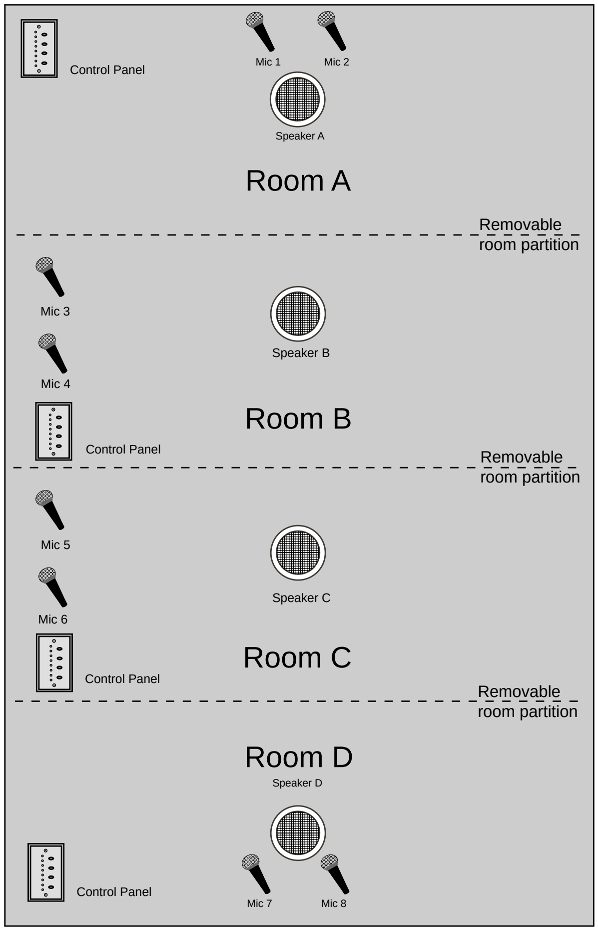

Figure 13. Hotel/Convention Center

PSR1212 Hotel/Convention Center Installation (basic example)

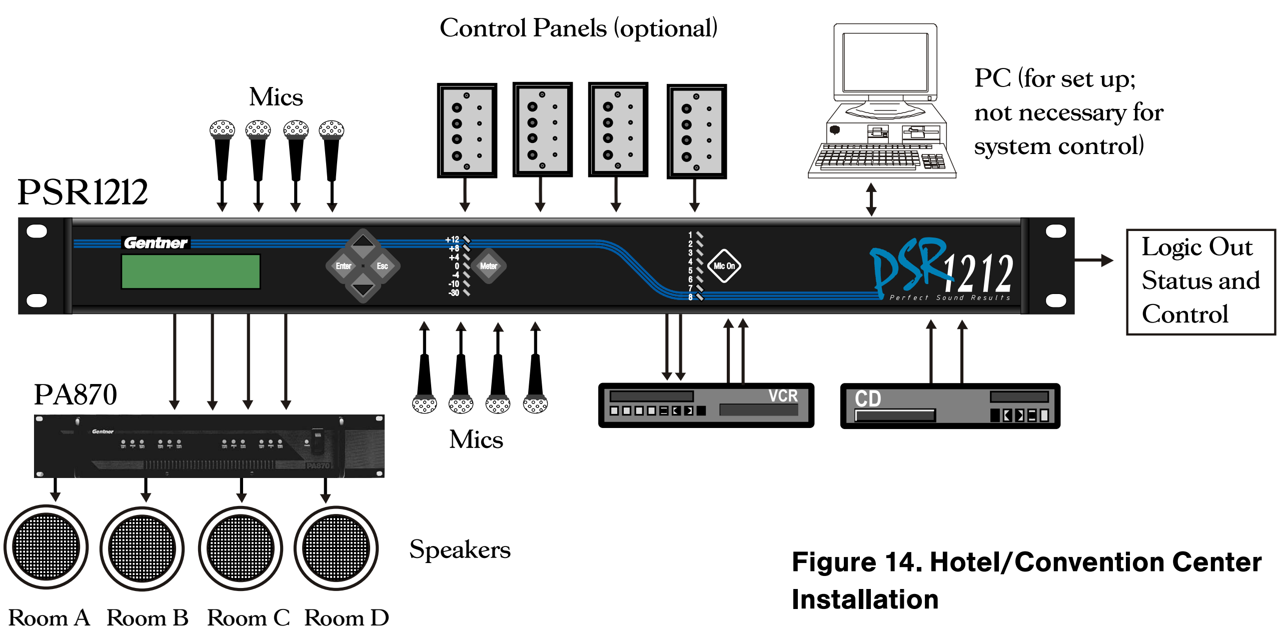

Figure 14. Hotel/Convention Center Installation

Conference Room Installation

Figure 15 shows a conference room area with a main meeting area and six smaller meeting rooms adjoining the main area. The PSR1212 enhances the flow of audio between the main meeting area and the adjoining small rooms, and can also facilitate exclusive communication between the small rooms.

Scenarios where these kinds of audio flow are necessary include team-building exercises, where participants meet in smaller groups and then report progress to the main meeting area. Another scenario is one in which emergency officials from various agencies assemble in the main meeting area to coordinate response activities to an emergency. These officials convene in the smaller rooms as needed to assess needs and troubleshoot problems at the agency level while remaining accessible to the main meeting area.

A PSR1212 preset could be established to give gating priority to the podium microphone in the main meeting area, with audio routed between rooms as desired to provide a seamless communication flow.

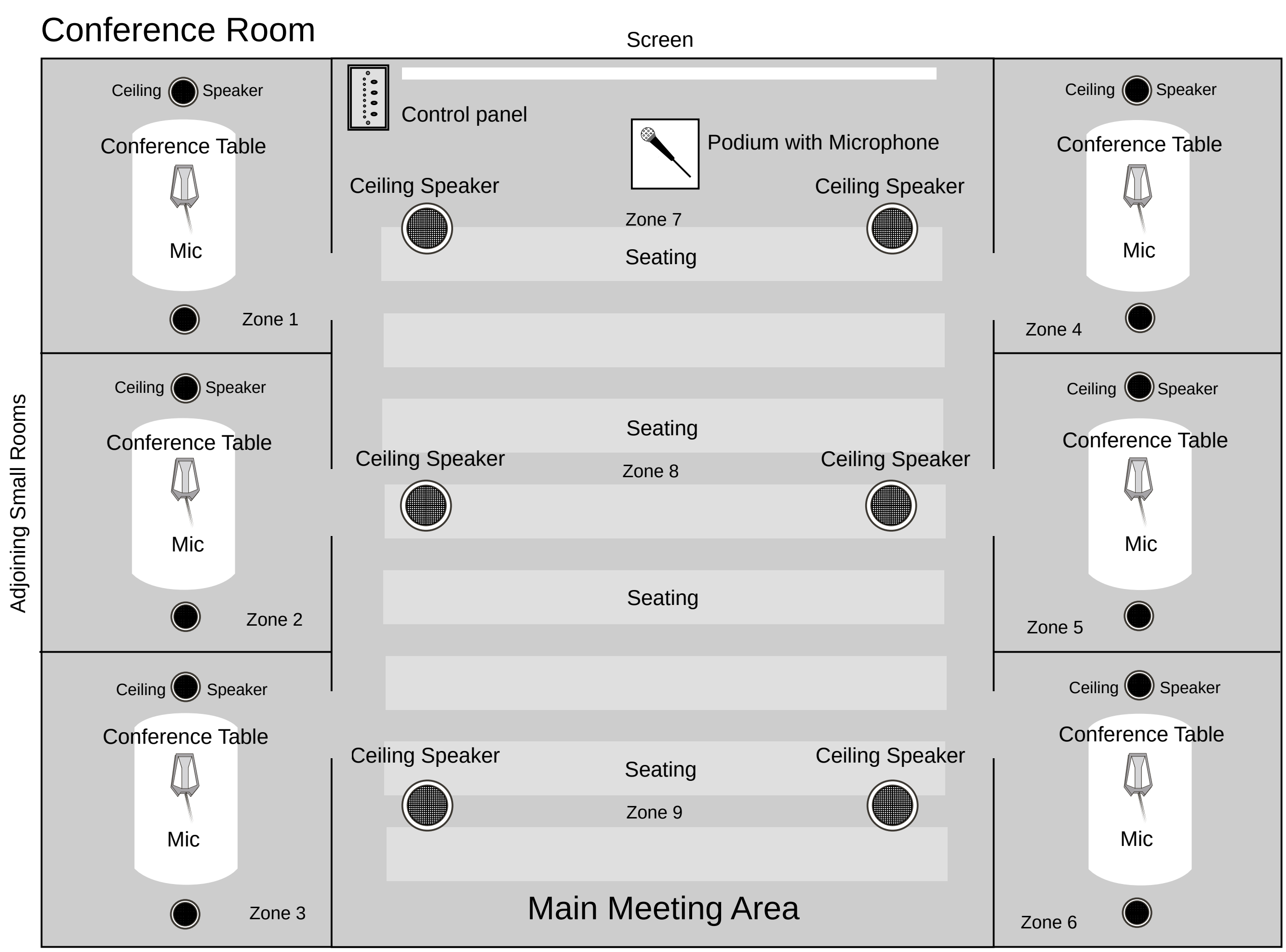

Figure 15. Conference Room

PSR1212 Conference Room Installation (basic example)

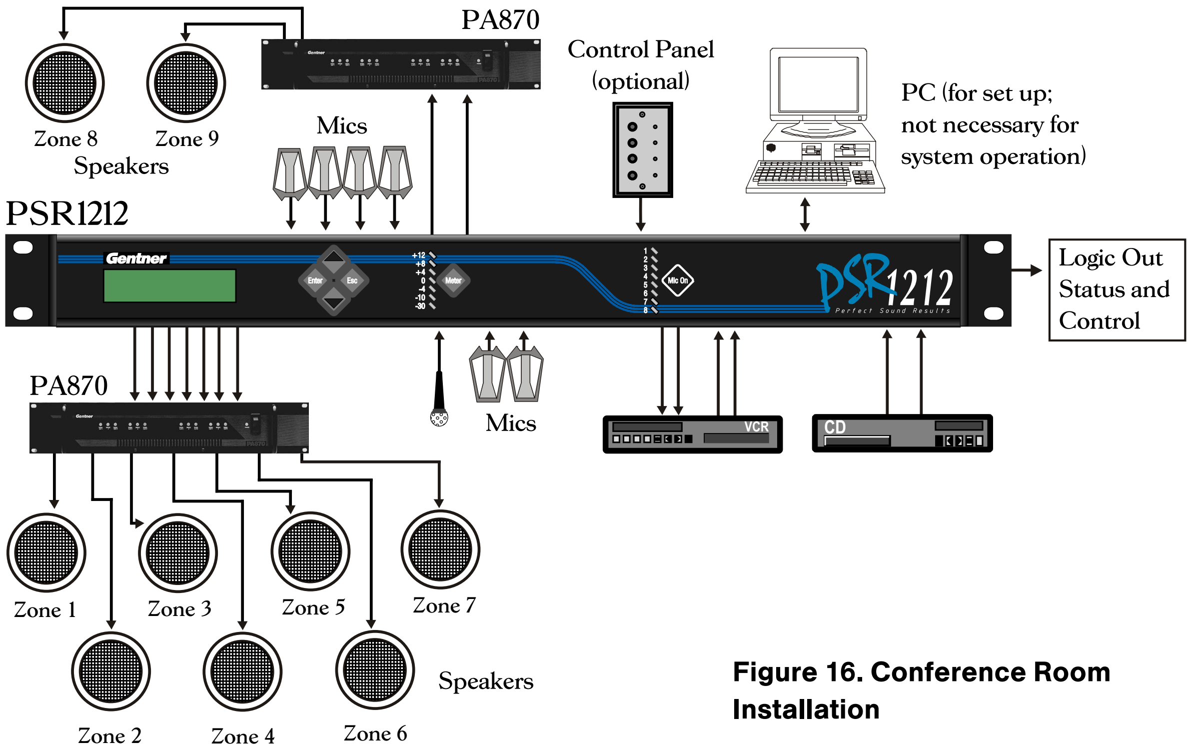

Figure 16. Conference Room Installation

Training Room Installation

In a typical training room application, microphone audio from the trainer is the primary source of audio. A wireless mic is normally used. Secondary audio is often provided through presentation segments sourced from a VCR or CD player. In larger training room settings, participants often use desktop microphones to enable all to hear questions and comments.

