PSR1212

White Paper

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PSR1212 White Paper
ClearOne Part No. 800-155-002
June 2002 (Rev. 2.0)

PSR1212 White Paper

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CHAPTER 1: Introduction

Overview

Audio is critical to human communication. Media such as voice mail, the Internet, conference calling, video conferencing, and electronic presentations are driving the demand for better audio-communication technologies. The 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 12x12 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 want, while automatic gain control keeps the overall sound level consistent. Input channels 1–8 can be configured as an automatic microphone mixer.

Audio routing is simplified with G-Ware's Matrix Screen. 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.

Features and benefits

- 12x12 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.
- Gain, audio processing, muting, and automatic mic mixing can be configured for each mic input channel.
- 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.
- Four internal and four global (across E-bus) gating groups.
- Internal room combining capabilities.
- Logic outputs.
- Configurable microphone gating.
- Up to eight PSR1212s can be connected and controlled as a single unit, allowing 96 inputs and 96 outputs.
- Remote and local PC set up and diagnostics.
- One-year limited warranty.

PSR1212 enhancements

The current G-Ware release provides several enhancements to the PSR1212 that are designed to help with system configuration.

- **Virtual references.** Four user-definable virtual references can use multiple signals as the PA adapt reference without sacrificing an analog output.
- **Front panel gain and mute control.** Gain and mute adjustments can be made from the front panel without connecting through G-Ware.
- **Safety Mute.** Mutes all outputs with one simple click if feedback or audio problems occur during system configuration.
- **Preset and macro passwords.** Presets and macros can be password protected to prevent unauthorized users from making changes.
- **Clear Matrix.** This new button on the Matrix Screen makes it easy to clear all cross point cells in the matrix.
- **Ramp serial command.** The new Ramp serial command gives you greater control over gain adjustments. You can specify the rate at which the gain increases and decreases, as well as the target level. Multiple outputs may be controlled simultaneously by writing a macro containing multiple ramp commands.
- **800x600 resolution.** A scroll bar has been added to the Inputs configuration window so you can view Input 8 when using 800x600 resolution. 1024x768 is still recommended.
- **Signal Generator indicator.** This toolbar indicator illuminates when the signal generator is active.
- **Preset/Programming Output Mute.** When syncing to the unit, all outputs will be muted to prevent extraneous noises or popping sounds. When changing a preset, the outputs associated with a particular preset will be muted again to prevent unwanted noise.
- **The new XAP IR Remote Control.** Upgraded systems now support the XAP IR Remote Control.

Controls and Connections

Front panel

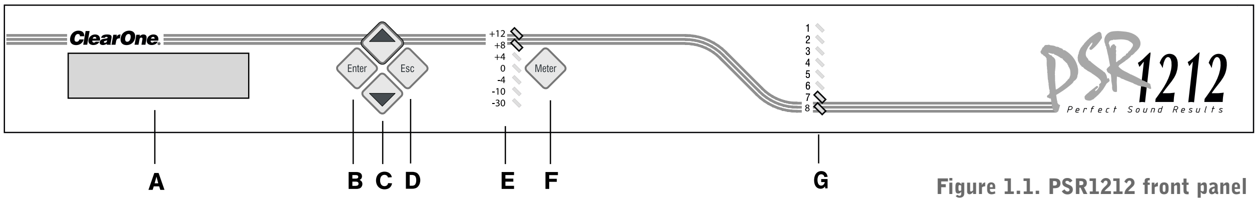

Figure 1.1. PSR1212 front panel

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

B. **Enter/▲▼/ESC.** Used to navigate the PSR1212's menu system.

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

D. **Meter.** Takes you directly to the Meter branch of the PSR1212's LCD menu tree.

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

Rear panel

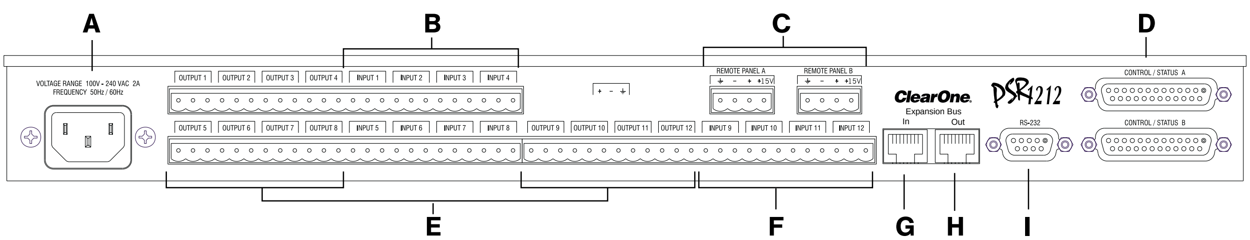

Figure 1.2. PSR1212 rear panel connections

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

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

C. **RS-485 Remote Control Ports.** These four-pin Phoenix ports allow you to control the PSR1212 with a ClearOne Control Panel or the XAP IR Remote Control.

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

E. **Inputs 9–12.** For line level inputs.

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

G. **RS-232.** This DB-9 serial port is to connect the PSR1212 to a PC, modem, or other custom remote controller.

H. **Control/Status Ports A and B.** These DB-25 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.

Expansion Bus Connections

The expansion bus (E-bus) network architecture allows up to eight PSR1212s to be controlled as if part of a single unit and makes it possible to route audio 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. The E-bus also contains four global gating buses and one control bus.

- **O–R buses.** These four audio buses are defaulted as the mic mix buses; they can communicate the NOM count 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 14 for more information about PA Adaptive mode.
- **Global Gating Groups A–D.** These mix-minus buses 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.
- **Control bus.** The control bus is an independent channel from the E-bus's audio channel; it uses a different pair of wires on the same E-bus cable. This design allows control information to pass even if the units are not using the audio link.

Network requirements

The maximum distance allowed between any two PSR1212 units on an expansion bus network is 80 feet (24 meters). ClearOne recommends category five twisted-pair (10BaseT) cable be used to connect units.

Equipment placement

The PSR1212 is designed for a 19" equipment rack. With a desktop kit, it can be modified for tabletop placement. The PSR1212 can safely operate in temperatures between 32°–110° F/0°–38° C. Do not block any of the ventilation holes.

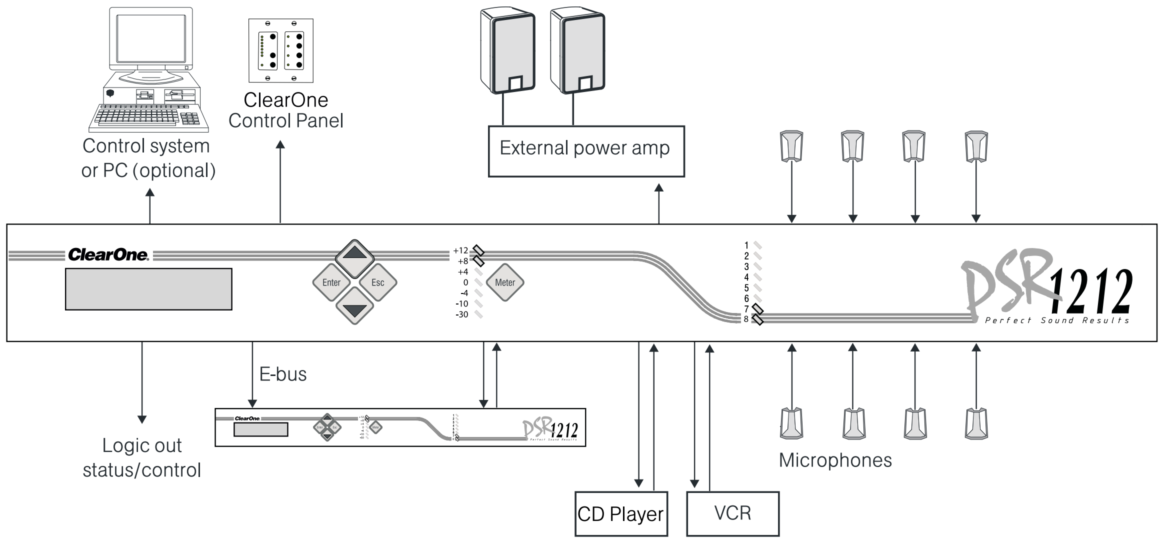

Figure 1.3. Typical PSR1212 installation (simplified for illustrative purposes)

PSR1212 block diagram

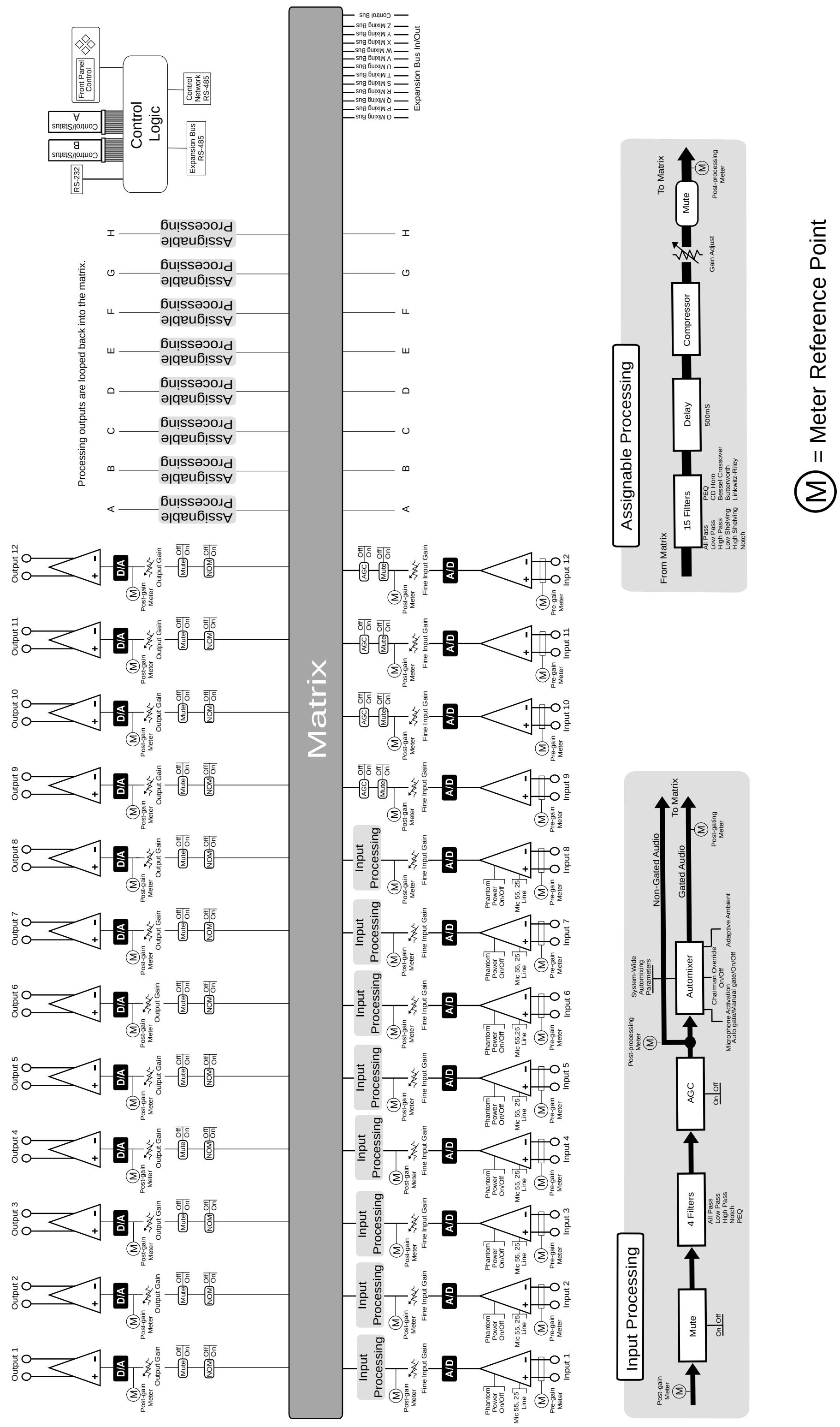

Figure 1.4. PSR1212 Block Diagram

CHAPTER 2: Inputs and Outputs

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 4kΩ of terminating impedance while line level inputs 9–12 provide >20kΩ of termination. Outputs provide a source impedance of 50Ω. All levels are referenced to a 0dBu level.

Input and output level control is executed in the digital domain. As a result, input levels should never exceed +20dBu. The unit will deliver a maximum output level of +20dBm. 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 and a pass band from 20Hz to 20kHz. 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.

Mic Input Audio Path

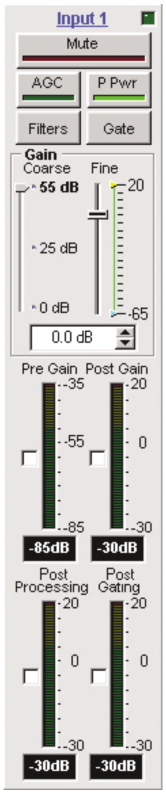

Figure 2.1. Mic input configuration in G-Ware

Mic input settings are configured using G-Ware. Some settings (gain adjustments for example) can be changed using front panel controls, serial commands, or parallel control through the Control/Status port.