Figure 17 shows a scenario including a wireless lapel mic for the trainer; desktop mics 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.

Microphone mixing and gating parameters can be set to favor the trainer's mic to facilitate effective dialogue in the room. When a particular microphone gates on, nearby speakers can be set to reduced volume 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 PSR1212's parametric equalizers and filters accordingly.

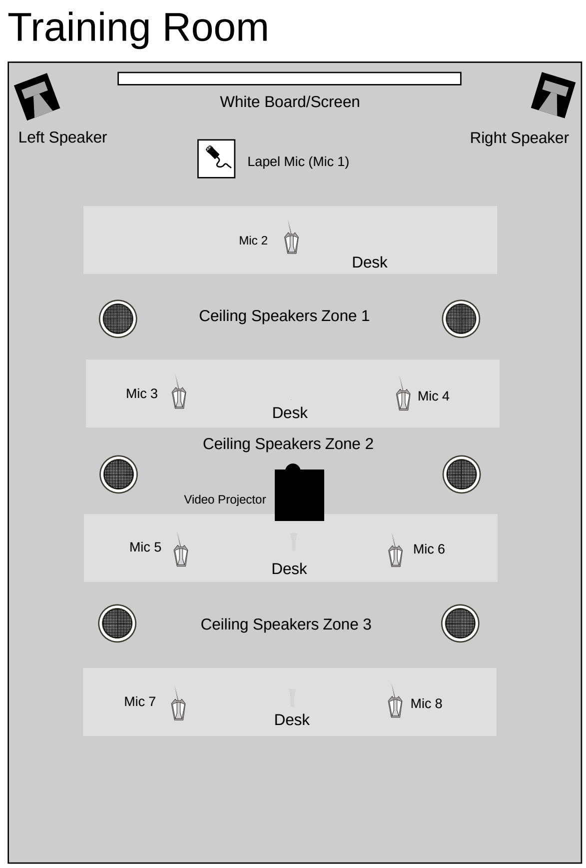

Figure 17. Training Room

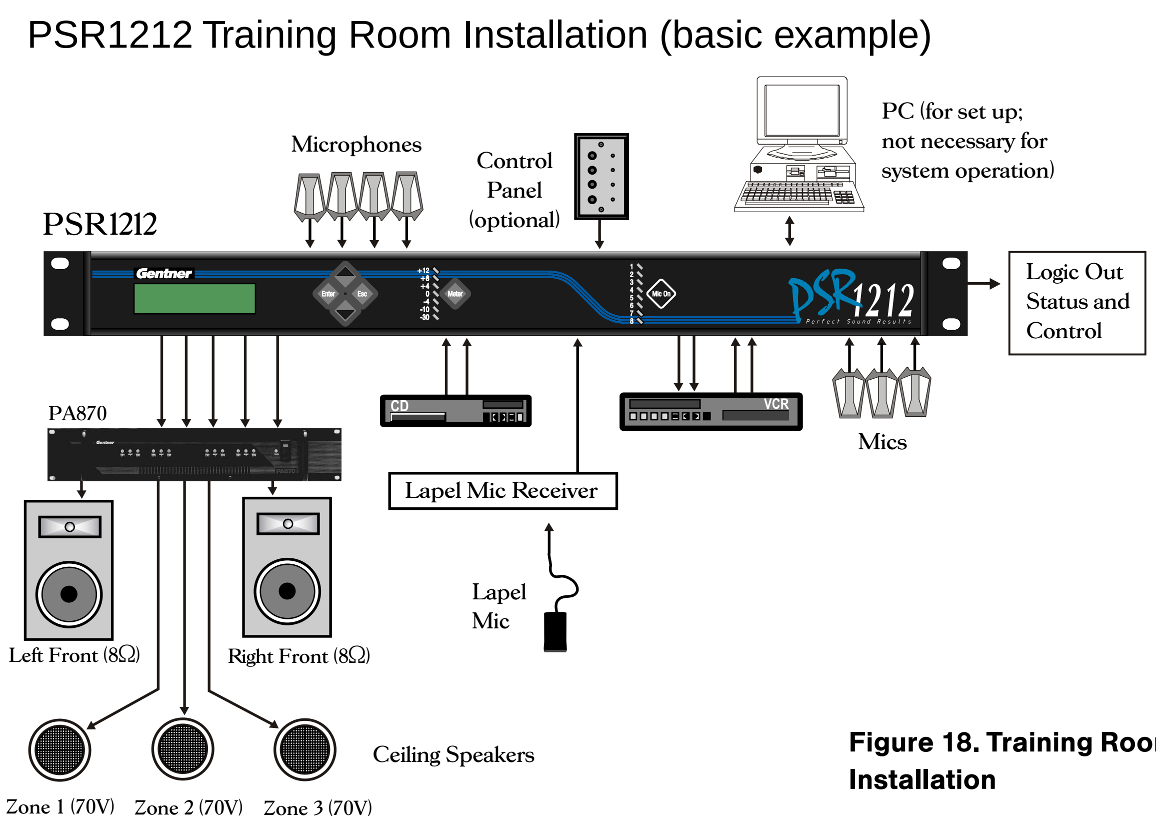

Figure 18. Training Room Installation

Boardroom Installation

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. Figure 19 shows a boardroom scenario with eight 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.

A typical PSR1212 installation would involve establishing a preset where the chairman's microphone (microphone 1) would be given gating and override priorities over the other microphones. Also, whenever a given microphone 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.

A preset for secondary audio, such as from a VCR or CD player, would route sound to all speakers in the room and gate off microphones as desired. The PSR1212's parametric equalizers and filters would be configured to enhance audio quality.

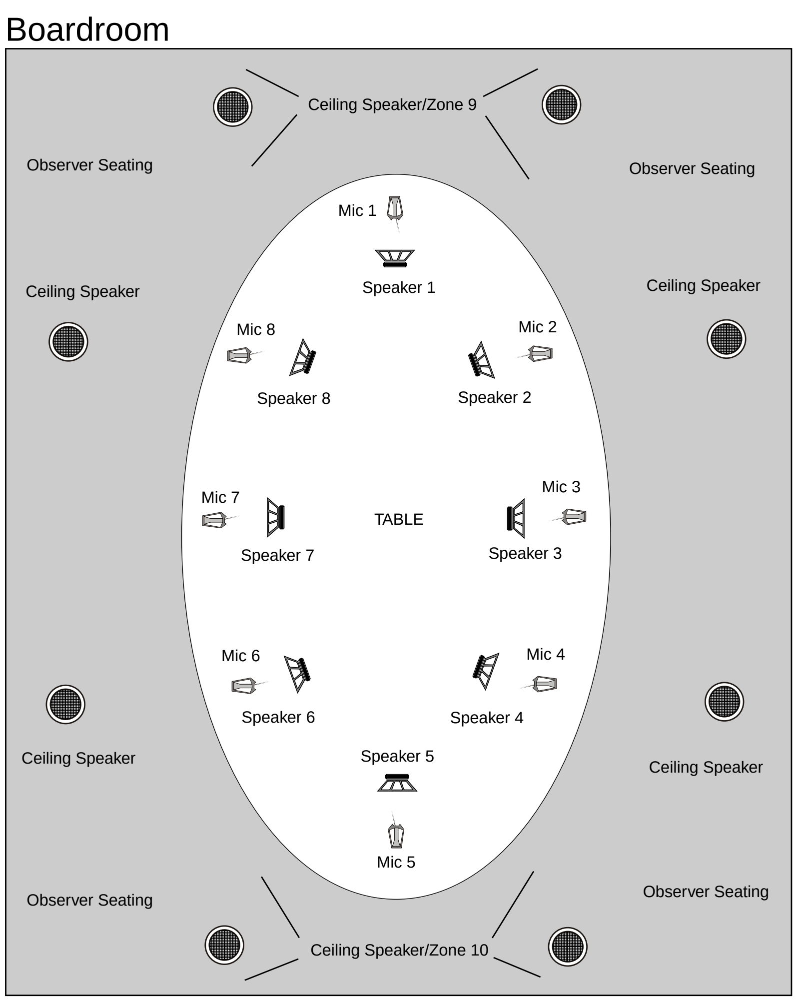

Figure 19. Boardroom

PSR1212 Boardroom Installation (basic example)

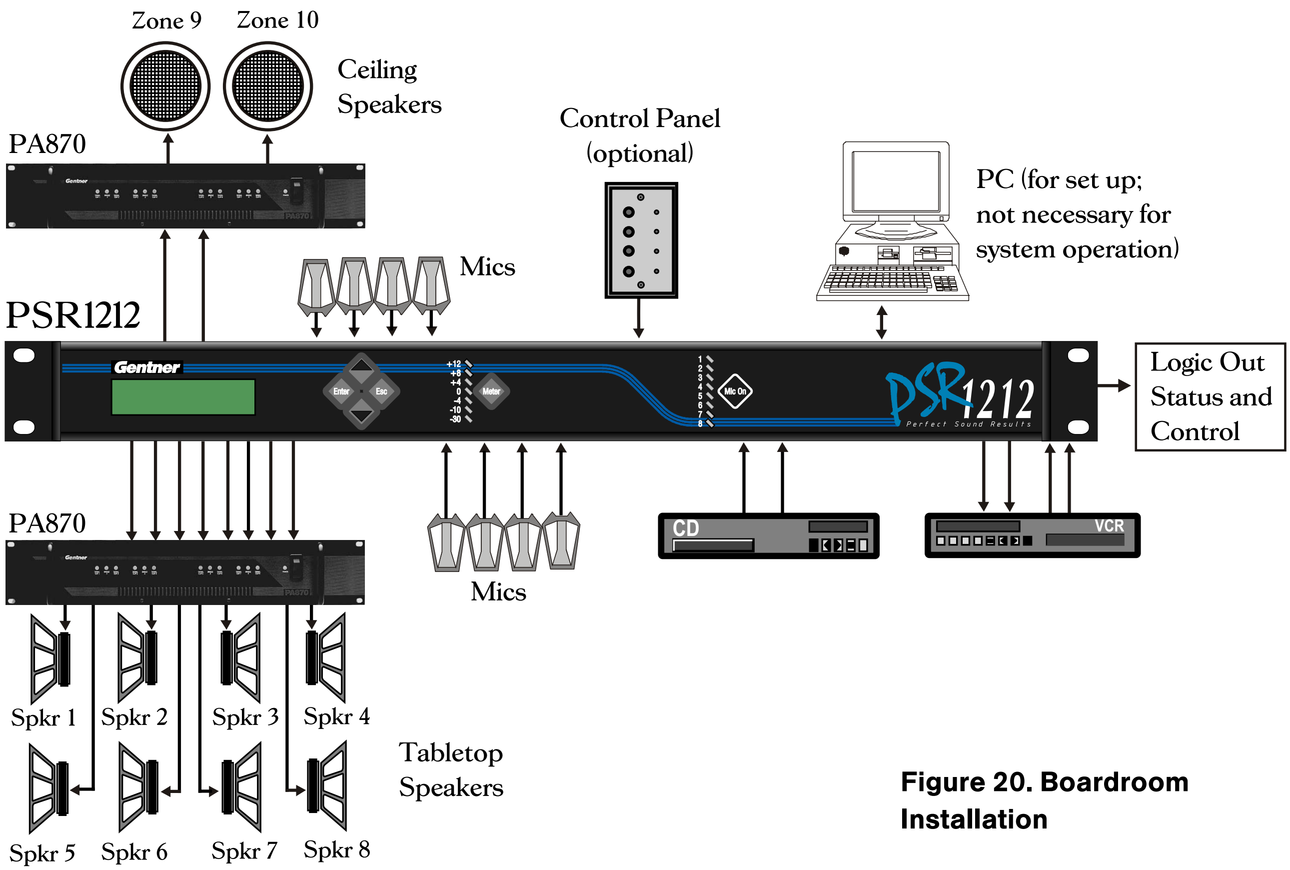

Figure 20. Boardroom Installation

Corporate Paging System Installation

A corporate paging system usually consists of several sound distribution zones, typically using 70V speakers, for paging various departments of an organization or areas within a building. Such systems are often used for playing background music, particularly in lobby areas.