The audio signal path for mic inputs is illustrated in the diagram on the following page. Balanced audio is input through the rear panel Phoenix connections. Mic/line level selection occurs first and phantom power is applied (if activated). The PSR1212 then converts the audio from analog to digital for processing by the DSP engine. Once converted to digital, audio is level controlled and passes through the channel mute function.

Audio next passes through filters. The PSR1212 has four configurable filters that can each be set individually as an all-pass filter, a low-pass filter, a high-pass filter, or a parametric equalizer (EQ). Each may be activated to equalize different types mics 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.

After the filters, automatic gain control (AGC) is applied. 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 mic 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.

At this point, non-gated audio is sent to the routing matrix for outputs that need non-gated audio. The final stage (automixing) determines how the audio is directed through the post-gating input to the routing matrix. A brief description of the automixing settings follows on the next page. For more detailed information, see Automixing with the PSR1212 on page 12.

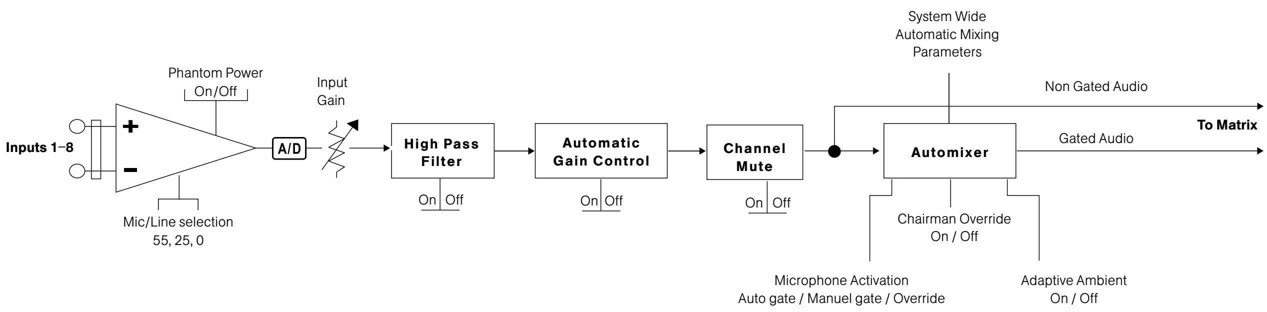

Figure 2.2. Inputs 1–8 signal path

Automixing

Each input can be configured with 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 selected gating group 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).** Provides gating priority for chairman override enabled microphones within the same gating group. When a mic with chairman override gates on, all mics that don't have chairman override enabled and are in the same gating group will gate off.
- **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.

Line Inputs

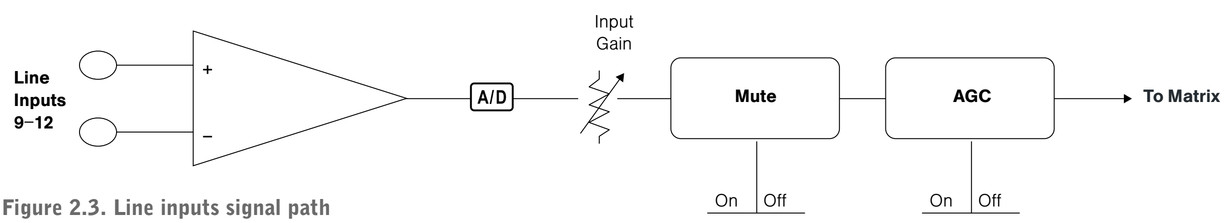

Figure 2.3. Line inputs signal path

The audio signal path for line inputs is shown above. The Gain, Mute, and AGC functions operate identically to the mic/line inputs. Line input settings are configured in G-Ware.

Line Outputs

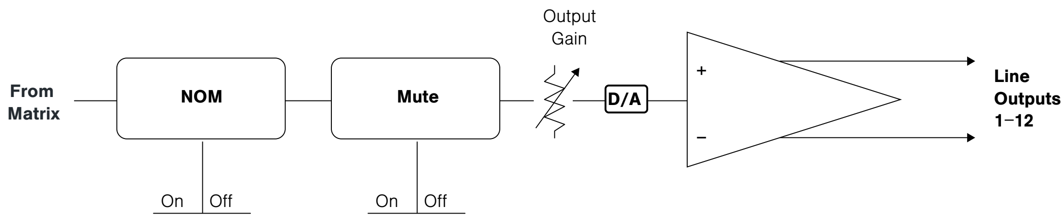

Figure 2.4. Outputs 1–12 signal path

Each of the 12 line outputs is identical. Three functions are associated with each output: number of open mics (NOM), mute, and gain control.

Unlike most automixers that have a single master NOM output, the PSR1212 provides a NOM setting at every output. 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 and is most useful in sound reinforced applications.

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. For example, 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.

CHAPTER 3: 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.

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.

Figure 3.1 illustrates one of the causes of unintelligible audio. In this diagram, direct voice audio being 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.

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

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

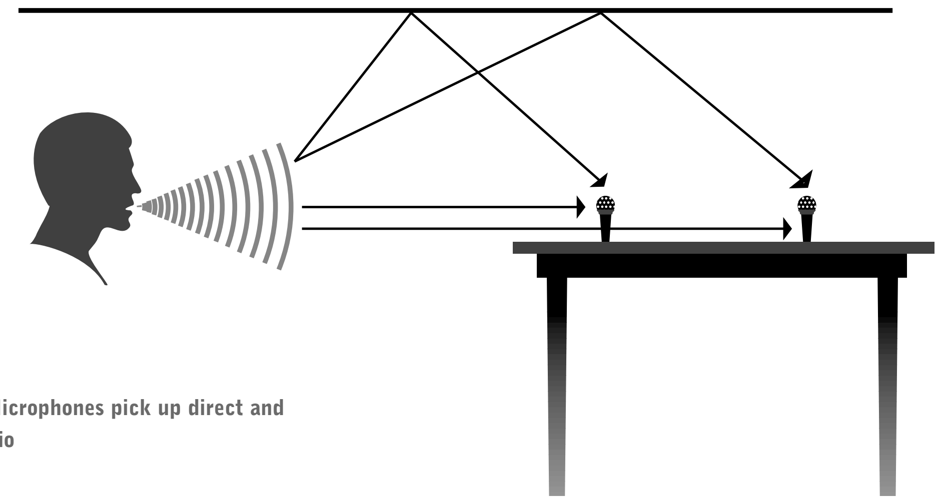

Figure 3.1. Microphones pick up direct and reflected audio

Automixing with the PSR1212

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 loudspeaker 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 loudspeakers 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 3.2) because it raises the mic gate ratio based on the audio level from the loudspeaker.

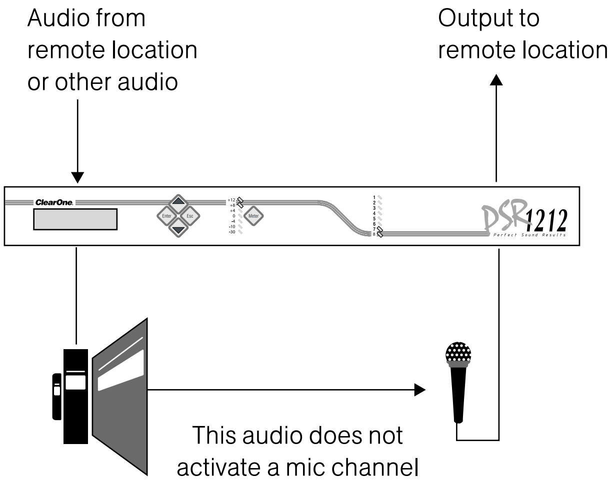

Figure 3.2. Microphone activation

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 page 13. 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 3.3.

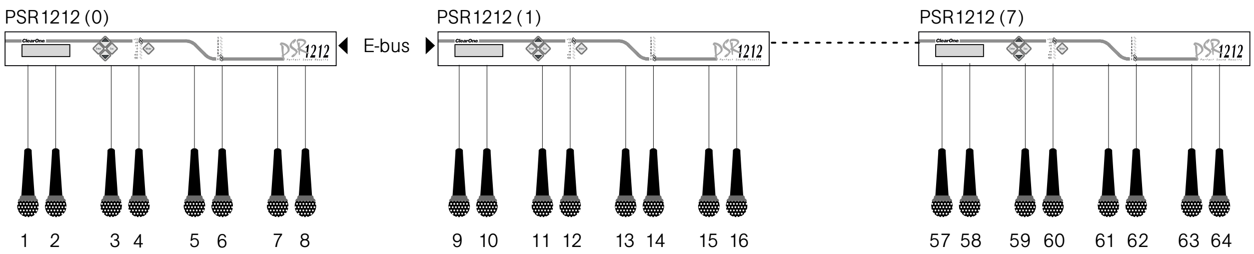

Figure 3.3. Expansion Bus Control of 64 Mics

Automixing parameters

Automixing or gating parameters are configured using G-Ware and provide high precision, reliable microphone mixing.

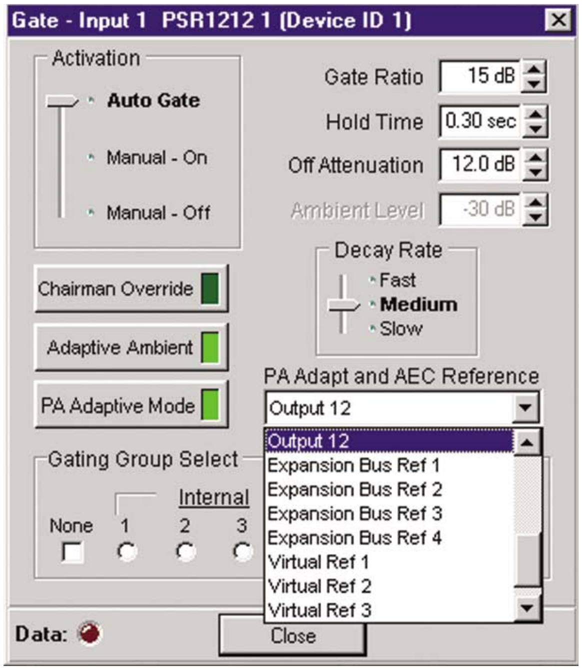

Figure 3.4. Gating parameters window in G-Ware

Activation

There are three mic activation settings: Auto Gate, Manual On, and Manual Off.

- **Auto Gate** determines mic gating based on the input level and gating settings for the gating group the input is assigned to. It contributes to and is affected by all gating group settings such as NOM, chairman override, etc.
- **Manual On** activates a mic, provided it does not exceed max NOM requirements of the gating group that the input is assigned to. It is included in the NOM count.
- **Manual Off** deactivates a mic.

Chairman Override

Chairman Override provides gating priority for this mic input over any other mic input within the same gating control (mixer) groups. When a mic with Chairman Override enabled gates on, all mics which don't have Chairman Override enabled will gate off. Default is off.

Adaptive Ambient

Adaptive Ambient adjusts the ambient reference level as noise and room conditions change. When adaptive ambient is on, the mic channel monitors the ambient noise level on the input and adjusts the ambient level reference automatically. This means that the gate threshold level automatically increases or decreases based on back-ground noise. If Adaptive Ambient is turned off, the input will use the fixed ambient level specified in the Ambient Level box as its gating reference. Default is on.

PA Adaptive Mode

PA Adaptive Mode uses loudspeaker audio level on a specified output as the new ambient level when audio is present at the power amplifier. This prevents loudspeaker audio from gating on the mic, while still allowing people in the room to gate on microphones as they speak—provided that their voices are louder than the loudspeaker audio. For example, you might decide to play background music from a CD player during a presentation. PA Adapt Mode allows you to use the output routed from the CD player as the ambient reference to prevent the CD player's audio from gating on microphones. An output must be specified as the PA Adaptive Reference for each mic in the system. Default is on.

Gate Ratio

Gate Ratio specifies how much louder the audio level must be above the ambient level before the channel gates on. The gate ratio range is from 0 to 50dB. Default is 15dB.

Hold Time

Hold Time determines how long the channel stays gated on after the audio is below the threshold. The hold time range is from .1 to 8.0 seconds. Default is .3 seconds.

Off Attenuation

Off Attenuation sets the amount of level reduction applied to a channel when it is gated off. The range is from 0 to 50dB. Default is 12dB.

Ambient Level

Ambient Level is available only if the Adaptive Ambient feature isn't enabled. Use Ambient Level if you want to specify a fixed reference point rather than one that adjusts for background noise. The range is from -80 to 0dB. Default is Off.

Decay Rate

Decay Rate determines how fast a channel gates off after the hold time expires. Default is Medium.

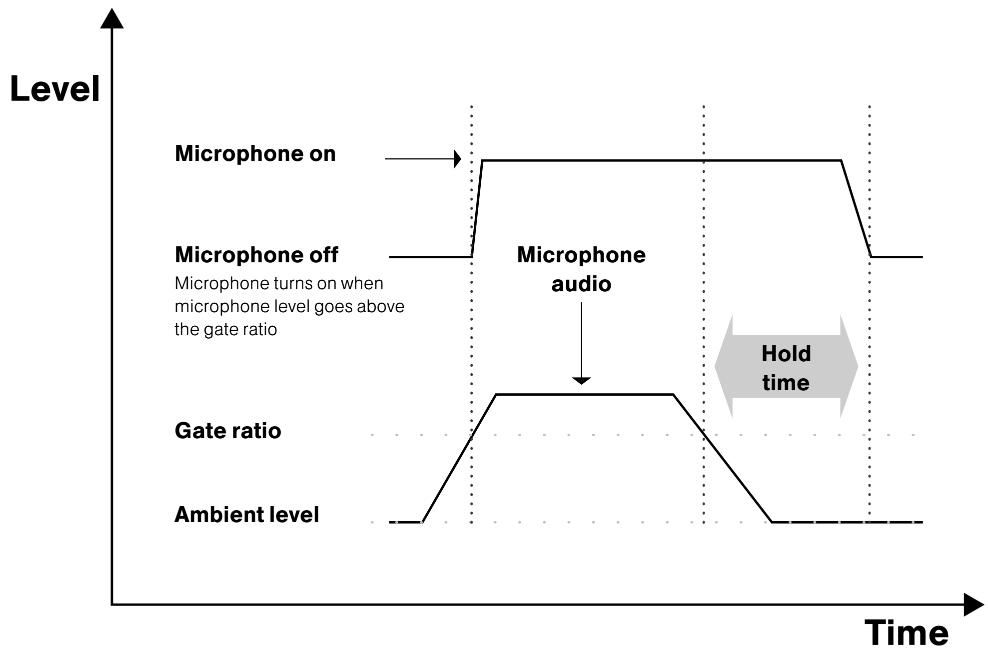

Figure 3.5. PSR1212 automixing gate functions

Mixer mode

There are two mixer mode settings: slave (default) and master. A master unit is not required in a networked system and in most installations, all units will be slaves. Master units ignore audio from upstream units. This prevents audio from being received from units above the master unit in the network. However, global control of the system is still maintained by whichever PSR1212 unit is connected to a control device through its RS-232 or RS-485 ports. Control is not affected by master/slave designations.

In Figure 3.6, the third unit in the network is a master. It prevents the audio from the first and second units from being passed down the network chain. Likewise the second master unit in the network will not pass on the audio from the unit before it. The third unit provides system-wide control through a connection to its RS-232 port.

Mixer mode can be selected in the Unit Properties window or from the front panel programming menu. Consult the PSR1212 user manual for more information.

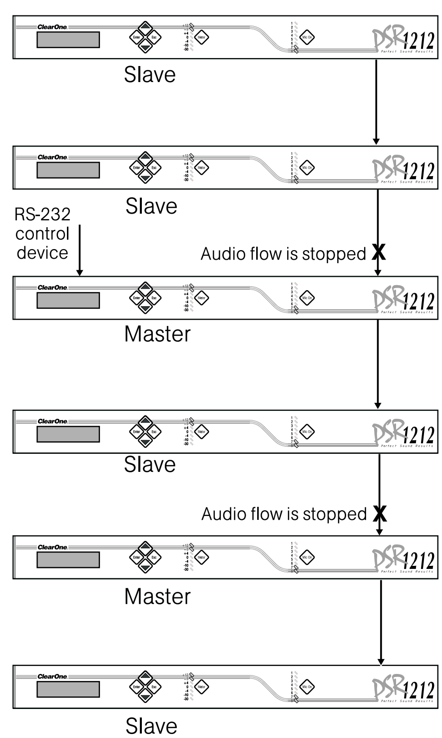

Figure 3.6. PSR1212 automixing gate functions

Gating groups

In addition to specifying gating characteristics for each mic input, you can assign the inputs to a gating group for greater flexibility and control. When inputs are assigned to a gating group, the gating information from the inputs is used to control how the entire mixer behaves.

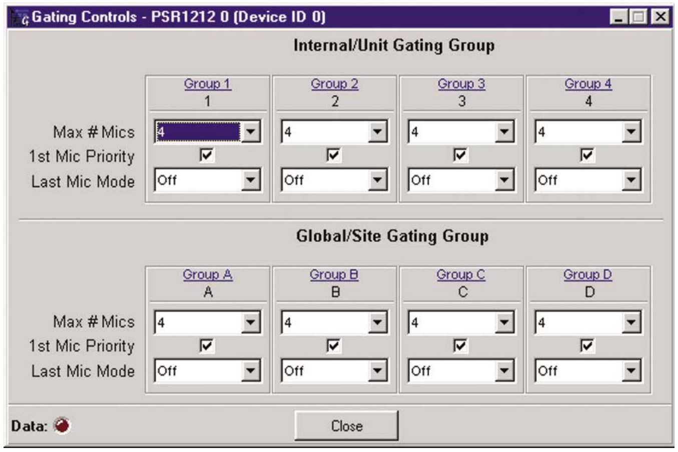

Figure 3.7. PSR1212 automixing gate functions

The PSR1212 features four internal gating groups (Internal 1–4) and four global gating groups across the expansion bus (Global A–D). Microphones can only be used in one gating group at a time. If an input is not assigned to a gating group, that mic's gate properties are independent and have no effect on any other gating group. When gating groups span two or more units (global gating groups), the group settings must be the same for each unit in the global gating group. Mic inputs are routed to Global Group A by default.

- **Max # Mics.** This sets the maximum number of microphones that can be gated on at any one time within a gating group. For internal groups, the maximum number of mics can be from 1 to 8. For global groups the maximum number of mics will vary depending on how many mic inputs are assigned to the gating group (up to 64). You can also select All for the global groups—which means all mics could gate on.

- **1st Mic Priority.** This setting helps maintain maximum audio intelligibility by allowing only one mic to gate on to a participant's voice. 1st Mic Priority allows more than one microphone to be activated at same time—it simply restricts mics from gating on to the same audio source. It does this by determining the audio level received by all mics when the first mic is gated on. This audio level is then used as the ambient level for the gating group. If this feature is disabled, usually two or more microphones gate on when only one person speaks.

- **Last Mic Mode** list. There are two options for Internal groups: Last On and Off. If you are configuring Global groups, you can select from the mic inputs assigned to the Global group in addition to Last On and Off.

  Last On leaves the mic that was activated last full on until another mic input is gated on. The Input setting allows you to select which mic input the PSR1212 leaves on when all mics gate off. In a boardroom or meeting room application, this feature can be used to specify the chairperson's mic as the last mic on. If Off is selected, all mics will gate off when no audio is present.

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

CHAPTER 4: Audio Routing

Matrix Mixing

One of the more important functions of the PSR1212 is matrix routing of audio signals. Like all device functions, routing is executed in the digital domain. Routing is configured using G-Ware and a direct connection between a PC and the PSR1212 unit. Changes in routing can be executed via the RS-232 port and/or via presets on the control/status connector.

The PSR1212 audio matrix has 32 input and 32 output destinations, with level control at each cross point. Custom labels can be configured for each input and output. Inputs and outputs to the matrix are described on the next page.

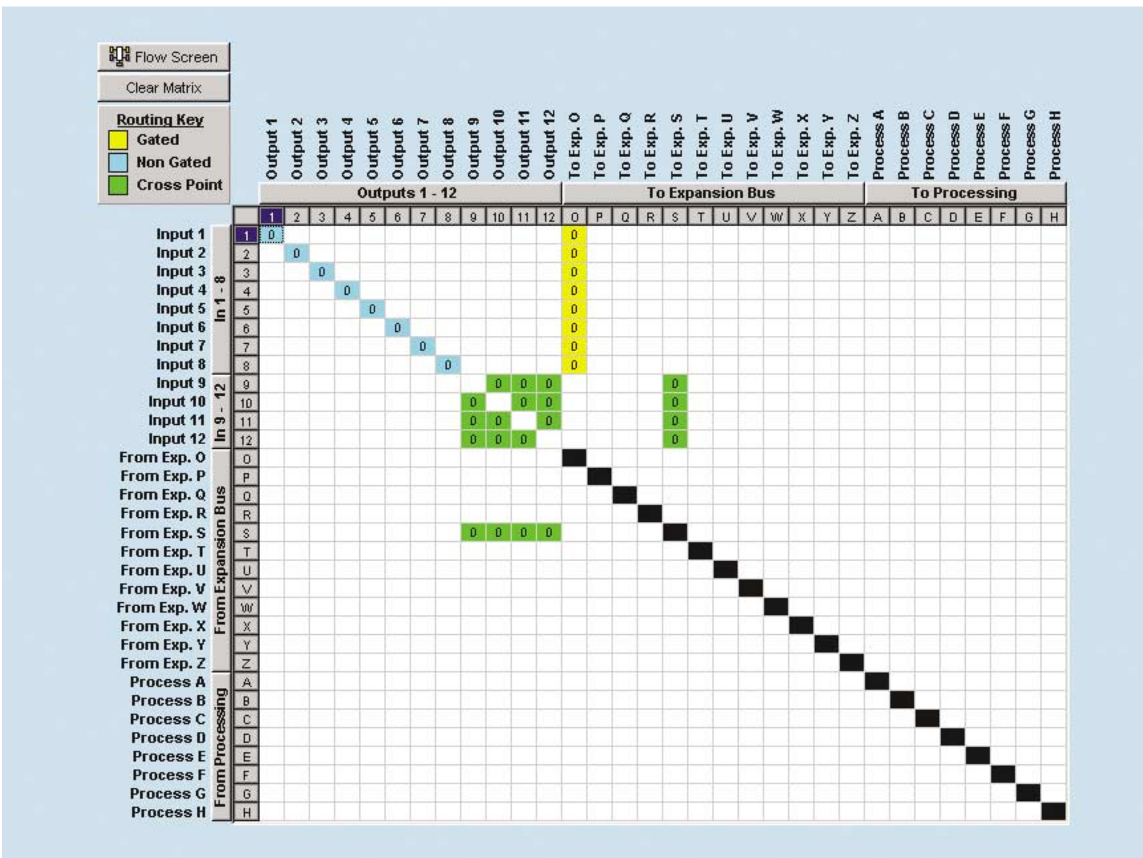

Figure 4.1. Default routing in the Matrix Screen

Inputs

- **Gated and Non-gated Inputs 1–8.** Inputs 1–8 (selectable for mic or line level) are located on the rear terminal block. Both gated and non-gated inputs are provided on the matrix for delivery to desired destinations. This is provided because, in some applications (such as a courtroom), direct, non-gated outputs are required. Default routing for gated microphone inputs are to the O-Bus. Non-gated outputs are routed by default to their corresponding output number (i.e., Input 1 is routed to Output 1).
- **Inputs 9–12.** 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

- **Outputs 1–8.** 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 with 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 O–Z

Audio on any PSR unit in the E-bus network can be placed on the bus or taken off the bus and routed to any destination within the unit. The PSR1212 has 12 digital mix-minus buses:

- O–R buses are defaulted as the mic mix buses and can communicate the NOM count. Gated mics are default routed to the O bus.
- S–Z buses are defaulted as the auxiliary mix buses. They are used to carry auxiliary audio such as that from VCRs and CD players. These buses are used as mic mix buses when NOM count is not required.

Processing A–H

There are eight processing blocks on the PSR1212 (Processing A–H). With these processing channels, you can apply filters, EQ, or other processing settings to an input or a group of inputs which can then be routed to a single output or group of outputs. These buses are typically used to reduce feedback in the venue and provide crossovers for different speaker systems.

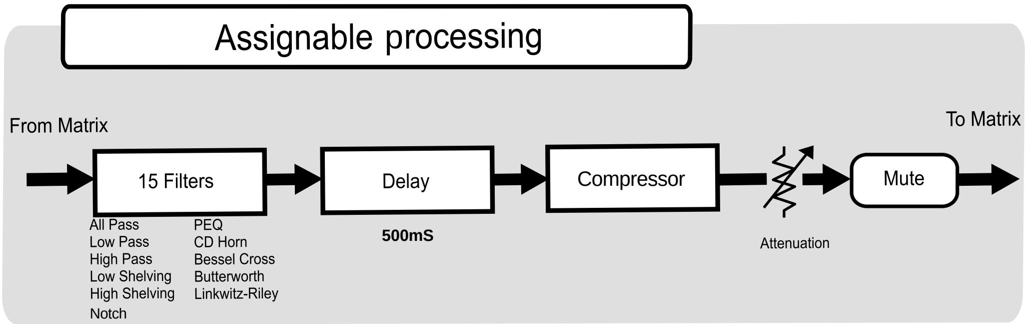

Figure 4.2. Assignable processing

Worksheets

The worksheets provided on the following page are to help you understand all PSR1212 system functions as you design a system. Refer to the back of this document for a complete set of PSR1212 worksheets. Default settings appear in bold.

.

PSR1212 System Parameters Worksheet

ClearOne PSR1212

Systemwide Parameters

Program Parameter	Selection Range		Program Parameter	Selection Range	
Timeout	0 - 15 (**5**)		Mixer Mode	**Master**, Slave	
Lock Front Panel	On, **Off**		RS-232 Baud Rate	9.6, 19.2, **38.4**, 57.6, 115.2 kbps	
Set Passcode	Any 5 Front Panel Keys		RS-232 Flow Control	On, **Off**	
Device ID No.	**0** - 7		Modem Mode	On, **Off**	
Unit ID No.	Factory Programmed		Clear Modem Password	**Enter**	
			Default Meter	Input/Output Channel 1-12 **(Out 12)**	

Mixer Group Parameters

		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 4.3. System Parameters worksheet

PSR1212 Input/Output Parameters Worksheet

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

Figure 4.4. Input/Output Parameters worksheet

CHAPTER 5: System Control

The PSR1212 provides a variety of options for system control. You can create up to 32 presets and up to 255 macros to change whole room configurations or run a series of commands. Presets, macros, and commands can be executed using any of the following control options: custom control through Control/Status port A, contact closure through Control/Status port B, ClearOne Control Panels or XAP IR Remote through the RS-485 ports, serially with a touch panel, modem, or PC through the RS-232 port, or front panel LCD menus.