With the PSR1212, you can establish customized paging zones where various equalizers and filters are configured to enhance the fidelity of sound in the system. This can help compensate for room acoustical characteristics and equipment deficiencies to provide a consistent audio response throughout a given zone or combination of zones.

Figure 21 depicts a paging system with nine zones. Each zone could have its own preset to facilitate paging only within the zone. Zones could also be grouped for broader paging capabilities, depending on requirements. A preset could be configured to page via all zones. This preset would be used to make organization-wide announcements, including emergency notifications.

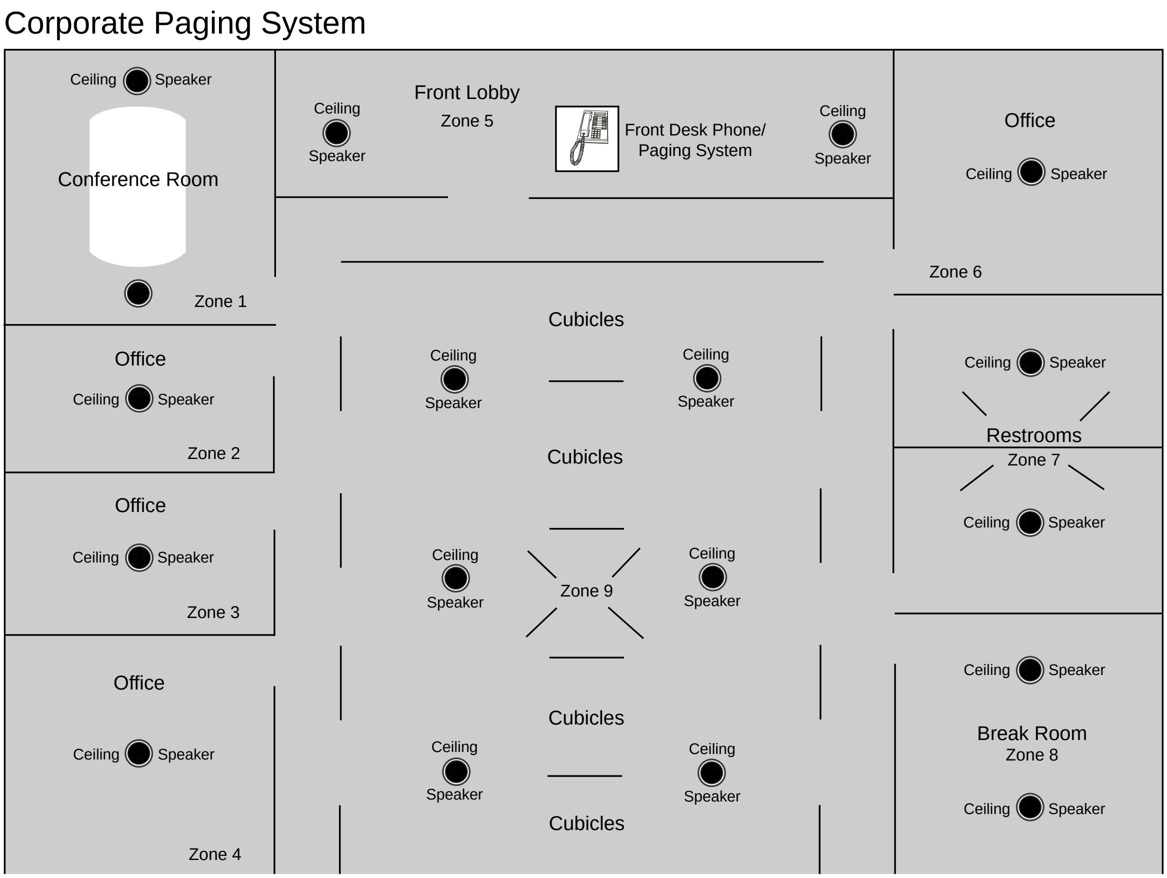

Figure 21. Corporate Paging System

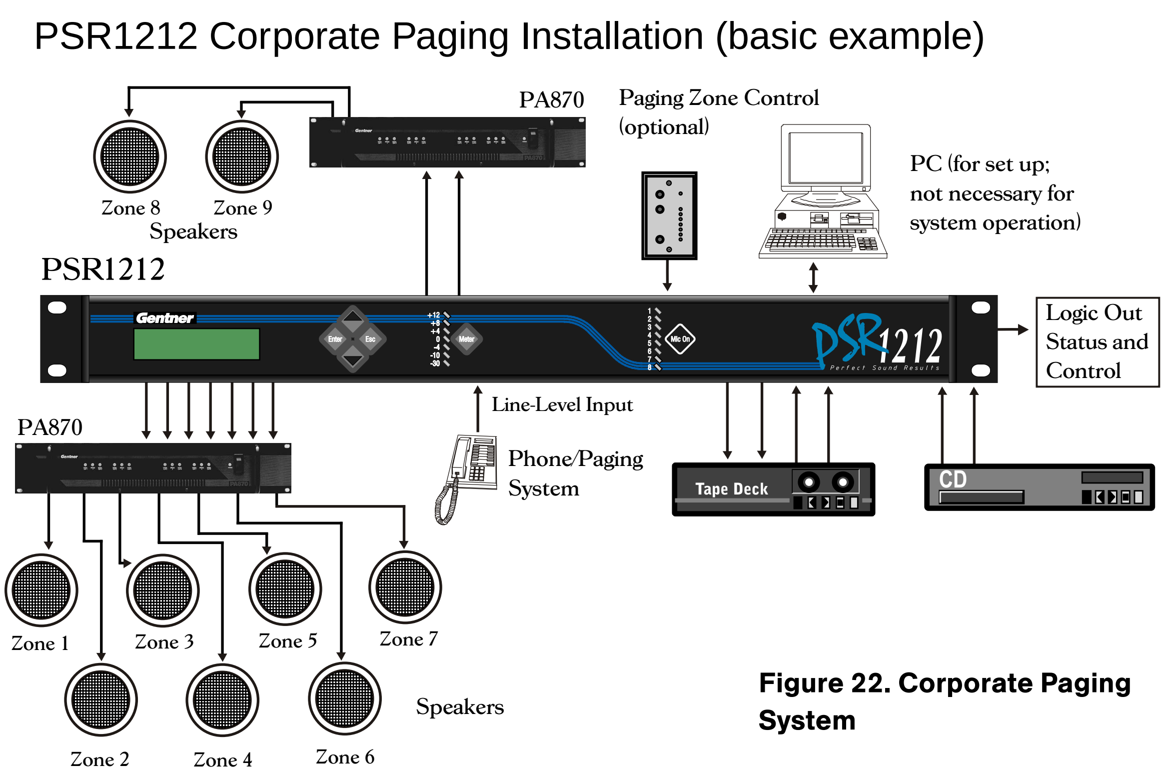

Figure 22. Corporate Paging System

4 Inputs and Outputs

Input and Output Level Control

The PSR1212 has 12 inputs consisting of eight mic/line inputs and four line inputs. The unit has 12 line outputs.

All inputs and outputs are actively balanced. Inputs 1-8 have 4kOhms of terminating impedance while line-level inputs 9-12 provide >20kOhms of termination. Outputs provide a source impedance of 50 Ohms. 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. The PSR1212 utilizes 24-bit A/Ds and D/As while sampling at a 48kHz rate. This results in a system-wide dynamic range of 100dB, a pass band from 20Hz to 20kHz, and a SNR >100dB at maximum input level. All input and output levels can be monitored in real time on the front-panel LCD 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.

Inputs 1-8
As shown in Figure 23, balanced audio is input at the rear panel Phoenix™ connector. Mic or line level is selected and phantom power is provided (if required). The PSR1212 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.

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.

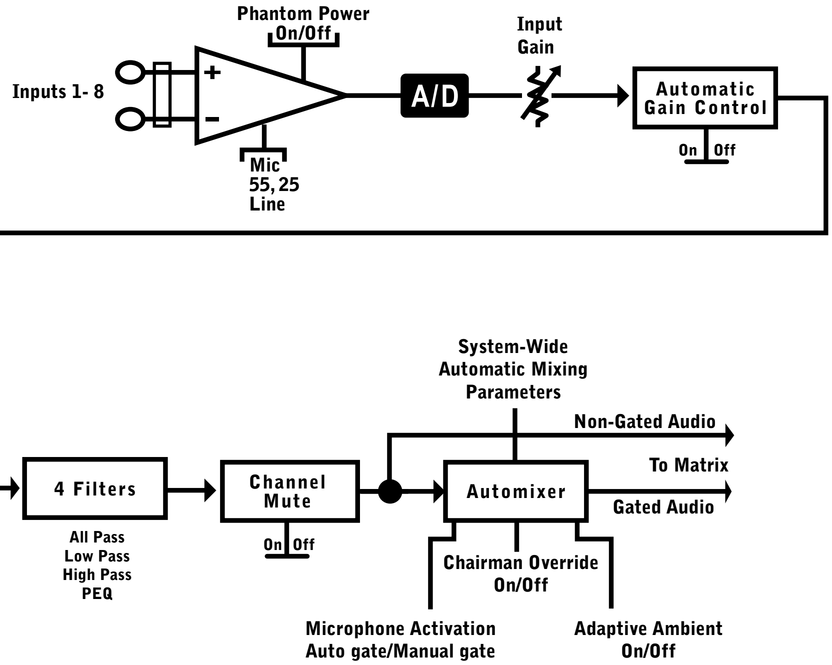

Figure 23. Inputs 1-8 Parameters

Next, four configurable filters can each be set individually as an all-pass filter, a low-pass filter, a high-pass filter, or a parametric equalizer (PEQ). 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 channel mute function is applied after the filter stage. 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 that are not in the chairman override group.

■ 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 24 shows inputs 9-12. These line-level inputs can be level controlled, muted, and gain controlled through the AGC. All of these functions operate identically to inputs 1-8.