Presets and Macros

A preset is simply a group of routing and configuration settings stored in the PSR1212. These settings are applied to the unit when the preset is executed. A good way to think of presets is to consider each preset as a room configuration option. You can create up to 32 presets which enables you to accommodate changing room requirements quickly and efficiently. PSR1212 presets are unique in the sense that they operate independently of other presets in the unit. When a preset is run, only the selected inputs/outputs are changed—all other settings in other presets remain unchanged and are not reset. This means you can change audio routing and configuration settings in a room without affecting settings in other rooms (such as gain).

Presets can be executed in a variety of different ways including the Execute Preset utility in G-Ware, the front panel controls of the PSR1212, RS-232 external control devices, RS-485 control devices (ClearOne Control Panel and XAP IR Remote), logic in/out, and contact closure. You can also create macros which can run multiple presets. These options give you tremendous flexibility when designing your installations and are described in more detail in the PSR1212 Installation and Operation manual.

Macros provide powerful options for controlling and operating your PSR1212. A macro can contain multiple commands that can reference a single unit or multiple units across the expansion bus. Each PSR1212 is designed to support up to 255 macros, with an average of 150 command lines each. Macros are created in G-Ware using the Macro Recorder, which records your onscreen selections, or the Macro Editor, which allows you to directly create command lines. The Macro Editor is also used to edit macros created using the Macro Recorder. For more information on creating and using macros, consult the PSR1212 Installation and Operation manual.

Control and Status Connectors

Control/Status connections are provided on two DB-25 connectors. These connectors are labeled Control/Status A and Control/Status B and contain different types of pins. Control pins on Control/Status A are momentary while control pins on Control/Status B are latching. The inputs on these connectors are internally pulled high and are activated by connecting the pin to ground. The outputs are open collectors, which are open when inactive and grounded when active. This allows the PSR1212 to control and be controlled by a wide variety of external devices, including relays, lamps, switches, and other equipment.

Control/Status A

The GPIO (general-purpose input/output) Builder in G-Ware is used to establish the pin assignments for the 16 user definable pins on Control/Status Port A. These pins provide control via contact closure and status via open collector functions in the unit. Note that the pins numbered in blue are command pins; the pins numbered in green are status pins. The default pin assignments are listed on page 26.

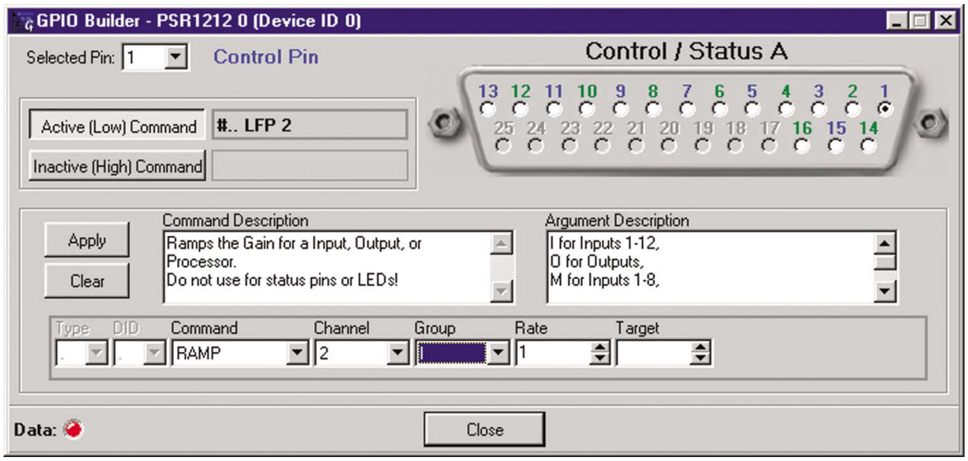

Figure 5.1. Customizing pin assignments with GPIO Builder

Control/Status B

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

Figure 5.2. Preset Mask Control/Status B

A typical use for Preset Mask Control Status B is a room combining application which uses automatic partitions with sensors or triggers which set the pin to high (H) when the partition is closed and to low (L) when the partition is open. For example, if pin 1 is connected to the first partition and pin 3 is connected to the second partition, then the Preset Mask Control Status B settings shown in Figure 5.2 will activate the preset when the first partition is open and the second partition is closed.

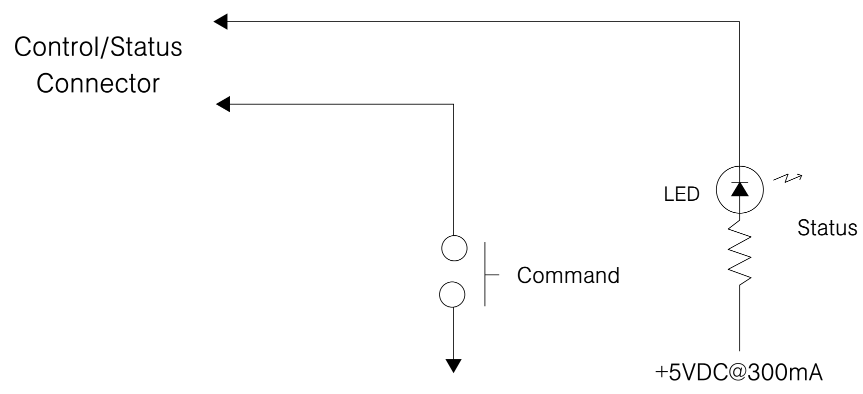

Figure 5.3. Direct control/status operation

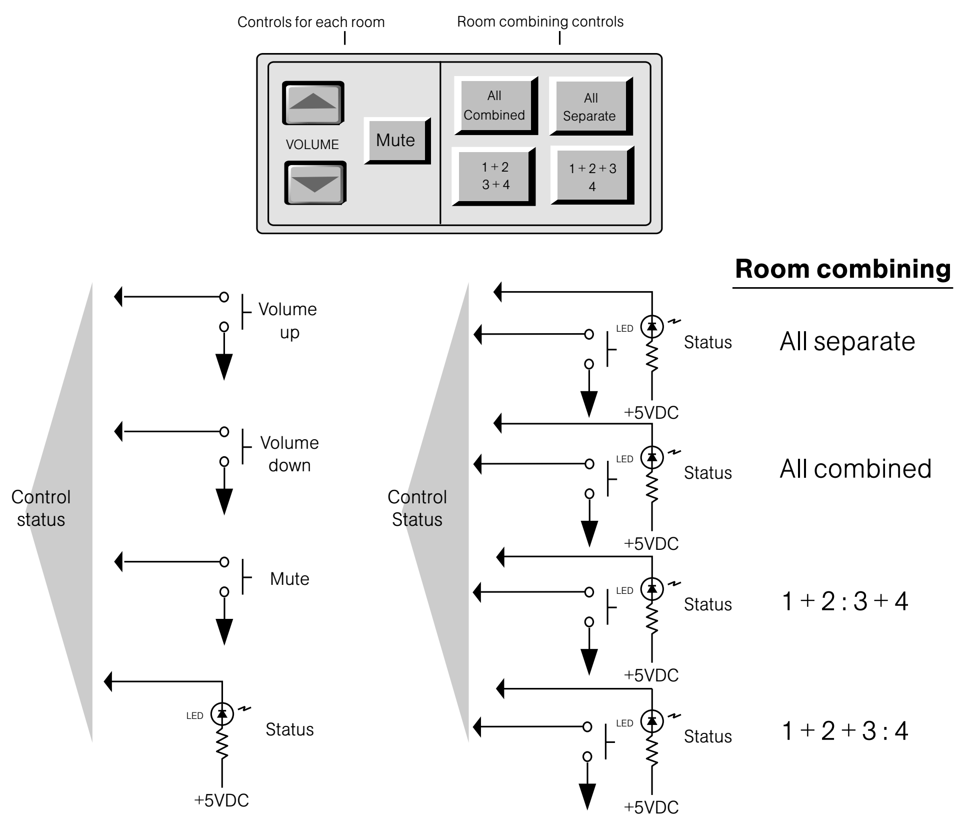

Figure 5.4. Room combining using control/status pins

Control/Status A

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

Control/Status B

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

Figure 5.5. Default pin programming

ClearOne Control Devices

ClearOne manufactures three control devices designed for use with the PSR1212: Volume Control Panel, Select Control Panel, and XAP IR Remote Control. These devices are programmed using the Remote Builder in G-Ware. These control devices are connected to Remote Panel A or Remote Panel B—the RS-485 connectors.

Volume and Select Control Panels

ClearOne Control Panels are convenient wall panels which provide control over the PSR1212 system. There are two Control Panel models: Volume and Select. Volume has three programable buttons that were primarily designed to control gain and mute but can be programmed with any command. It also has eight LEDs that can be programmed to indicate certain gain levels or muting. As with the buttons, these LEDs can be programmed to represent any command status. The Select Control Panel can be programmed to execute various commands such as presets for room combining applications. You can connect up to six Control Panels in daisy chain fashion to each RS-485 port.

XAP IR Remote

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

Figure 5.6. Volume (left) and Select Control Panels

Figure 5.7. XAP IR Remote

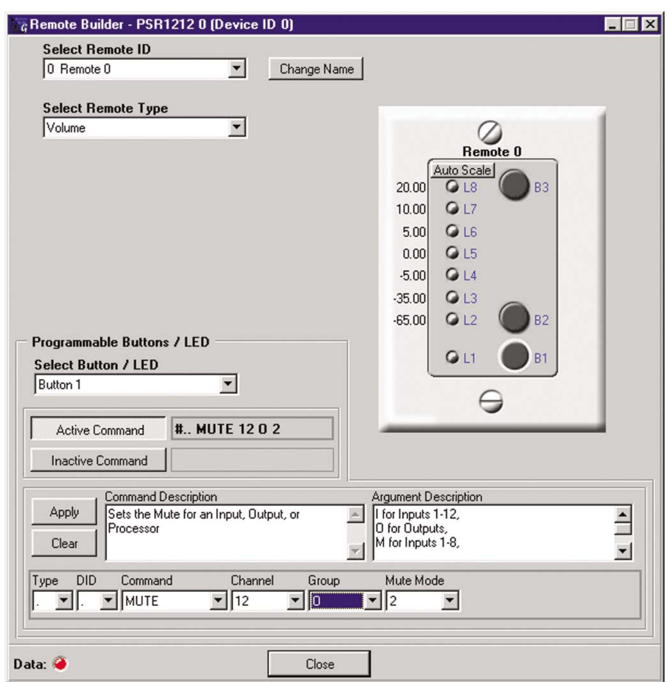

Figure 5.8. Programming a Volume Panel in Remote Builder

Serial Control (RS-232)

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 G-Ware, 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

Front Panel

The PSR1212's front panel is intuitive to operate, thanks to its simple interface: a 2x16 character LCD, menu buttons, and a peak-level LED bar meter. Although most of the PSR1212's features are programmed with G-Ware software, the front panel can be used for simple adjustments and meter monitoring.

To prevent unauthorized changes, the PSR1212 can be password protected. When the unit is locked, navigation of the menus is allowed without a password; however, changes to programming require a valid password.

LCD menu tree

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

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

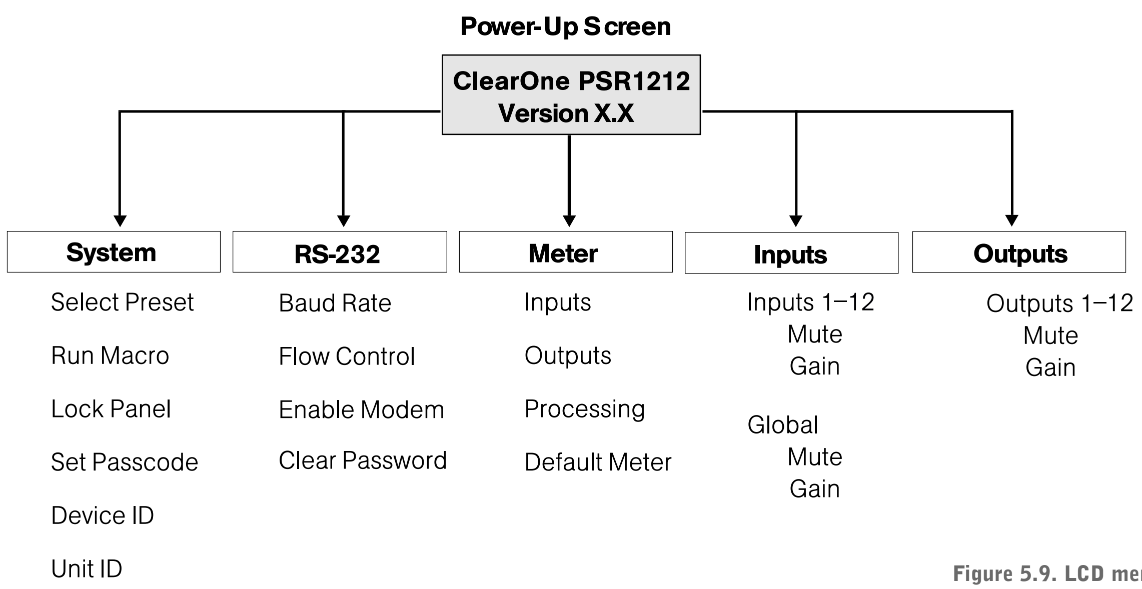

Figure 5.9. LCD menu tree

CHAPTER 6: 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 DB-25 connectors on the rear of the PSR1212.