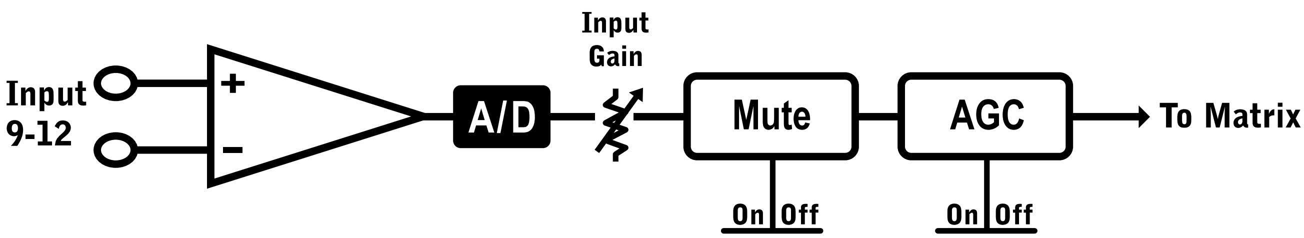

Figure 24. Inputs 9-12

Outputs
All 12 of the line outputs are identical, as shown in Figure 25. 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 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 occuring.

A feature of the PSR1212 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.

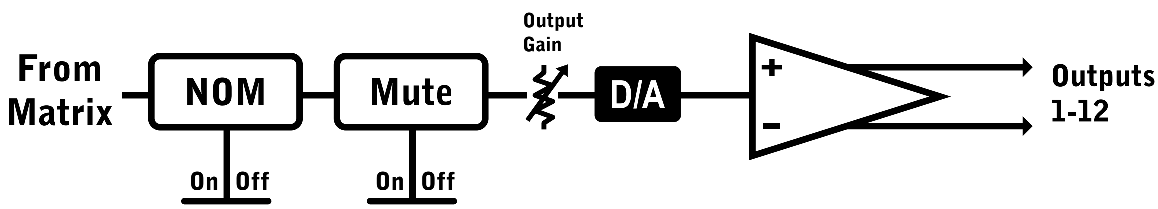

Figure 25. 12 Outputs from Matrix

5 Automatic Mic Mixing

Intelligible, Reliable Audio

Audio systems are in constant use in auditoriums, arenas, boardrooms, training rooms, and many other applications. Systems that produce intelligible and reliable audio are key to facilitating effective communication. Quality audio 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 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 26, 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.

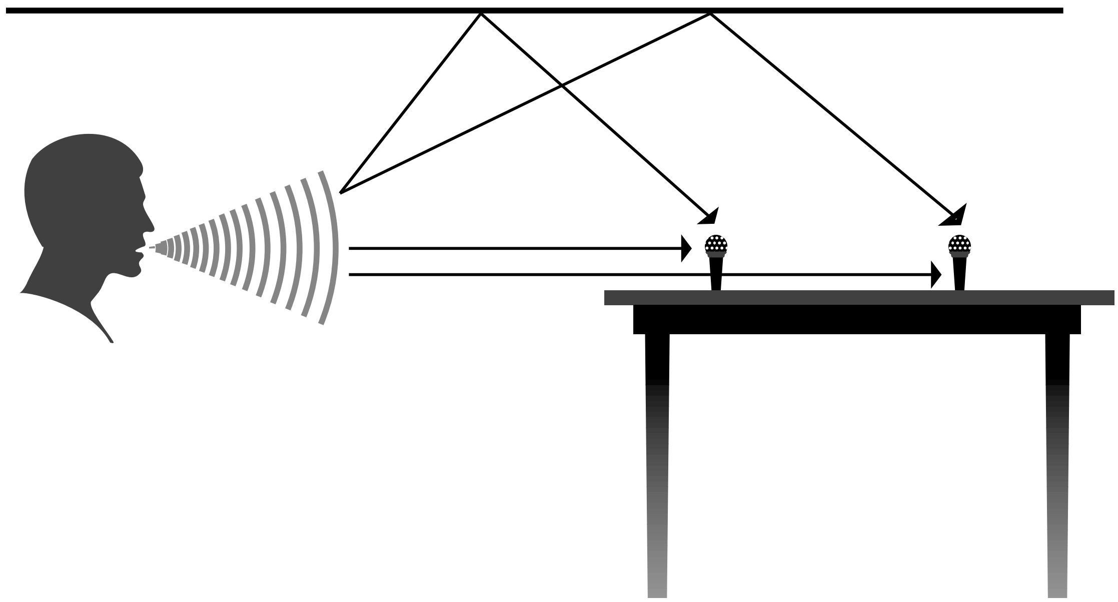

Figure 26. Microphones Pick Up Direct and Reflected Audio

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.
- Acoustically treat the room to reduce reverberation and noise.
- Eliminate or reduce the source of noise.

The PSR1212 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 PSR1212 implements its mixing function completely in the digital domain. This greatly increases precision in making automixing decisions.

All audio is routed through the PSR1212 (both microphone and speaker audio), which means the PSR1212 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. 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 PSR1212 (see Figure 27) because the unit can determine that this audio is not voice audio.

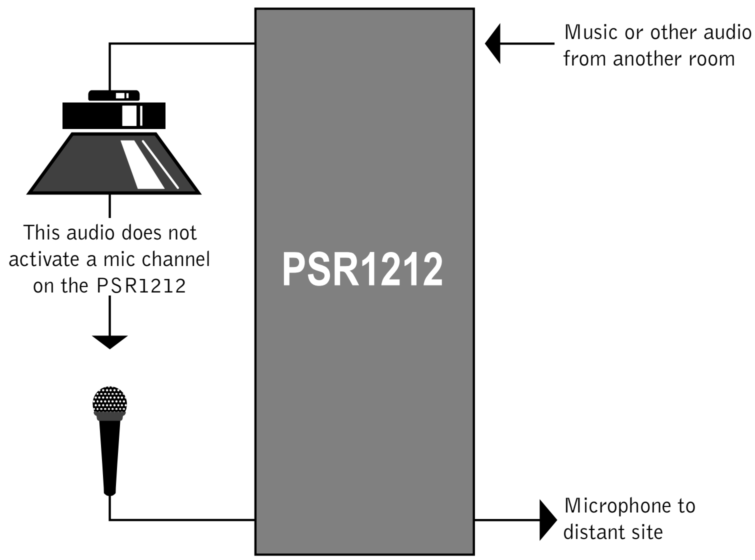

Figure 27. Microphone Activation

The PSR1212 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 PSR1212 can have four separate automatic mixers working independently or as a single unit. In addition, more microphone channels can be added by linking PSR1212 units via the Expansion Bus, the digital network bus. Unlike other "expandable" automatic microphone mixers, the PSR1212 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 PSR1212 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). See Figure 28.

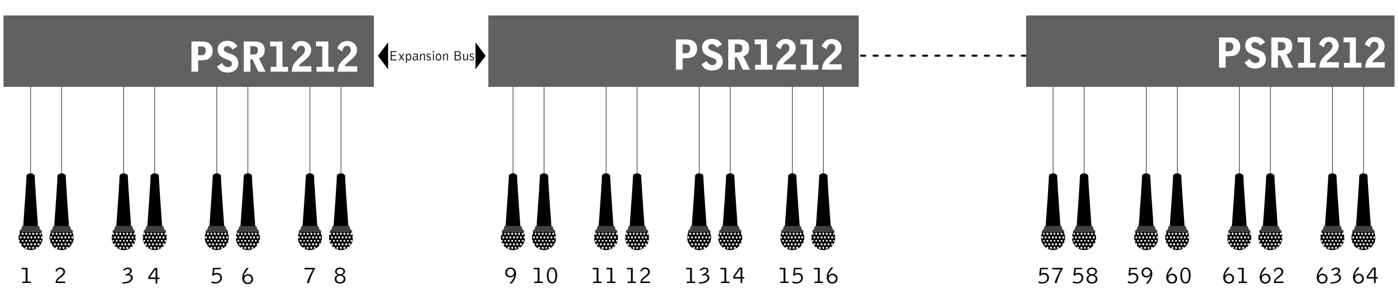

Figure 28. Expansion Bus Control of 64 Mics

The following parameters and modes are used on the PSR1212 to provide high precision and reliability in microphone mixing:

■ Mixer Mode. The PSR1212 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.

■ 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. As shown in Figure 29, the reason the PSR1212 can accurately determine when speaker audio is present is because audio goes through the PSR1212 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.

■ First Mic Priority Mode (On or Off). This is a useful feature for even better precision in microphone activation. This mode establishes a condition in which a particular microphone—typically the one with the most audio—is assigned gating priority over other microphones in a PSR1212 system.

■ 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 Mic 1-8 On when all other mics gate off. These features are useful to ensure the audio never goes completely away (see Figure 29). 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.

■ 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 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. 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.

■ Hold Time (.1 to 8.0 seconds). This programs the amount of time it takes until the microphone input starts the off attenuation process.

■ 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 PSR1212 solution set. Because all decisions regarding automixing are made by the same digital engine, better decisions in automixing can be made.

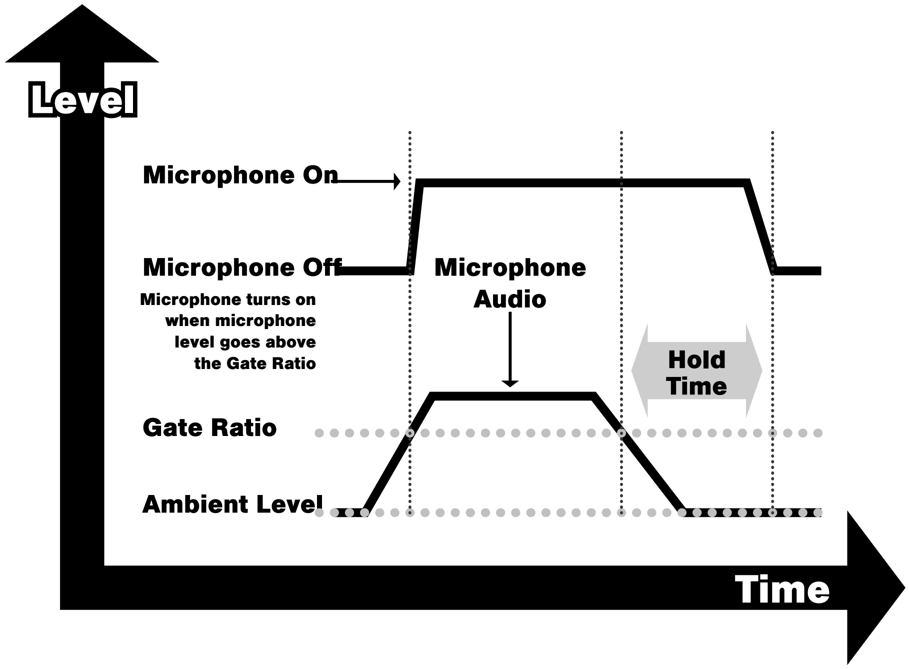

Figure 29. PSR1212 Automixing Gate Functions

Automatic microphone mixing is a key part of the PSR1212 solution set. Because all decisions regarding automixing are made by the same digital engine, better decisions in automixing can be made.

Mixing Parameter	System, Mixer, or Channel	Range	Description
Mixer Mode	System-wide	Master, Slave	Selects mixer mode of operation.
Microphone Activation	Input channel 1-8	Auto Gate, Manual Gate On/Off	Sets the method of microphone gating.
Chairman Override	Input channel 1-8	On, Off	When a chairman override channel is gated on, all non-chairman channels are gated off.
Adaptive Ambient Mode	Input channel 1-8	On, Off	Automatically sets the ambient audio level of the room averaged over time.
PA Adaptive Mode	Input channel 1-8	On, Off	This prevents mic channels from gating when distant audio or other non-microphone audio is heard through the speakers.
Maximum number of Microphones On	Mixer-wide	1-8 or Off	Sets the maximum number of micro phones 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 on.
Last Mic Mode	System-wide	Last, Mic 1-8, Off	Keeps the last gated microphone or Mic 1-8 on when no mics are providing a gating input.
Gate Ratio Adjust	Input channel 1-8	0 to 50dB	Specifies how much louder above the ambient level the audio level must be to gate on.
Off Attenuation Adjust	Input channel 1-8	0 to 50dB	Sets how much the microphone will be attenuated when it is not gated.
Hold Time	Input channel 1-8	.1 to 8.0 Seconds	Programs the amount of time it takes until the microphone starts the off attenuation process.
Decay Rate	Input channel 1-8	Slow, Medium, Fast	Programs how quickly the audio level is attenuated once a channel hold time has expired.
Manual Ambient Level	Input channel 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 30. Mixing Parameters

6 Audio Routing

Matrix Mixing

One of the more important functions of the PSR1212 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 PSR1212 audio matrix has 40 possible input sources and 32 output destinations, with level control at each cross point. The routing chart (Figure 32) describes the default PSR1212 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 below.

Inputs

<u>Gated and Non-gated Inputs 1-8</u> - Inputs 1-8 (selectable for mic or line level) 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 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).

<u>Inputs 9-12</u> - These are line-level inputs that appear on the rear panel terminal blocks. This is typically audio that comes from a CD player, VCR, and other auxiliary audio sources. In typical applications, this audio must be heard in the local PA system (as well as networked PSR1212 units). In the default routing, audio is routed to every other device except itself.

Outputs

<u>Outputs 1-8</u> - These are exactly the same as outputs 9-12. Their default routing is for each non-gated input 1-8 to go directly to these outputs.

Outputs 9-12 - These are line-level outputs that appear on the rear panel terminal blocks. This is typically audio that goes to a power amp w/speakers, VCR, tape recorder, or other auxiliary audio device. Normally, this audio contains auxiliary audio and audio from other networked PSR1212 units. In the default routing, inputs 9-12 (minus your channel input) and master auxiliary mix (all auxiliary audio from other PSR1212 units) are contained in this audio.

Expansion Bus

Expansion Bus This is a digital bus that appears at every PSR1212 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 PSR1212 can be placed on a bus or audio can be taken off a bus and routed to any destination within the unit. The PSR1212 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 PSR1212s. 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 or VCR, 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 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. Here's an example to clarify: Let's say you have four PSR1212 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 microphones. 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 PSR1212. 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, limited, and attenuated to provide specific enhancements to the audio (see Figure 31). These buses would typically be used to reduce feedback in the venue and provide crossovers for different speaker systems.

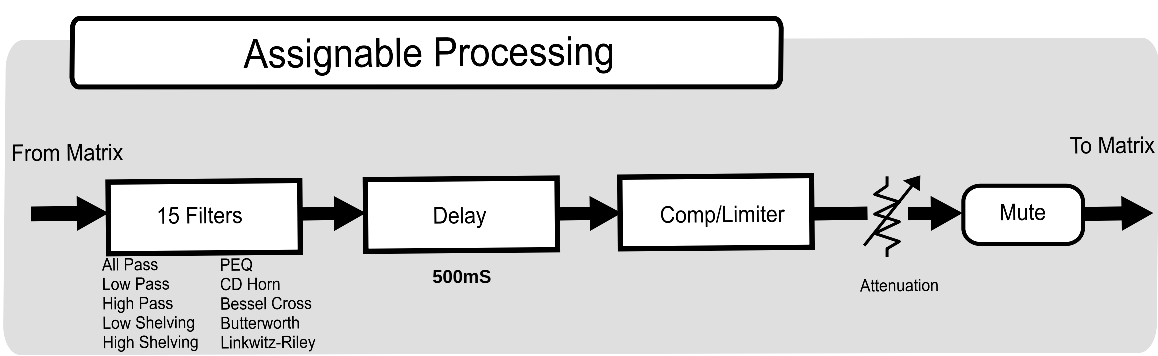

Figure 31. Assignable Processing

PSR1212
Routing Matrix Diagram
Application Name: Default

			Outputs 1 - 12												To Expansion Bus												To Processing							
															Inputs 1 - 8 to other PSRs				Inputs 9 - 12 to other PSRs															
Notes:			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
	In 1 - 8	1	N												G																			
		2		N											G																			
		3			N										G																			
		4				N									G																			
		5					N								G																			
		6						N							G																			
		7							N						G																			
		8								N					G																			
	In 9 - 12	9										X	X	X					X															
		10									X		X	X					X															
		11									X	X		X					X															
		12									X	X	X						X															
Inputs 1-8 from other PSRs	From Expansion Bus	O																																
		P																																
		Q																																
		R																																
Inputs 9-12 from other PSRs		S									X	X	X	X																				
		T																																
		U																																
		V																																
		W																																
		X																																
		Y																																
		Z																																
	From Processing	A																																
		B																																
		C																																
		D																																
		E																																
		F																																
		G																																
		H																																

0 Gated with cross point attenuation 0 Non-Gated with cross point attenuation 0 Cross point with attenuation OR G = Gated N = Non-Gated X = Cross point

Figure 32. Default Routing Diagram

Figure 32 shows the PSR1212's Default Routing Diagram. The unit defaults to the parameters indicated until you establish routing information.

System Parameters Worksheet

Figure 33 is a system parameters worksheet that indicates all of the system-wide programming functions of the PSR1212 (except for control/status pin configurations). It is provided to help you understand all PSR1212 system functions as you design a system. Both the default routing diagram and the programming worksheet are provided as loose worksheets for designing, programming and troubleshooting the PSR1212.

PSR1212 System Parameters Worksheet

Gentner PSR1212

System-Wide Parameters

Program Parameter	**Selection Range**		**Program Parameter**	**Selection Range**	
Timeout	0 - 15 (**5**)		Mixer Mode	**Master**, Slave	
Lock Front Panel	On, **Off**		RS232 Baud Rate	9.6, 19.2, **38.4**, 57.6, 115.2 kbps	
Set Passcode	Any 5 Front Panel Keys		RS232 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

		Internal/Unit Mixer Groups				Global/Site Mixer Groups			
Program Parameter	**Selection Range**	**1**	**2**	**3**	**4**	**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 Expansion Bus References

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

Hints:
Each Expansion Bus PA Adapt reference can only be defined once across the system (all linked PSR1212s).
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**

Figure 33. System Parameters Worksheet

Room Preset/Configuration Worksheet

The Room Preset/Configuration Worksheet (see page 63) is used for recording preset information such as description, command list, port usage, and other parameters.

Values in bold on the worksheet indicate the default values. The PSR1212 has 32 configurable presets.

Input/Output Parameters Worksheet

The Input/Output Parameters Worksheet (Figure 34) details each configurable input, output, and processing channel parameter and provides space to record settings for each. Default settings appear in bold.

PSR1212 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, 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			

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 (**0**)												
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/Limiter	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. In 5ms **(1s)**								
Processing Attenuation	0dB to -60dB **(0dB)**								

Gentner PSR1212

Figure 34. Input/Output Parameters Worksheet

Refer to the back of this document for a complete set of PSR1212 worksheets.

7 Application Examples

Auditorium, Gymnasium, and Hotel/Convention Center

Auditorium Application Example
Refer to Figures 35, 36, and 37 for connection, routing, and parameter information for this application.

■ Four microphones are connected to inputs 1-4, and are routed to outputs 1-6, which are connected to the power amplifier. The mics are also routed to outputs 9 and 10, which are connected to the tape recorder. The mics are gated according to the preferences you choose, which might vary with different presets.

■ The tape recorder is connected in a playback/recording configuration using inputs 9 and 10 and outputs 9 and 10. Playback audio from the tape recorder is routed to the amp, and audio from the microphones is routed to the tape recorder.

■ The CD player is connected to inputs 11 and 12, and is routed to the outputs connected to the power amp and tape recorder.

■ Parametric equalizers and filters are configured for audio being routed to the power amp. This is to enhance sound quality through the various speaker elements of the front horn speaker cluster.

PSR1212 Auditorium Installation (basic example)

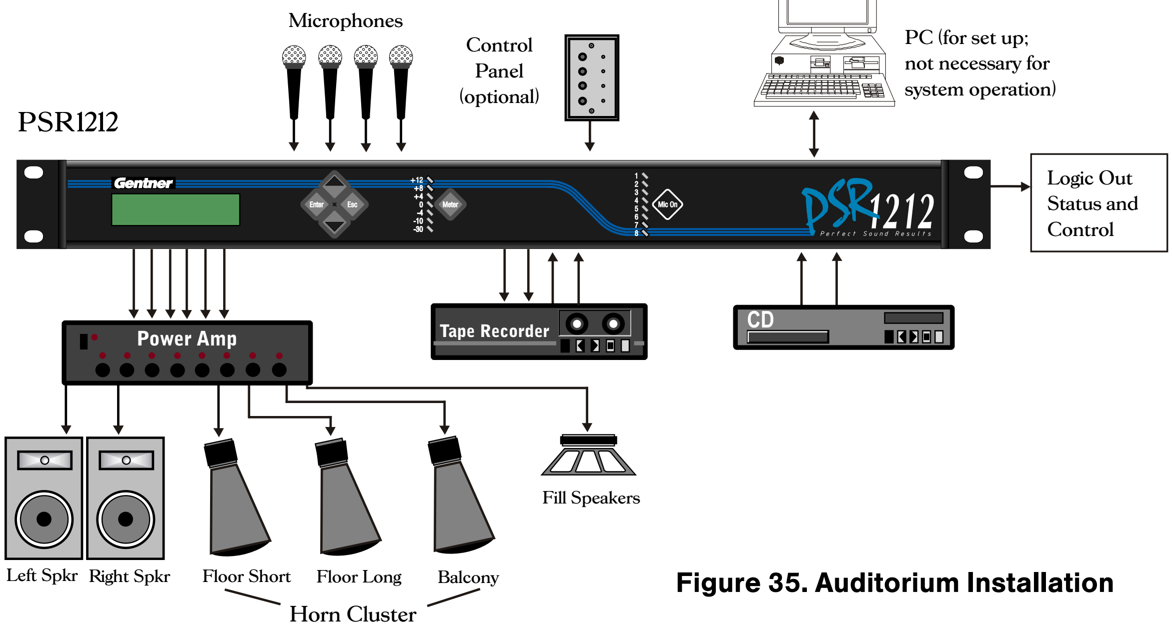

Figure 35. Auditorium Installation

PSR1212
Routing Matrix Diagram
Application Name: Auditorium

Notes:			1 To Left Program Speaker	2 To Right Program Speaker	3 To Short Floor Horn	4 To Long Floor Horn	5 To Balcony Horn	6 To Fill Speakers	7	8	9 To Tape Recorder Left	10 To Tape Recorder Right	11	12	O Inputs 1-8 to other PSRs	P Inputs 9-12 to other PSR's	Q	R	S	T	U	V	W	X	Y	Z	A Mix Audio for Fill Speakers	B	C	D	E	F	G	H
			Outputs 1 - 12												To Expansion Bus												To Processing							
Microphone 1	In 1 - 8	1	G	G	G	G	G				G	G															G							
Microphone 2		2	G	G	G	G	G				G	G															G							
Microphone 3		3	G	G	G	G	G				G	G															G							
Microphone 4		4	G	G	G	G	G				G	G															G							
		5																																
		6																																
		7																																
		8																																
Tape Recorder Left In	In 9 - 12	9	X		X	X	X				X																X							
Tape Recorder Left Out		10		X	X	X	X					X															X							
CD Left In		11	X		X	X	X				X																X							
CD Left In		12		X	X	X	X					X															X							
Inputs 1-8 from other PSRs	From Expansion Bus	O																																
Inputs 9-12 from other PSRs		P																																
		Q																																
		R																																
		S																																
		T																																
		U																																
		V																																
		W																																
		X																																
		Y																																
		Z																																
Delayed Mix Fill Audio	From Processing	A						X																										
		B																																
		C																																
		D																																
		E																																
		F																																
		G																																
		H																																

0 Gated with crosspoint attenuation 0 Non-Gated with crosspoint attenuation 0 Crosspoint with attenuation OR G = Gated N = Non-Gated X = Crosspoint

Figure 36. Auditorium Routing Matrix

PSR1212 System Parameters Worksheet (Auditorium)

Gentner PSR1212

System-Wide Parameters

Program Parameter	Selection Range		Program Parameter	Selection Range	
Timeout	0 - 15 (**5**)		Mixer Mode	**Master**, Slave	
Lock Front Panel	On, **Off**		RS232 Baud Rate	9.6, 19.2, **38.4**, 57.6, 115.2 kbps	
Set Passcode	Any 5 Front Panel Keys		RS232 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 Expansion Bus References

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

Hints:
Each Expansion Bus PA Adapt reference can only be defined once across the system (all linked PSR1212s).
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

Figure 37. Auditorium System Parameters

Gymnasium Application Example

Refer to Figures 38, 39, and 40 for connection, routing, and parameter information for this application.