Expansion Bus connection

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

Serial RS-232

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

Passcodes

To prevent unwanted access via the front panel or modem, the unit can be programmed to require a passcode. 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 6.1). 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.

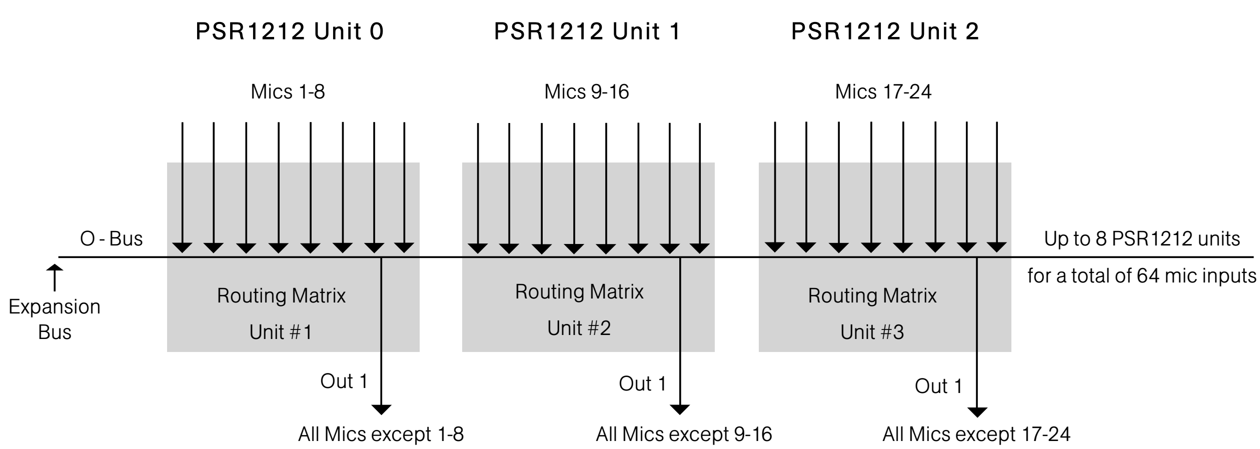

Figure 6.1. Mix-minus configuration of the O bus

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 9) 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 14 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 (24 meters). ClearOne Communications recommends that category five twisted-pair (10BaseT) cable be used.

CHAPTER 7: Applications

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 7.1) 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 8.1). 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

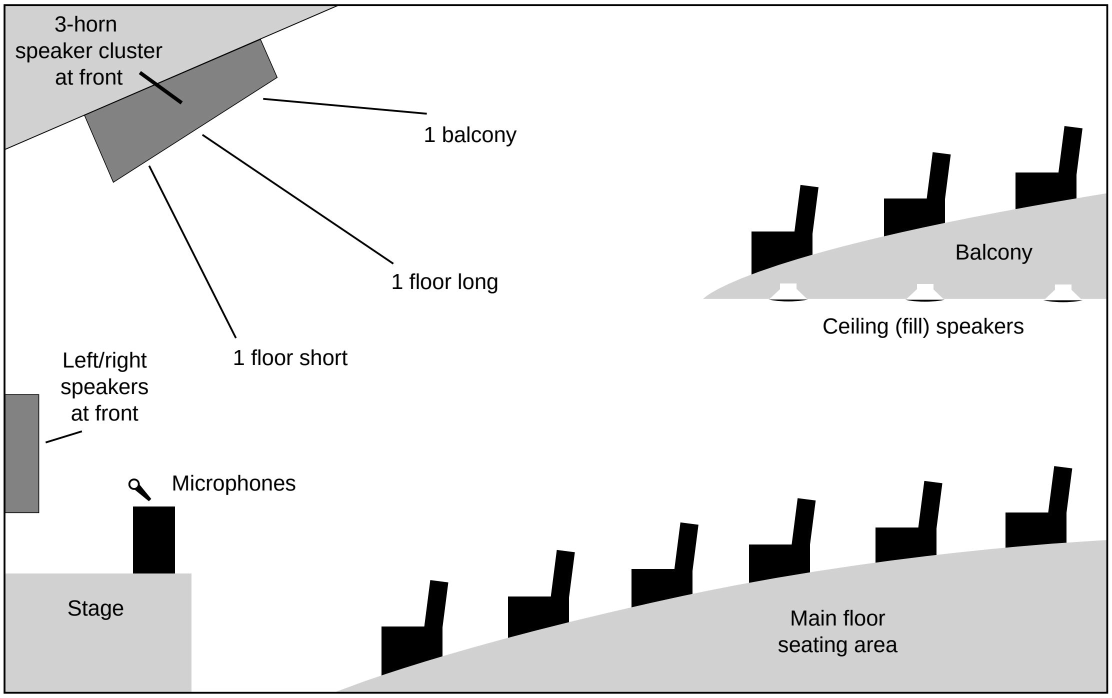

Figure 7.1. Typical auditorium layout

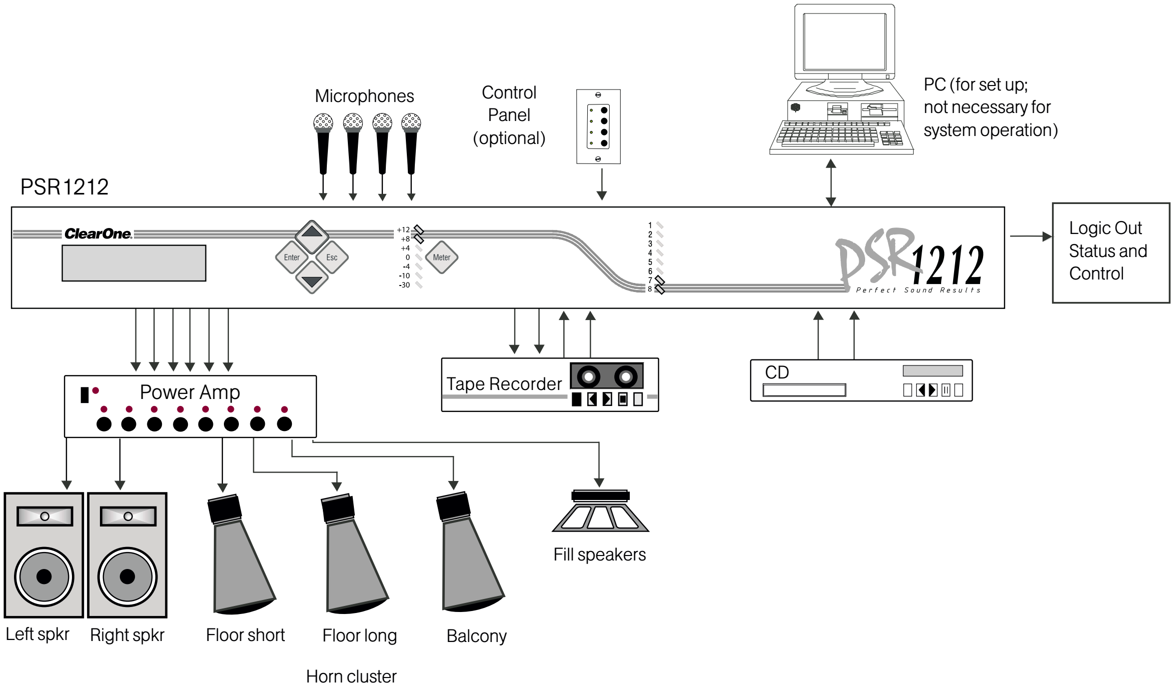

Figure 7.2. Auditorium installation diagram

Arena Installation

Arenas present numerous acoustical challenges, including high ambient sound levels and generally poor acoustical characteristics. The PSR1212 can help adapt the sound 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 7.3) and the arena installation diagram (Figure 7.4) 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.

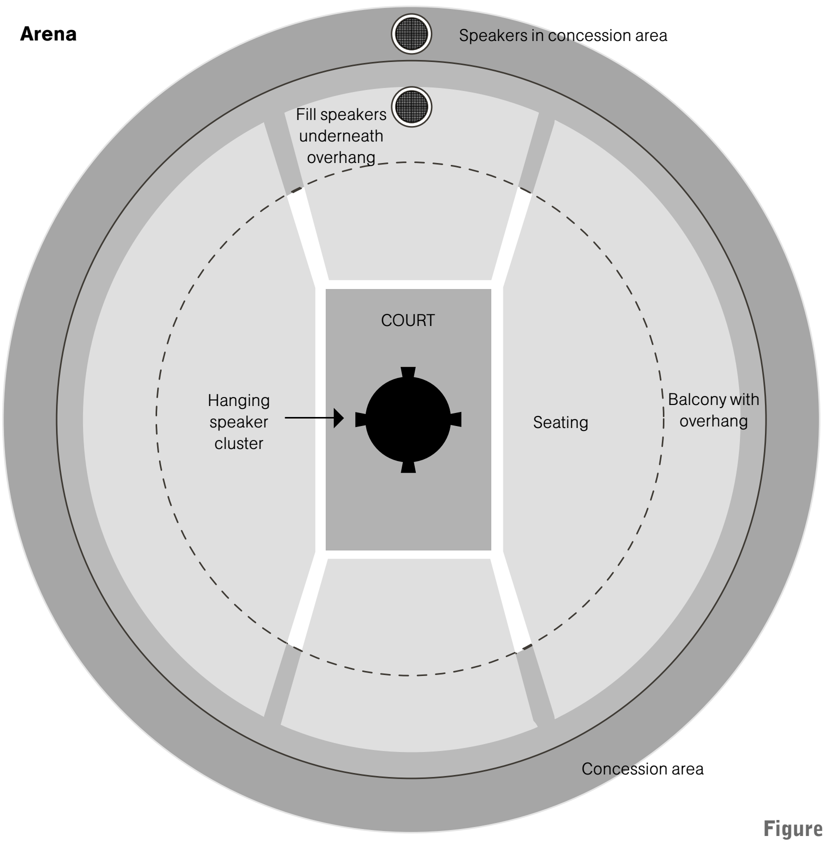

Figure 7.3. Arena layout

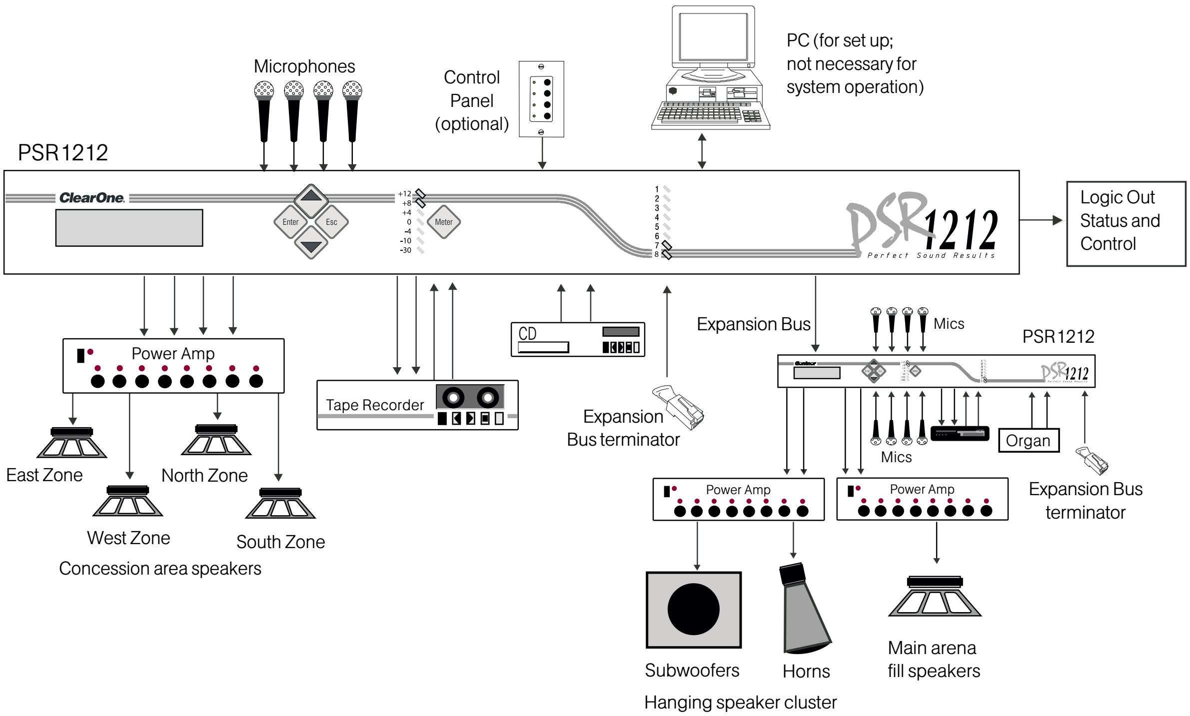

Figure 7.4. Arena installation

Movie Theater Installation

A properly designed and calibrated sound system makes all the difference in the enjoyment of a film or movie presentation. Unlike 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 7.5, and a connection diagram is shown in Figure 7.6. 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 7.5. Movie theater layout