■ Microphones are connected to inputs 1-4, and are routed to outputs 1 and 2 via processing channels A and B; and outputs 1 and 2, which are connected to the PA870 power amplifier. The mics are also routed to outputs 9 and 10 (tape recorder). The mics are gated according to the preferences you choose.

■ The tape recorder is connected in a playback/recording configuration using inputs 9 and 10 and outputs 9 and 10. Playback audio from the tape recorder is routed to the amp, and audio from the microphones is routed to the tape recorder.

■ The CD player is connected to inputs 11 and 12, and is routed to the outputs connected to the power amp and tape recorder.

■ Parametric equalizers and filters are configured for audio being routed to the PA870 power amp. This is to enhance sound quality through the various speaker elements of the hanging speaker cluster.

PSR1212 Gymnasium Installation (basic example)

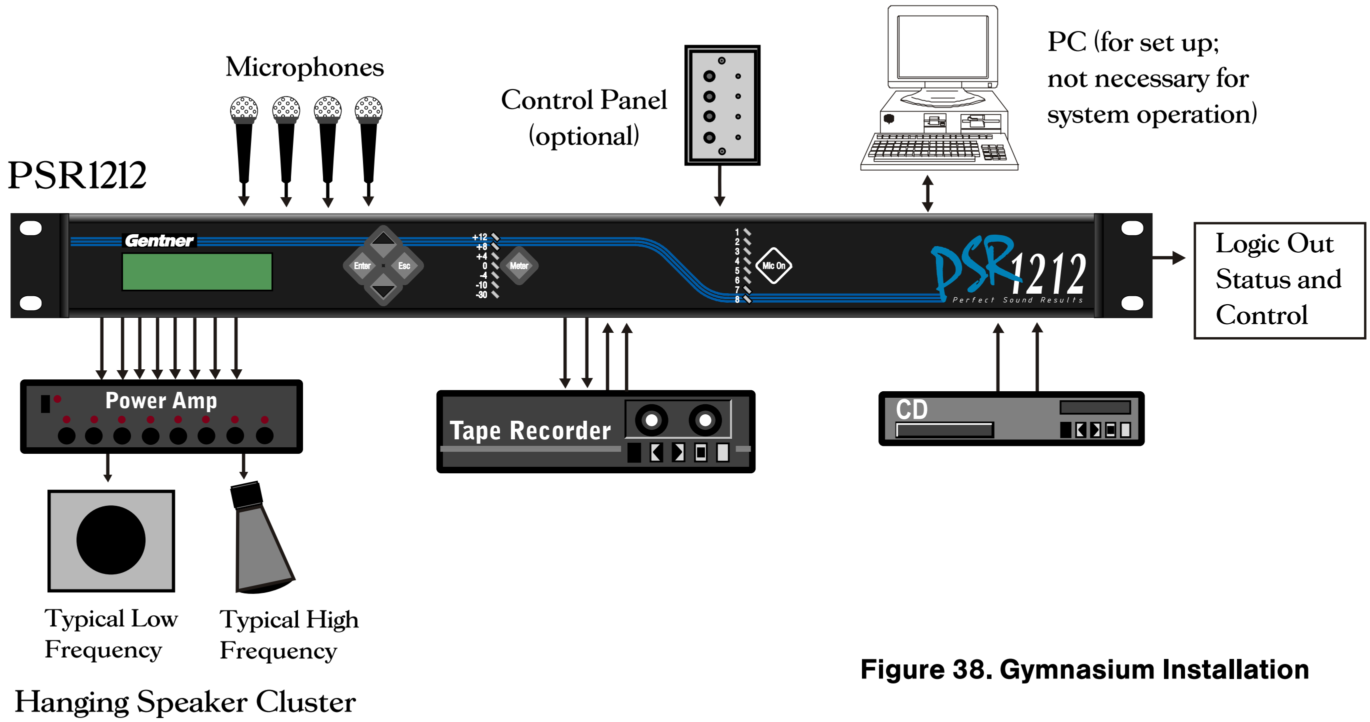

Figure 38. Gymnasium Installation

PSR1212
Routing Matrix Diagram
Application Name: Gymnasium

Input	Group	No.	1 To Speaker Cluster Horns	2 To Speaker Cluster Subwoofer	3	4	5	6	7	8	9 To Tape Recorder Left	10 To Tape Recorder Right	11	12	O Inputs 1-8 to other PSRs	P Inputs 9-12 to other PSRs	Q	R	S	T	U	V	W	X	Y	Z	A Mix Audio for Horns	B Mix Audio for Subwoofer	C Tape Recorder Left	D Tape Recorder Right	E CD Player Left	F CD Player Right	G	H
Notes:			Outputs 1 - 12												To Expansion Bus												To Processing							
Microphone 1	In 1 - 8	1									G	G															G	G						
Microphone 2	In 1 - 8	2									G	G															G	G						
Microphone 3	In 1 - 8	3									G	G															G	G						
Microphone 4	In 1 - 8	4									G	G															G	G						
	In 1 - 8	5																																
	In 1 - 8	6																																
	In 1 - 8	7																																
	In 1 - 8	8																																
Tape Recorder Left In	In 9 - 12	9									X																		X	X				
Tape Recorder Right In	In 9 - 12	10										X																	X	X				
CD Left In	In 9 - 12	11									X																				X	X		
CD Left In	In 9 - 12	12										X																			X	X		
Inputs 1-8 from other PSRs	From Expansion Bus	O																																
Inputs 9-12 from other PSRs	From Expansion Bus	P																																
	From Expansion Bus	Q																																
	From Expansion Bus	R																																
	From Expansion Bus	S																																
	From Expansion Bus	T																																
	From Expansion Bus	U																																
	From Expansion Bus	V																																
	From Expansion Bus	W																																
	From Expansion Bus	X																																
	From Expansion Bus	Y																																
	From Expansion Bus	Z																																
Audio Mix w/HPF	From Processing	A	X																															
Audio Mix w/LPF	From Processing	B		X																														
Tape Recorder Left	From Processing	C	X																															
Tape Recorder Right	From Processing	D		X																														
CD Player Left	From Processing	E	X																															
CD Player Right	From Processing	F		X																														
	From Processing	G																																
	From Processing	H																																

0 Gated with crosspoint attenuation   0 Non-Gated with crosspoint attenuation   0 Crosspoint with attenuation   OR   G = Gated   N = Non-Gated   X = Crosspoint

Figure 39. Gymnasium Routing Matrix

PSR1212 System Parameters Worksheet (Gymnasium)

Gentner PSR1212

System-Wide Parameters

Program Parameter	Selection Range		Program Parameter	Selection Range	
Timeout	0 - 15 (**5**)		Mixer Mode	**Master**, Slave	
Lock Front Panel	On, **Off**		RS232 Baud Rate	9.6, 19.2, **38.4**, 57.6, 115.2 kbps	
Set Passcode	Any 5 Front Panel Keys		RS232 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

		Internal / Unit Mixer Groups				Global / Site Mixer Groups			
Program Parameter	**Selection Range**	**1**	**2**	**3**	**4**	**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 Expansion Bus References

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

Hints:
Each Expansion Bus PA Adapt reference can only be defined once across the system (all linked PSR1212s).
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

Figure 40. Gymnasium System Parameters

Hotel/Convention Center Application Example

Refer to Figures 41, 42, and 43 for connection, routing, and parameter information for this application.

■ Eight microphones—two for each room—are connected to inputs 1-8. These inputs are routed to the PA870 power amp, which is connected to speakers from all four rooms. Various presets are established to route particular mics to particular speakers, depending on room configuration.

■ The VCR is connected in a playback/recording configuration using inputs 9 and 10 and outputs 9 and 10. Playback audio from the VCR is routed to the amp. Audio from the microphones is routed to the VCR.

■ The CD player is connected to inputs 11 and 12, and is routed to the outputs connected to the power amp and VCR.

■ Parametric equalizers and filters are configured for audio being routed to the PA870 power amp. This is to enhance sound quality, particularly from audio sourced from the CD player and VCR.

PSR1212 Hotel/Convention Center Installation (basic example)

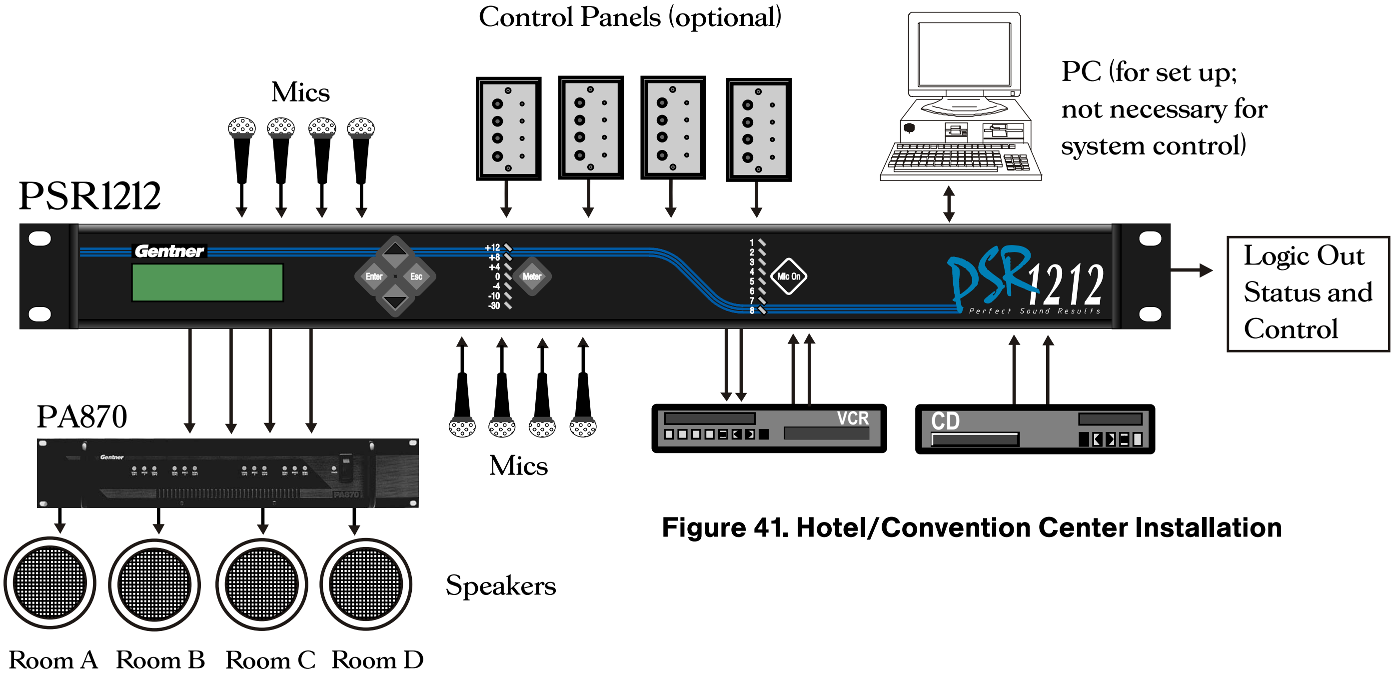

Figure 41. Hotel/Convention Center Installation

PSR1212
Routing Matrix Diagram
Application Name: Hotel

Notes:			To Room A	To Room B	To Room C	To Room D					To VCR Left	To VCR Right			Inputs 1-8 to other PSRs	Inputs 9-12 to other PSRs																		
			Outputs 1 - 12												To Expansion Bus												To Processing							
			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
Microphone 1 (Room A)	In 1 - 8	1	G								G	G																						
Microphone 2 (Room A)		2	G								G	G																						
Microphone 3 (Room B)		3		G							G	G																						
Microphone 4 (Room B)		4		G							G	G																						
Microphone 5 (Room C)		5			G						G	G																						
Microphone 6 (Room C)		6			G						G	G																						
Microphone 7 (Room D)		7				G					G	G																						
Microphone 8 (Room D)		8				G					G	G																						
VCR Left In	In 9 - 12	9									X																							
VCR Right In		10										X																						
CD Left In		11									X																							
CD Right In		12										X																						
Inputs 1-8 from other PSRs	From Expansion Bus	O																																
Inputs 9-12 from other PSRs		P																																
		Q																																
		R																																
		S																																
		T																																
		U																																
		V																																
		W																																
		X																																
		Y																																
		Z																																
	From Processing	A																																
		B																																
		C																																
		D																																
		E																																
		F																																
		G																																
		H																																

0 Gated with crosspoint attenuation 0 Non-Gated with crosspoint attenuation 0 Crosspoint with attenuation OR G = Gated N = Non-Gated X = Crosspoint

Figure 42. Hotel/Convention Center Routing Matrix

PSR1212 System Parameters Worksheet (Hotel)

System-Wide Parameters

Program Parameter	Selection Range		Program Parameter	Selection Range	
Timeout	0 - 15 (**5**)		Mixer Mode	**Master**, Slave	
Lock Front Panel	On, **Off**		RS232 Baud Rate	9.6, 19.2, **38.4**, 57.6, 115.2 kbps	
Set Passcode	Any 5 Front Panel Keys		RS232 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 Expansion Bus References

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

Hints:
Each Expansion Bus PA Adapt reference can only be defined once across the system (all linked PSR1212s).
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

Gentner PSR1212

Figure 43. Hotel/Convention Center System Parameters

8 Remote Control and Status

Control and Status Configuration

To allow many different control configurations, the PSR1212 can be controlled serially via the RS-232 port; directly through the two control/status A or B labeled connectors on the rear panel; or through a combination of the above. Also, a few functions can be controlled from the front panel. This section discusses direct remote control.

There are two functions available: control and status (see Figure 44). 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 at a maximum of 40mA of current through an open collector @20VDC.

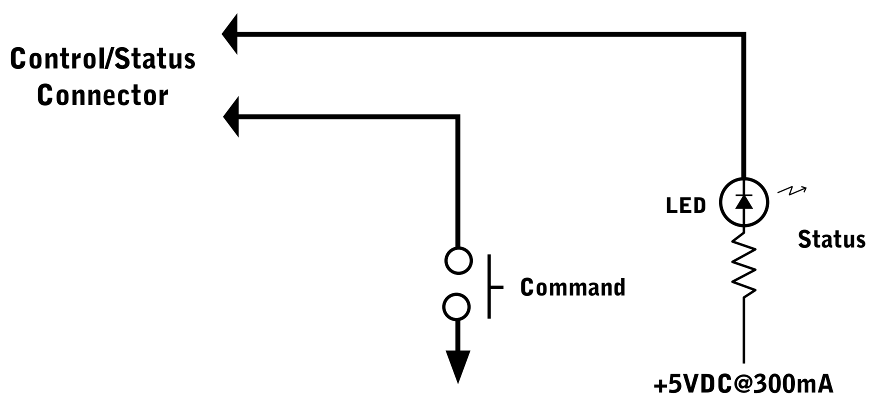

Figure 44. Direct Control/Status Operation

Any valid function of all PSR1212s connected to the Expansion Bus is capable of being controlled from the control/status pins of any connected PSR1212. These functions include volume up, volume down, mute, etc. In addition, pins can be programmed to call up to 32 preprogrammed presets (each preset representing 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 Center Application Example in Chapter 7: Application Examples.

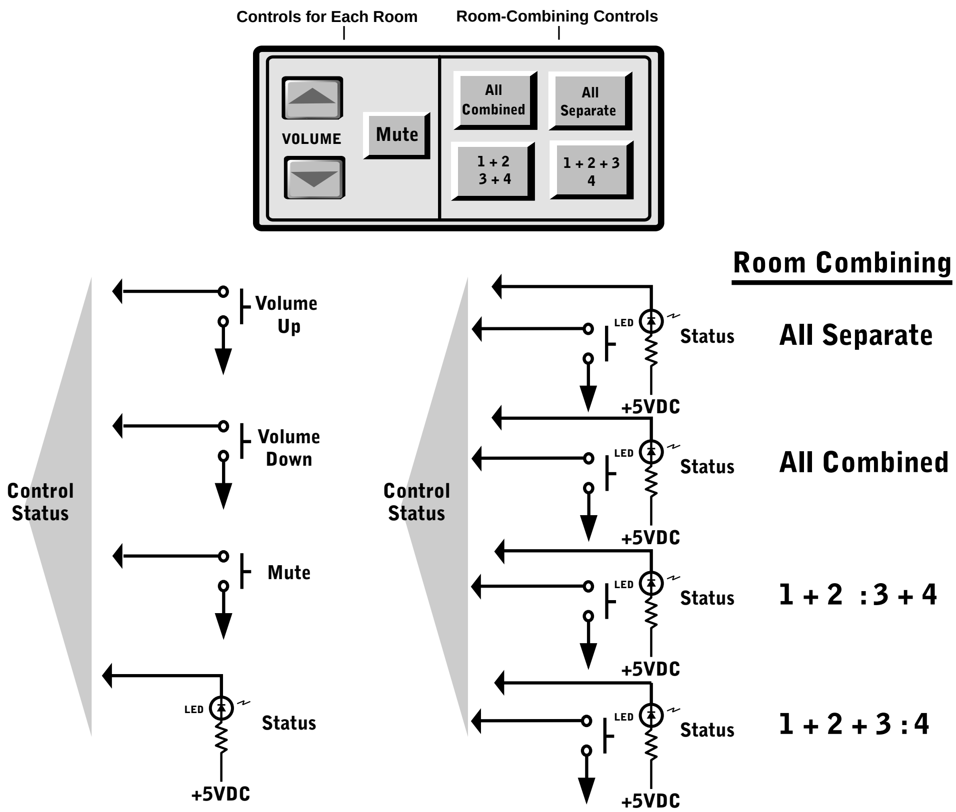

Figure 45. Room Combining Using Control/Status Pins

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 (M) 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 9 output toggle
6	Yes	S		Status of 9 output mute
7	Yes	C	M	Mute 10 output toggle
8	Yes	S		Status of 10 output mute
9	Yes	C	M	Mute 11 output toggle
10	Yes	S		Status of 11 output mute
11	Yes	C	M	Mute 12 output toggle
12	Yes	S		Status of 12 output mute
13	Yes	C	M	Not programmed
14	Yes	S		Volume up D output 1 (1dB)
15	Yes	C	M	Not programmed
16	Yes	S		Volume down D output 1 (1dB)
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 (M) OR LATCHING (L)	DEFAULT DESCRIPTION
1	No	C	L	Preset select bit 0
2	No	S		Status bit 0
3	No	C	L	Preset select bit 1
4	No	S		Status bit 1
5	No	C	L	Preset select bit 2
6	No	S		Status bit 2
7	No	C	L	Preset select bit 3
8	No	S		Status bit 3
9	No	C	L	Preset select bit 4
10	No	S		Status bit 4
11	No	C	L	Preset select bit 5
12	No	S		Status bit 5
13	No	C	L	Preset select bit 6
14	No	S		Status bit 6
15	No	C	L	Preset select bit 7
16	No	S		Status bit 7
17	No	C		Preset select bit 8
18	No	S		Status bit 8
19	No	C		Preset select bit 9
20	No	S		Status bit 9
21	No connection			
22	No connection			
23	No	-		+5VDC 100mA
24	No	-		+5VDC 100mA
25	No	-		Ground

Often, a combination of serial and direct control might be used in an installation. For example, in the hotel/convention center application example, a custom remote controller touch screen might help with controlling the bulk of the equipment in the room. However, functions such as volume, mute and room combining configuration may be controlled directly from the unit.

Below is an example of how to use the control and status pins (example can be workbench tested):
1. Design the entire audio chain and document.

2. Determine which pins on the control/status connector will be used for control and status (refer to Control Status Connector tables).

3. If using a custom remote controller, program these units to communicate via RS-232 ports with the PSR1212. Connect to the PSR1212 and verify proper operation.

4. With G-Ware software, configure the desired number of presets. Routing, AGC, equalization, levels, etc. are programmed and saved as a preset. Enable presets from the custom remote controller, front panel, control/status connector, or combination of the above.

5. Define the pin number on the control/status connector as the preset using the software. Also, program any status outputs.

6. Connect the user panel(s) and verify direct control/status is operational.

7. Now install the audio system and complete the audio calibration procedure through the PC software. Test tones, test noise, and all gains and actual levels in dBu are provided.

Macro Pro Control Software
Macro Pro software, included with the PSR1212, allows you to customize macro commands for a unit or networked system.

A macro is a command consisting of a list of other commands. When a macro is sent to the PSR1212, it executes the macro, or list of commands. Upon completion of each macro, the PSR1212 issues a macro-completed response which is used to keep all connected system devices synchronized with the state of the system.

A macro can contain another macro as another command. This allows you to create small macros to perform specific tasks, then combine them to perform larger tasks.

G-Ware software is used for programming a macro. A macro is built by adding commands from the list of available commands to the macro execution list. When the macro is assembled, it is downloaded to the unit and is available for execution.

A macro can also contain a network address in the command. This allows you to issue commands to Expansion Bused PSR1212 units. This can be done to issue single commands, change a preset, or execute a macro in the other box.

9 Serial Remote Control

RS-232 Port

Operation of linked PSR1212 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 PSR1212, the system was designed primarily to be programmed and set up using a personal computer, and operated using a custom remote controller.

The PSR1212 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

The RS-232 serial port was generally intended to be used as follows:
1. Set up and programming - During installation, a PC is connected to the PSR1212, and the unit is programmed for the application gain structure, routing, equalization, etc.

2. Calibration - Using the connected PC, precise input and output levels can be directly read from the PC. Additionally, the PSR1212 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 be -6dBu. If you apply 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 PSR1212.

10 Programming

PC Programming

One of the most important features of the PSR1212 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, G-Ware software makes all functions available to expanded units, making a linked PSR1212 system capable of automixing 64 microphones and 32 line inputs.

G-Ware takes advantage of a DSP infrastructure in accomplishing this task. It uses the Expansion Bus, a high-speed network protocol, to allow up to eight PSR1212 units to be networked together. The Expansion Bus provides two primary system functions: 1) communications among units, and 2) audio linking.