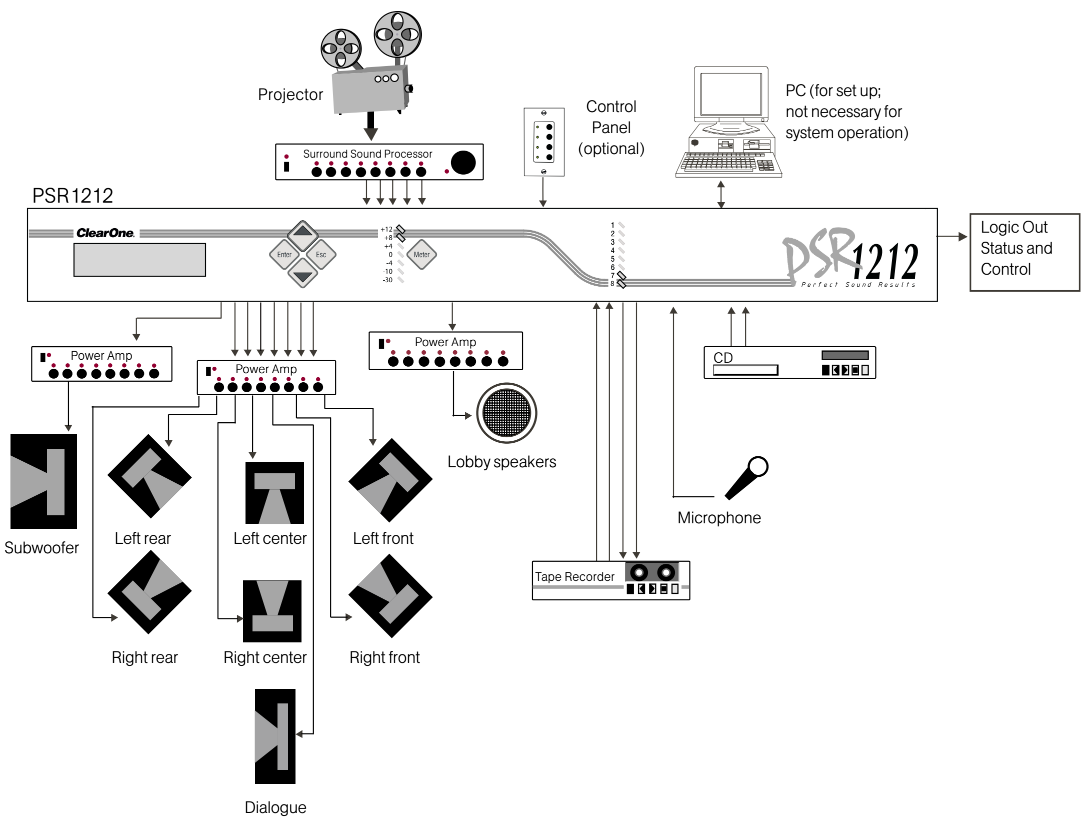

Figure 7.6. Movie theater installation

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

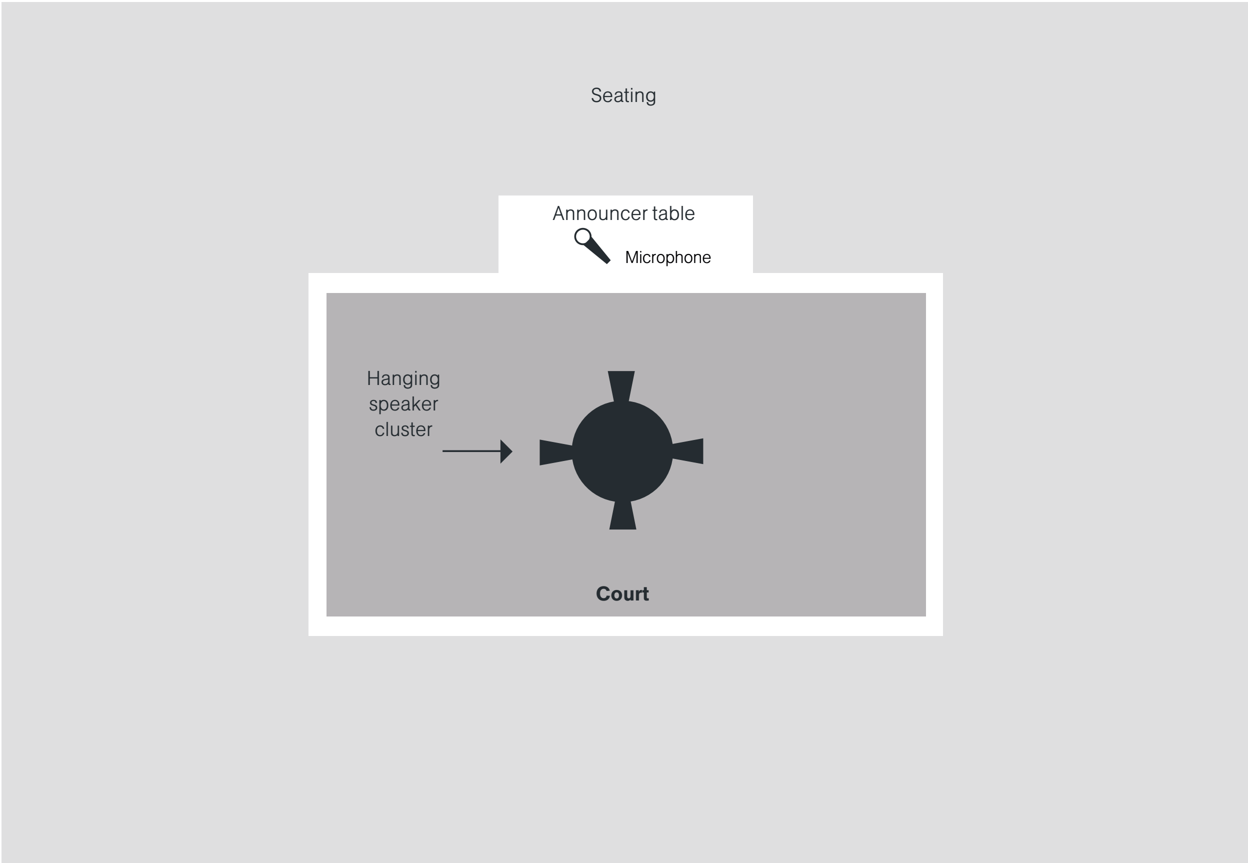

Figure 7.7. Gymnasium

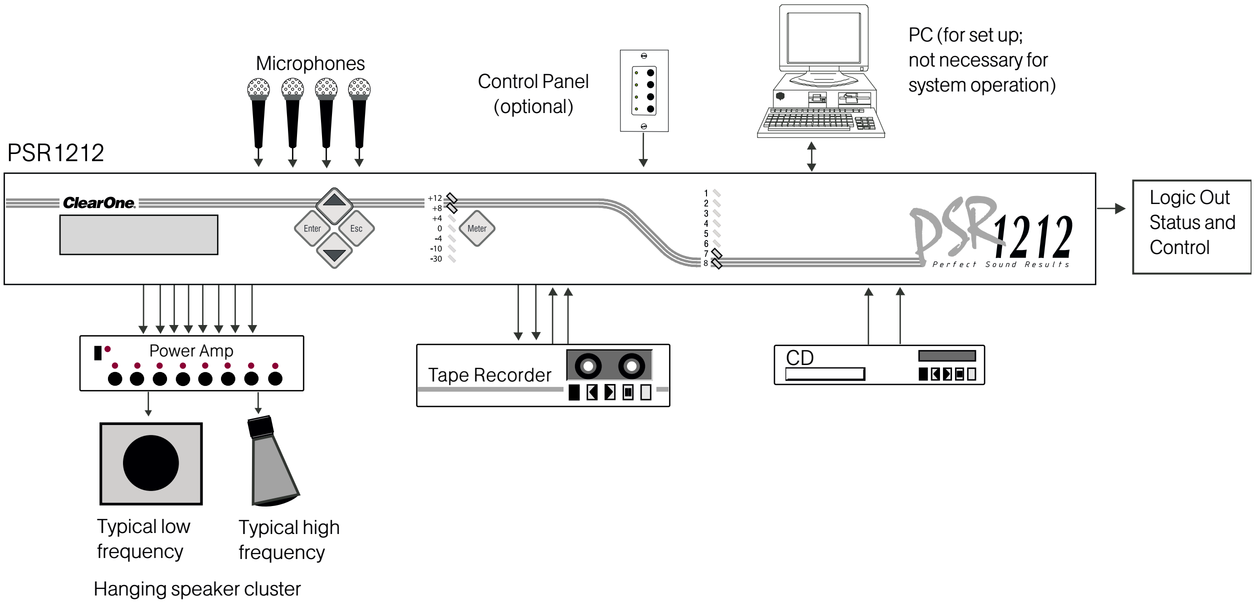

Figure 7.8. Gymnasium installation

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 7.9 shows four rooms with removable partitions. The PSR1212 can be pre-configured 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.

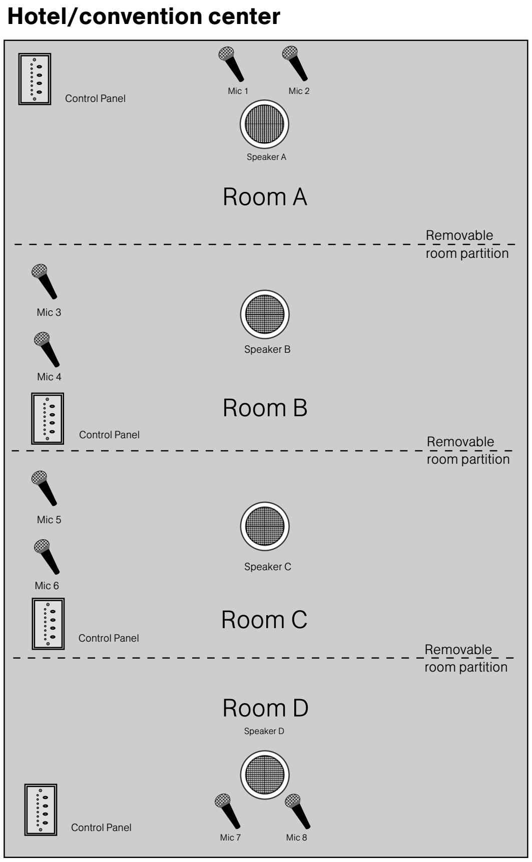

Figure 7.9. Hotel/convention center

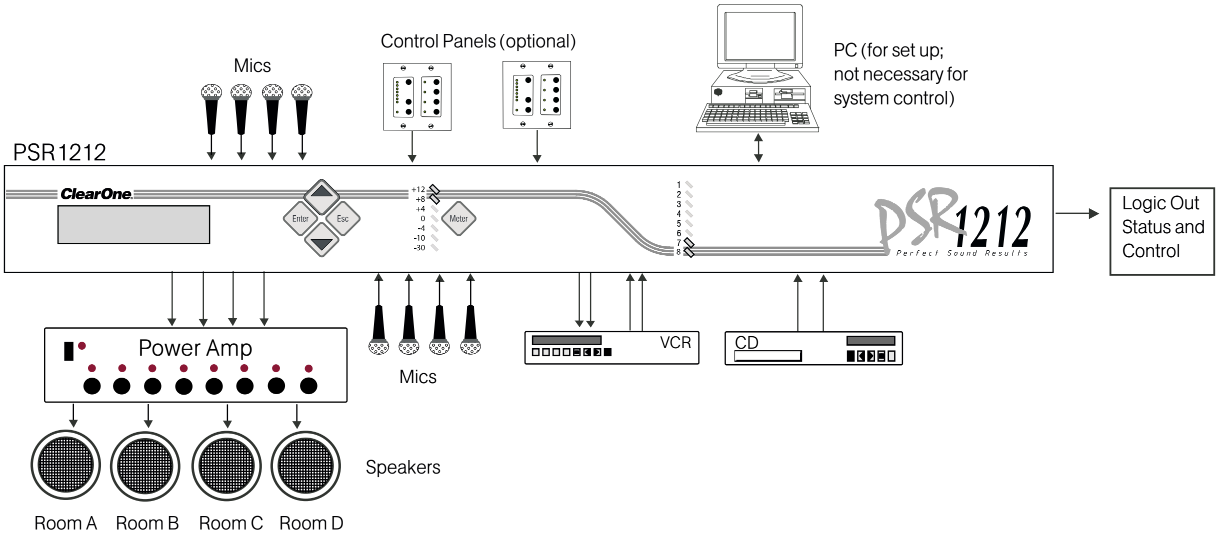

Figure 7.10. Hotel/convention center installation

Conference Room Installation

Figure 7.11 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.

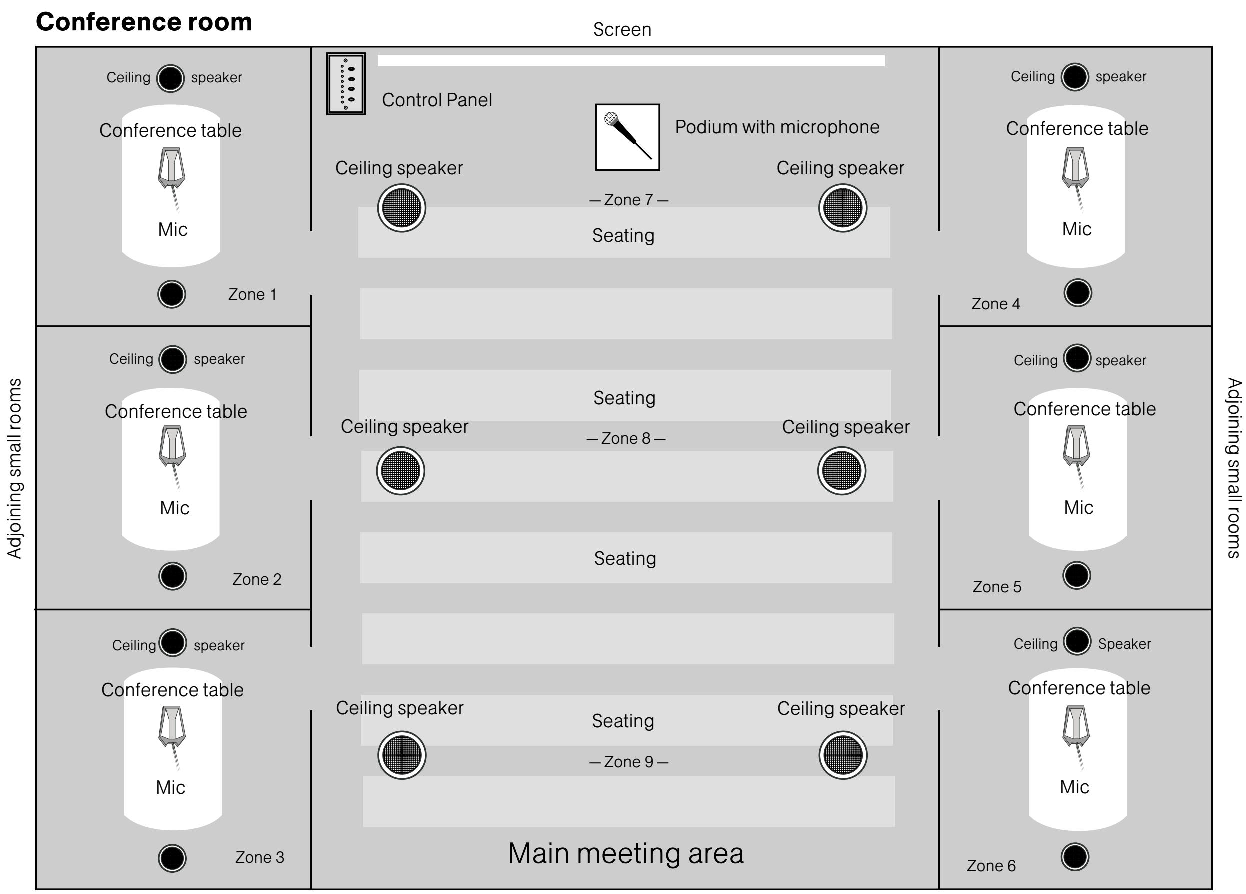

Figure 7.11. Conference room

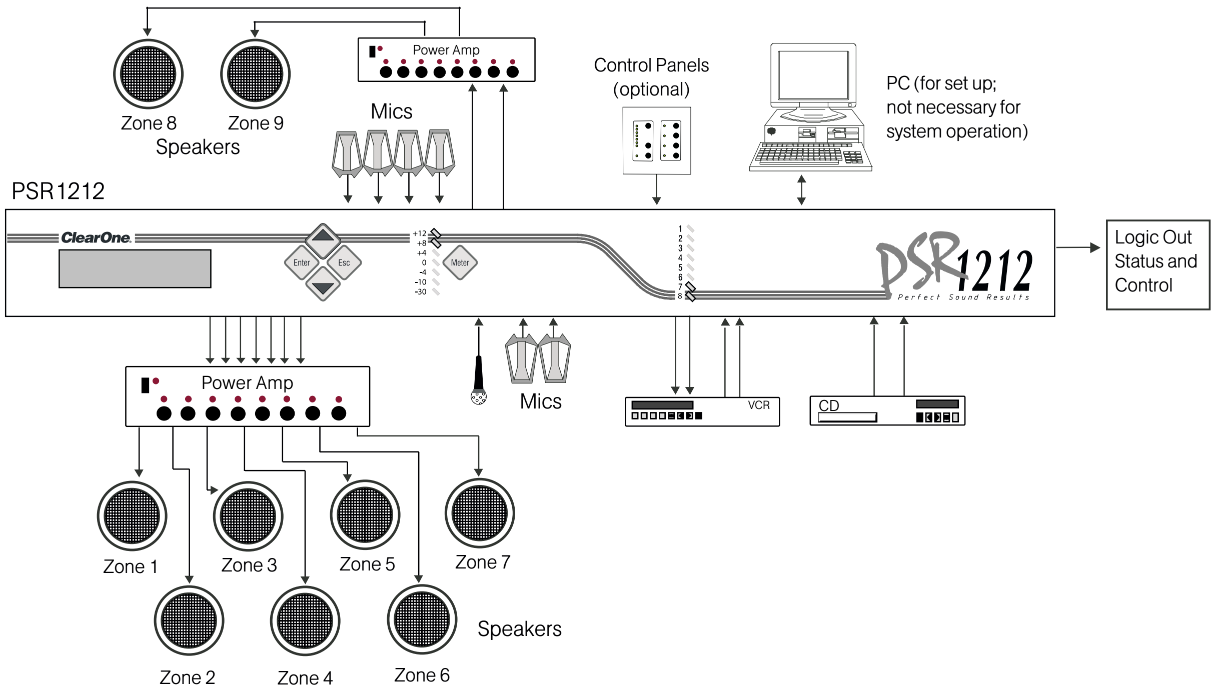

Figure 7.12. Conference room installation

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 7.13 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.

Training room

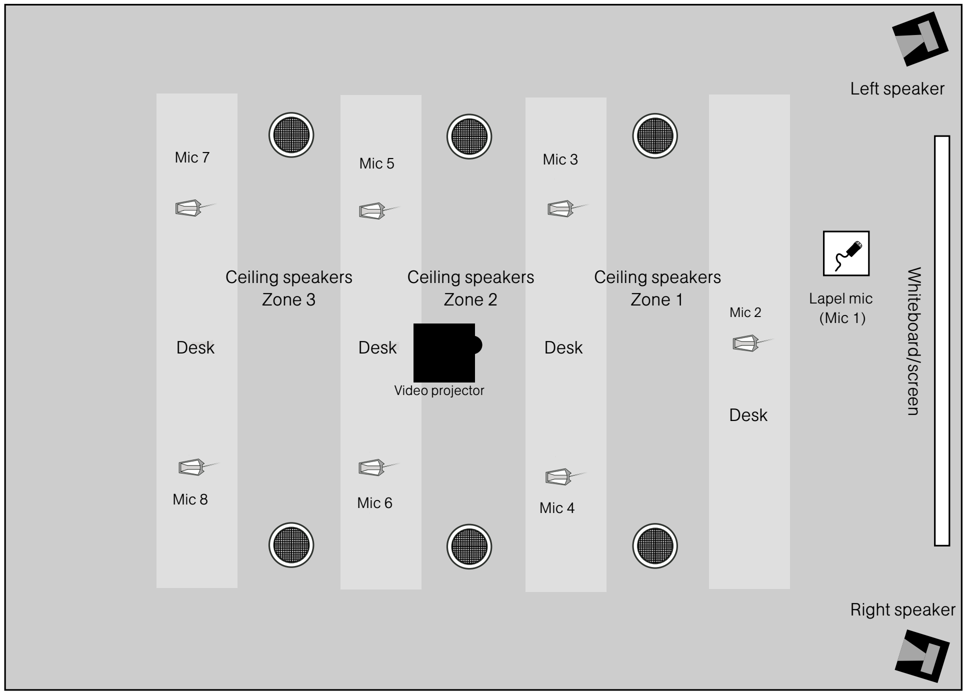

Figure 7.13. Training room

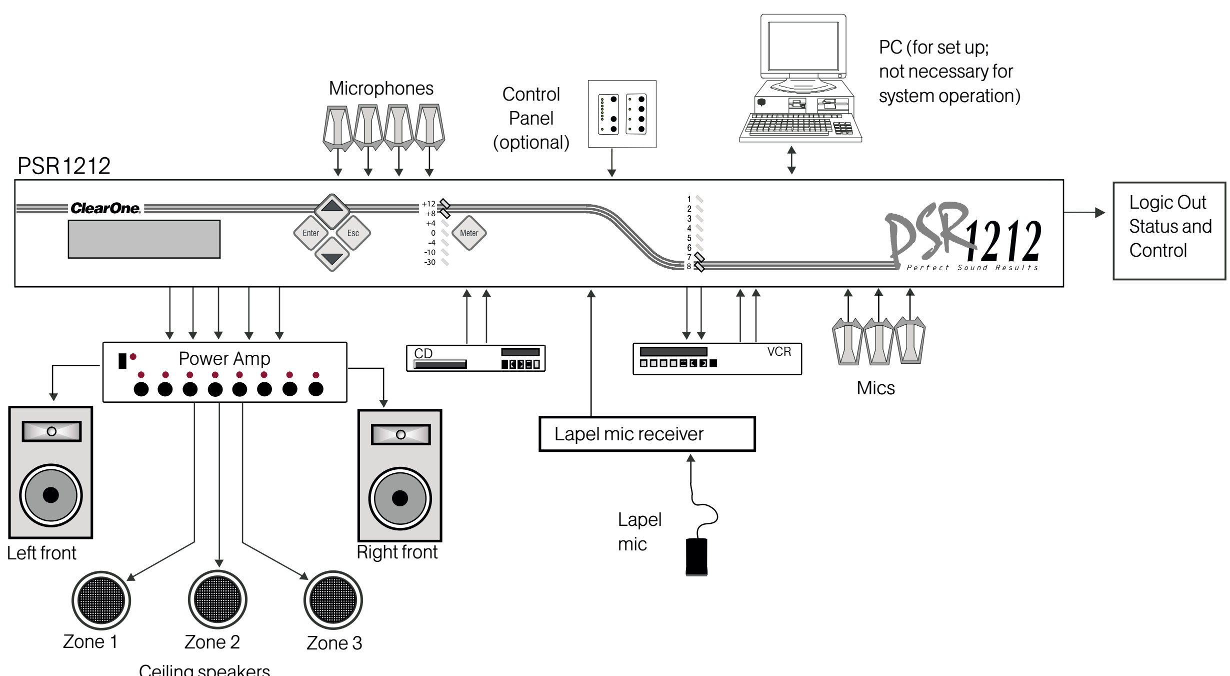

Figure 7.14. Training room installation

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 7.15 shows a boardroom scenario with eight participants, each with his own microphone and loudspeaker. Also, there are observer seating areas on the periphery of the room with loudspeakers for monitoring the discussion at the table.

A typical PSR1212 installation involves establishing a preset where the chairman's microphone (mic 1) is given gating and override priorities over the other microphones. Also, whenever a given microphone gates on, audio to that participant's loudspeaker gates 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 can be configured for particular microphones to enhance the voice qualities of regular participants who sit in assigned seats. All audio is routed to the observation areas at normal levels.