Front Panel

Using G-Ware software (described on the previous page) on a connected PC is the only method for programming all the features of the PSR1212, but a few parameters can also be adjusted through the front panel buttons and displays. The front panel consists of an LCD display, five parameter adjustment buttons, an LED VU meter, and eight mic LEDs.

To protect from unauthorized changes, the PSR1212 is pass-coded. Navigation of the menus is allowed without a password; however, changes to programming require a valid password.

The five parameter adjustment keys consist of ▲, ▼, Enter, Esc (Escape), and Meter buttons. The first four keys (▲, ▼, Enter, and Esc) are used for menu navigation and parameter selection. The Meter button determines what audio is selected on the front panel LED VU meter. In addition, numeric audio levels are displayed on the LCD panel.

Menu Structure

System - This menu is used to access presets 1-32 and set security parameters and the master/slave configuration. Also, the System menu is used for viewing the device ID number, unit ID number, and software version information.

RS-232 - This menu is used to configure baud rate and flow control through the RS-232 port. It is also used to enable the modem and clear passwords.

Meter - The Meter menu lets you select which audio source is displayed on the LED meter, including any input or output. You can set the meter to display speaker-to-mic room loss for any input 1-8. Also, you can set it to display a default input, output, or room loss level.

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

Gentner recommends that you use G-Ware PC software (described on the previous page) to program these parameters of the PSR1212.

11 PSR1212 Connections

System Connections

Audio Connections

The PSR1212 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 PSR1212.

Expansion Bus Connection

The Expansion Bus consists of two RJ45 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.

Pass Codes

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 PSR1212 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.

Power

A universal power connector is provided. The PSR1212 will operate on all global voltages and cycles.

Expansion Bus Connections

Communication Functions of the Expansion Bus

The Expansion Bus is a high-speed network protocol that provides two primary system functions: 1) communications among units, and 2) audio linking. All functions of the PSR1212 are available across a system of linked PSR1212 units, which allows automixing of up to eight PSR1212 units, 64 microphones, and 32 line inputs.

The PSR1212 takes advantage of its DSP infrastructure in accomplishing this task. Networked PSR1212 units communicate to one another via the Expansion Bus (see Figure 46). Control, status and addressing functions are performed via the network bus. To accomplish this, configure the first PSR1212 as the MASTER unit. 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 PSR1212 permits programming, operation and diagnostics to all PSR1212 units networked together. This permits a single connection for the installer and user, decreasing costs and complexity.

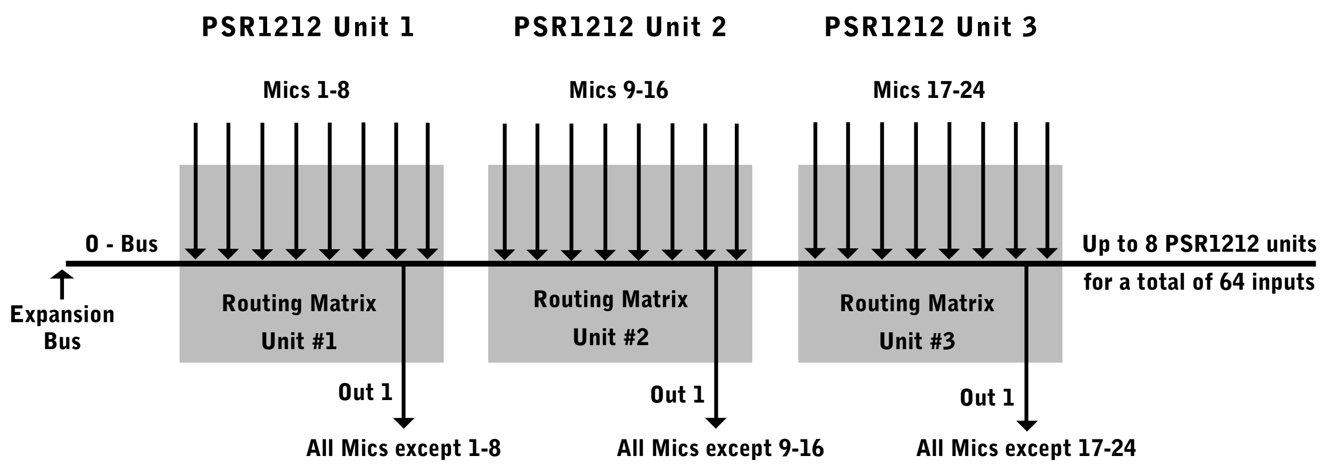

Figure 46. Mix-Minus Configuration of the O Bus

Expansion Bus Audio Functions
The Expansion Bus network architecture allows up to eight PSR1212s and up to 96 inputs, 96 outputs, and 64 microphones 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 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 PSR1212s. 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 or VCR, 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 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 PSR1212s. 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 PSR1212 comes standard with one Expansion Bus cable. The maximum distance allowed between any two PSR1212 units on an Expansion Bus network is 80 feet. Gentner Communications recommends that category five twisted-pair (10BaseT LAN) cable be used.

Appendices

Appendix A: Specifications

G-Ware (setup/diagnostics software)

Dimensions (LxDxH):
17¼″ x 10¼″ x 2¾″ (43.8 x 26.0 x 7.0cm)

Weight:
9.9 lb (4.5 kg) dry
13 lb (5.9 kg) shipping

Power Requirements:
Auto-adjusting power module
Fuse: 2 amp, 250VAC, slow blow type

INPUT RANGE:
100-240VAC; 50/60Hz

POWER CONSUMPTION:
30W, typical

95 BTU/Hr

PHANTOM POWER:
24V, input selectable

Audio Performance
FREQUENCY RESPONSE: 20Hz to 20kHz ± 1dB

NOISE: (EIN 20Hz to 20kHz) -125dB

THD: <0.02%

Dynamic Range: >100dB

Automatic Mic/Line Inputs:
CONNECTOR: Removable terminal block; -55, -25, or 0dBu nominal adjustable, balanced, bridging
Impedance: 7kOhms

Non-Automatic Line Inputs:
CONNECTOR: Removable terminal block; 0dBu nominal adjustable,
balanced, bridging
Impedance: >20kOhms

Line Outputs:
CONNECTOR: Removable terminal block; 0dBm nominal level adjustable,
balanced
Impedance: 50 Ohms

Other Connectors:
RS-232: DB9 female (DCE) 9,600/19,200/38,400/57,600 baud

Control/Status A: DB25 female

Inputs A: Active low (pull to ground)

Outputs A: Open collector, 20V max, 40mA each

Control/Status B: DB25 female

Inputs B: Active low (pull to ground)

Outputs B: Open collector,
20V max, 40mA each (2)

Operating Temperature:
32 - 100°F / 0 - 38°C

Approvals: FCC Part 15, CSA NRTL/C, CE

Audio Functions

Filters: All pass; low pass; high pass; low shelving; high shelving; PEQ; CD horn EQ; crossover; and Bessel, Butterworth, and Linkwitz-Riley crossovers; compressors; matrix mixer w/cross point level control; automatic gain control; and automatic microphone mixer.

Signal Delay: Adjustable up to 500ms. G-Ware calculates distances.

Appendix B: Glossary

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 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 18 dB) vs frequency (20Hz to 20kHz) on a logarithmic scale.

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 PSR1212.

Bandwidth The amount of spectrum space a signal occupies. For the PSR1212, it is adjustable from .05 to 5 octaves. Changing the bandwidth setting affects the Q setting.

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.

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 PSR1212's 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 Time The amount of time designated for a microphone to go from the On attenuation level to the Off level.

Digital Signal Processing (DSP) A method of signal enhancement which increases frequency response and dynamic range while reducing noise.

Expansion Bus Consists of two RJ45 connectors on the rear panel of the PSR1212. An Expansion Bus allows multiple PSR1212s to be networked together using category five twisted-pair (10BaseT LAN) cable.

Filter A device that passes audio signals at certain frequencies while attenuating signals at other frequencies.

Filter, 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 speaker driver deficiencies.

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

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

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.

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.

First Mic Priority A condition in which a particular microphone is assigned gating priority over other microphones in a PSR1212 system.

Gain The amount a signal is increased over a given reference, typically 0. Normally specified in dB (decibels). On the PSR1212, gain is adjustable from -18 to 18dB in 1dB increments.

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

Gating Ratio The voice (input) level that must be reached before a microphone will gate on.

GPIO (general purpose input/output) The Control/Status Port B on the rear of the PSR1212 unit.

G-Ware Software The PSR1212's setup and configuration software.

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

Last Mic Mode Enables one of three microphone activation values: Last On, Mic 1, and Off.

Last On Mode Leaves the last-activated mic gated on until another microphone input gates on.

Macro A computer command consisting of a sequence of other commands.

Macro Pro Software Software that allows you to customize and execute macro commands for a PSR1212 unit or network.

Manual Gating Provides 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 PSR1212, 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 scenario where microphone inputs are mixed according to parameters such as gain levels and gating priority.

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

Number of Open Mics (NOM) The number of microphones gated on at a given time.

Number of Open Mics/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 PSR1212 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.

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

Parametric Equalizer (PEQ) A multi-band variable equalizer with control of gain, center frequency, and bandwidth. A properly configured PEQ enables the PSR1212 to offset speaker or room acoustic deficiencies.

Phantom Power Power supplied by the host unit to power an auxiliary device, such as certain types of microphones. The PSR1212 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. This plot overlays the amplitude plot, and is generated only for the active filter display.

Pink Noise Acoustical noise whose amplitude is inversely proportional to the frequency within a defined frequency range.

Preset One of 32 configurable memories in the PSR1212. A preset can be programmed with a variety of routing, level, gating, delay, filter, and equalizer settings to meet specific application requirements.

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 PSR1212.

Reverberation Multiple reflections of sound waves in a room.

Signal Delay Used for introducing a delay to fill speakers in an audio system to provide balanced sound throughout the room.

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

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

White Noise Acoustical noise distributed evenly throughout a given frequency range.

PSR1212 System Parameters Worksheet

Gentner PSR1212

System-Wide Parameters

Program Parameter	Selection Range		Program Parameter	Selection Range	
Timeout	0 - 15 (**5**)		Mixer Mode	**Master**, Slave	
Lock Front Panel	On, **Off**		RS232 Baud Rate	9.6, 19.2, **38.4**, 57.6, 115.2 kbps	
Set Passcode	Any 5 Front Panel Keys		RS232 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 Expansion Bus References

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

Hints:
Each Expansion Bus PA Adapt reference can only be defined once across the system (all linked PSR1212s).
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

PSR1212 System Parameters Worksheet

PSR1212 Input/Output Parameters Worksheet

Gentner PSR1212

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, 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.			

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 (**0**)												
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/Limiter	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)**								

PSR1212 Input/Output Parameters Worksheet

PSR1212 Processing Filter Parameters Worksheet #1

Gentner PSR1212

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)

PSR1212 Processing Filter Parameters Worksheet #1

PSR1212 Processing Filter Parameters Worksheet #2

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)*																

Gentner PSR1212

PSR1212 Processing Filter Parameters Worksheet #2

PSR1212 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.						
DB-25 Remote Control Port "B" Setup						
DB-25 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

PSR1212 Macro Worksheet

Macro Setup

Macro #/Name						
Description:						
Command List:						
Macro #/Name						
Description:						
Command List:						
Macro #/Name						
Description:						
Command List:						
Macro #/Name						
Description:						
Command List:						
Macro #/Name						
Description:						
Command List:						
For extra large macros, continue commands in next column.						

PSR1212 Macro Worksheet