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

Boardroom

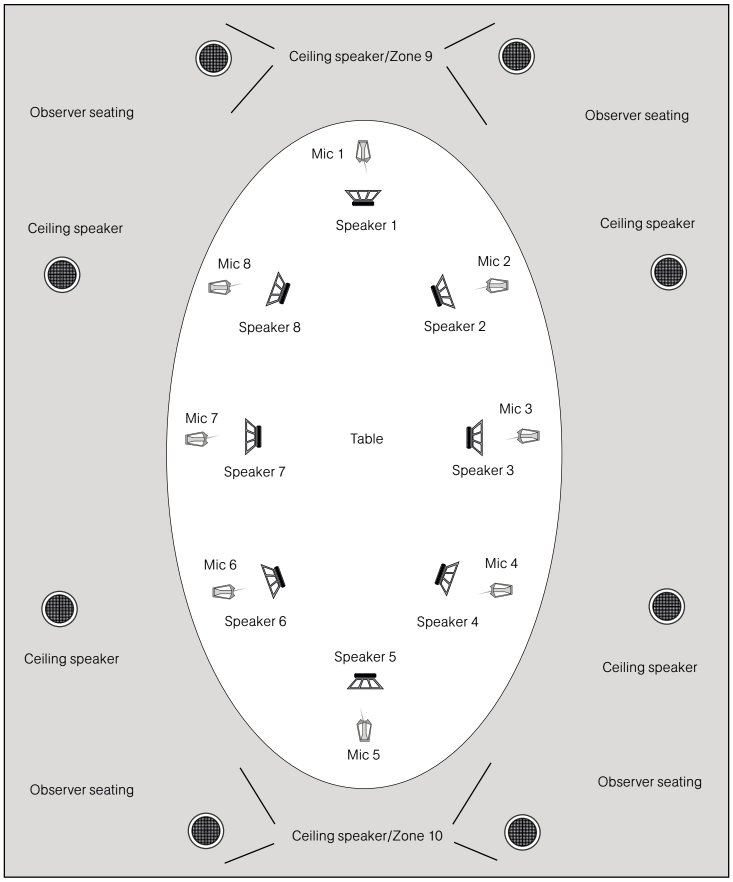

Figure 7.15. Boardroom

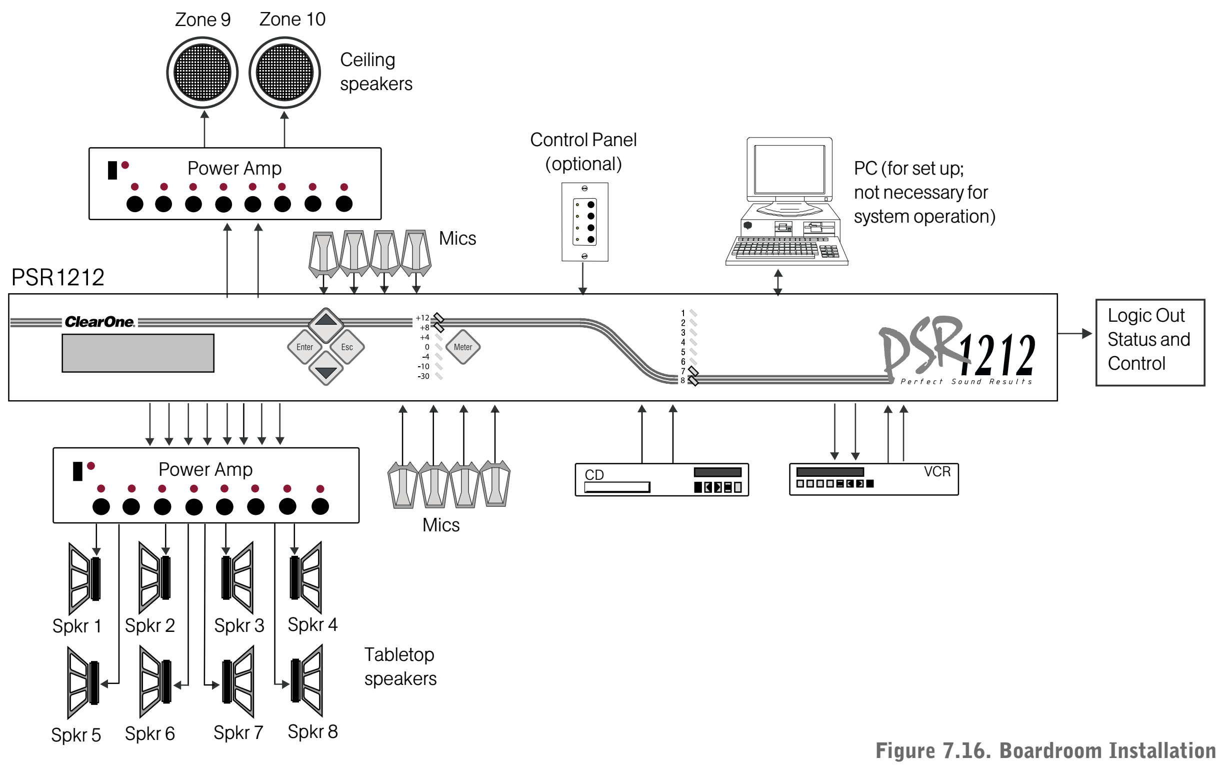

Figure 7.16. Boardroom Installation

Corporate Paging System

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 7.17 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.

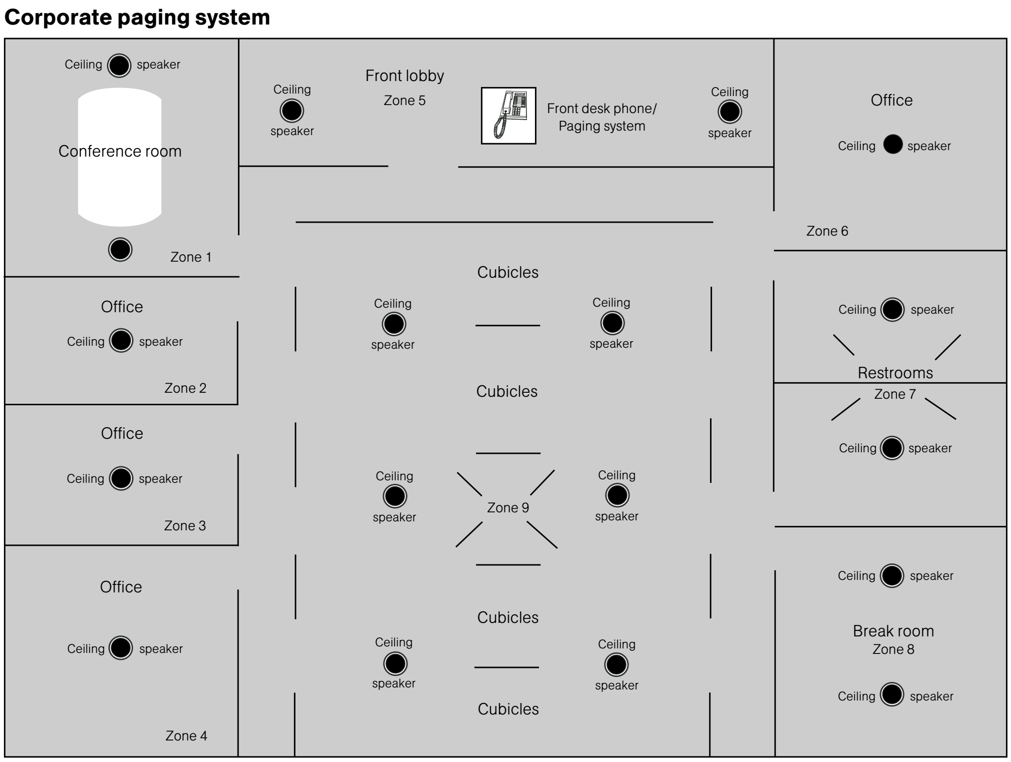

Figure 7.17. Corporate paging system

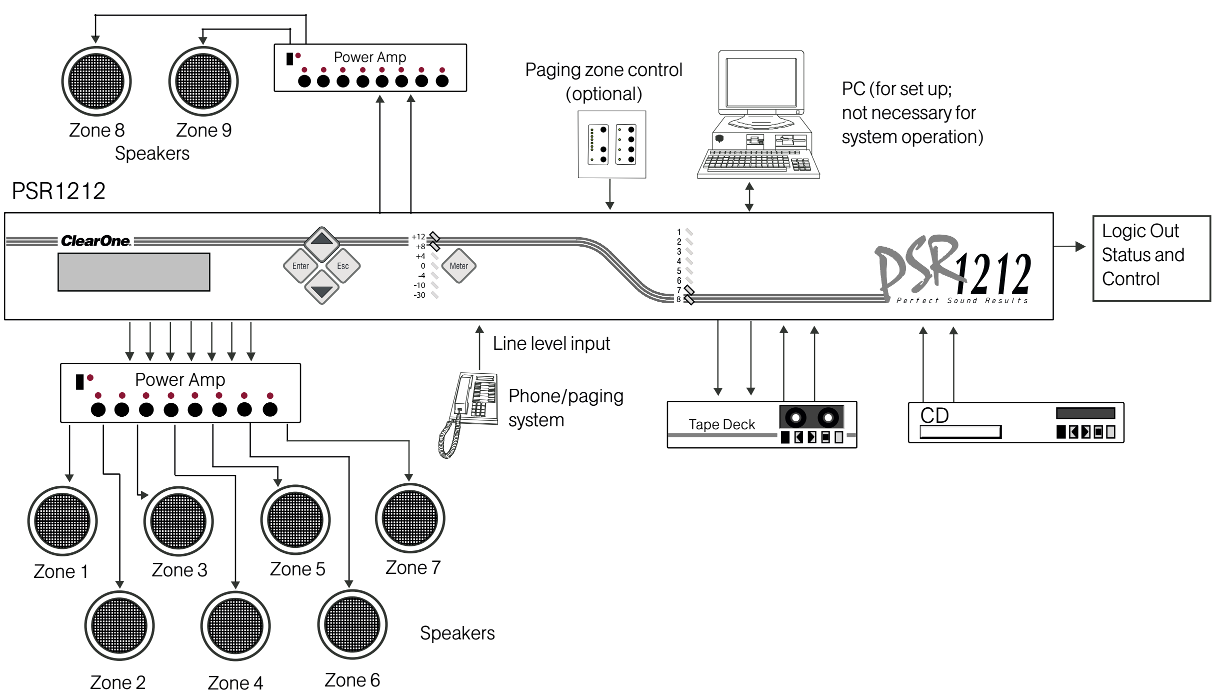

Figure 7.18. Corporate paging system installation

Appendices

Appendix A: Specifications

Dimensions (LxDxH)
17.25" x 10.25" x 1.25"
43.8 x 26 x 4.5 cm

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

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

Humidity
15% to 80%, non-condensing

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

Power Consumption
30W typical

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

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

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

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

Mic/Line inputs 1–8
Push-on terminal block, balanced, bridging
Impedance: 5kΩ
Nominal Level: adjustable -55dBu, -25dBu, 0dBu
Maximum Level: -35dBu, -5dBu, +20dBu
Phantom Power: 24V, selectable

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

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

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

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

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

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

Matrix Mixing Parameters
32x32 matrix
12 analog in/out
12 expansion bus in/out
8 assignable processing blocks in/out

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

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

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

Set-up Software
G-Ware

Appendix B: Architectural and Engineering Specifications

The digital matrix mixer with audio processing shall incorporate microphone mixing, matrix mixing, and signal processing in a single rack space unit.

The matrix mixer shall have 12 inputs and outputs: four line level inputs, eight microphone/line selectable inputs, and 12 line level outputs. Each mic/line input shall have four selectable filters, which include all-pass, high-pass, low-pass, and notch; automatic gain control; phantom power; and automatic microphone mixing capabilities. The unit shall have four internal and four global automatic microphone mixers, each with fully adjustable parameters. The microphone mixer shall use PA adaptive, adaptive ambient, chairman override, first mic priority, last mic mode, number of open mics.

The matrix mixer shall have a 12x12 internal matrix mixer with attenuation at every cross point in .5dB steps. Any input can be routed to any output or multiple outputs. The matrix shall consist of 12 analog inputs/outputs, 12 digital inputs/outputs from the network bus, and eight inputs/outputs from the processing blocks.

Signal processing shall be provided by eight assignable processing blocks, each with 15 programmable filters, delay, and compression. The processing blocks shall include such filters as high pass, low pass, all pass, low shelving, high shelving, notch, parametric EQ, CD horn, Bessel crossover, Butterworth crossover, and Linkwitz-Riley crossover. Filter setup shall be real-time. The unit shall include a signal generator for pink noise, white noise, and tone sweep capabilities, and shall be assignable to any input on any linked unit. Signal delay is adjustable up to 500ms.

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

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

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

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

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

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

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

The ClearOne PSR1212 is specified.

Appendix C: Worksheets

PSR1212 System Parameters Worksheet

ClearOne PSR1212

Systemwide Parameters

Program Parameter	Selection Range		Program Parameter	Selection Range	
Timeout	0 - 15 (**5**)		Mixer Mode	**Master**, Slave	
Lock Front Panel	On, **Off**		RS-232 Baud Rate	9.6, 19.2, **38.4**, 57.6, 115.2 kbps	
Set Passcode	Any 5 Front Panel Keys		RS-232 Flow Control	On, **Off**	
Device ID No.	**0** - 7		Modem Mode	On, **Off**	
Unit ID No.	Factory Programmed		Clear Modem Password	**Enter**	
			Default Meter	Input/Output Channel 1-12 **(Out 12)**	

Mixer Group Parameters

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

Define PA Adapt 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

ClearOne 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	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

ClearOne 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

ClearOne PSR1212

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)*																

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.						
DB25 Remote Control Port "B" Setup						
DB25 Port "B" Pin #	1 3 5 7 9 11 13 15 17 19	1 3 5 7 9 11 13 15 17 19	1 3 5 7 9 11 13 15 17 19	1 3 5 7 9 11 13 15 17 19	1 3 5 7 9 11 13 15 17 19	1 3 5 7 9 11 13 15 17 19
Preset Mask =						

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

Room Configuration Setup

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

Room Preset Configuration worksheet

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

Appendix D: 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 in the same gating group. When a chairman overide mic in this group gates on, all mics in the group that are not chairman override enabled 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 RJ-45 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 and blocks audio signals based on user-definable requirements of the system.

- **All Pass** A filter that provides only phase shift or phase delay without appreciably changing the magnitude characteristic. The filter produces a flat amplitude response. It is useful for matching the delay of two processing channels with different delays.
- **High Pass** A filter that passes high signal frequencies while attenuating low frequencies. The gain or loss
- **High Shelving** Provides boosting or attenuation of frequencies above a designated frequency. The transition between the spectrum above and below the designated frequency occurs at a fixed 6dB/octave rate. The gain or loss above the corner frequency is adjustable to +/- 15dB.
- **Low Pass** A filter that passes low frequencies while attenuating high frequencies.
- **Low Shelving** Provides boosting or attenuation of frequencies below a designated frequency. The transition between the spectrum above and below the designated frequency occurs at a fixed 6dB/octave rate. The gain or loss below the corner frequency is adjustable to +/- 15dB.

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.

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).

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.

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.

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