AudioNet® Products Manual
780+, 880+, 784+, 884+

Operation and Owner's manual for the
780+ and 880+
Programmable Input Expanders

and the

784+ and 884+
Programmable Matrix Mixer and Control Systems

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Introduction

The ***784+, 784T+, 884+***and ***884T+*** matrix mixers and control systems are all eight input by four output (8X4), computer controllable, matrix mixers. The T+ models are identical to the + models except for the addition of tone controls on each input. The T+ models are also two rack spaces high. (See **Appendix V** titled **Audio Net® T+ Products** on page 76) On all the mixers, any or all eight inputs may be assigned to any or all of the outputs of the mixer. While there are only four main outputs on the *784+, 784T+, 884+* and *884T+*, there are eight internal mix buses in each matrix mixer. This allows two eight by four mixers to be ganged together to provide a sixteen by eight matrix. Additional matrix mixers may be ganged together to increase the total number of inputs, but the number of outputs from the matrix cannot be increased above eight. Automixers (*884+, 884T+*) may be ganged with manual mixers (*784+, 784T+*) to provide a combination of automatic and manual inputs, inputs with tone controls and inputs without them. Two or more automatic matrix mixers may be ganged together to provide the 16 X 8, 24 X 8 etc. matrix, but the automatic inputs from one mixer will not be associated with the automatic inputs from the other mixers in so far as NOM and threshold sensitivity are concerned. Each individual unit will maintain its own individual NOM and threshold.

The ***780+, 780T+, 880+*** and ***880T+*** are eight channel input expanders which have no outputs of their own. They are intended for use with the manual and/or automatic matrix mixers. The *780+* is best described as a *784+* with no outputs, while the *880+* is an *884+* with no outputs. The *780T+* and *880T+* are best described as a *780+* and *880+* with tone controls. Any of the input expanders can be used with any of the matrix mixers. For example, the *880+* which provides eight gated inputs, each with ALC, can be ganged with the *784+* manual matrix mixer, or with the *884+* automatic matrix mixer, or with an *880T+* automatic matrix mixer if tone controls on some inputs are needed.

Almost all of the functions of the mixers and input expanders are controlled by their on-board microprocessors. There are only four internal controls per channel that must be set manually at the time of installation. These include: mic/line level, preamp gain, high pass filter, and phantom power.

The microprocessor controls all of the following functions of the mixers: gating, NOM, automix, VOX and manual input channel assignments (*880+, 880T+, 884+* and *884T+*only), channel input to output assignment, input levels, output levels, input gain trim, level control range, remote control access, presets and priorities. Audio Net+® front and rear panel connectors allow computer access to the mixers.

Manual Arrangement

This manual is divided into two sections. **Section I: Hardware Overview**, describes the mixer and input expander hardware, ie. input/output connections, physical switch selections, etc. All the information necessary to physically install a unit is contained in this section - everything from bolting it into a rack to the hardware activation of presets and room combining functions.

Section II: Using Audio Net+® Software, describes the programming of the mixers using the Windows® based ANSW+ software from Ivie. Computer system requirements are stated, the ANSW+ software (which is very simple and intuitive) is explained, and programming examples are presented.

Section I: Hardware Overview

Power Supply *(Mixers and Input Expanders)*

All of the these products use the same universal power supply. It is UL, CSA and CE approved, and will operate at line voltages from 100 VAC to 240 VAC, 47 Hz to 63 Hz. It comes with a United States standard, three pin power cable. Since the power cable plugs into the power supply, it can be replaced with a power cable having a connector other than the U.S. standard connector.

Power Failure *(Mixers and Input Expanders)*

A reasonable question is, "What happens to the programming on a matrix mixer or input expander if power fails?" The answer is that the memories of these products are nonvolatile, so they "remember" where they were when power went off, and will return to that same state when power is restored.

Getting Started *(Mixers and Input Expanders)*

Default Settings: The *784+* and *884+* are all preset at the factory so that all eight inputs are assigned to all outputs. All inputs are set for microphone input levels. The preamp gain is set to 50dB. The high pass filter is bypassed and the phantom power is turned off. Total gain through the mixer is set for 60dB, which is about 75% of the mixer's available gain.

Important Note: *Before installing a mixer or expander in the rack, be sure to set all input channel options (mic/line, hp filter, phantom power) using the internal dip switches on the PC board. These controls are accessed by loosening the four screws on the side of the mixer and lifting off the top cover.*

Mixers are shipped with a 78C-6, six inch, audio bus interconnect cable. This cable must be plugged into the rear panel of the mixer. It should connect the "Mix bus 1, 2, 3, and 4 Output" to the "Master bus A, B, C, and D Input". A white line is drawn between these two connectors, indicating the standard 78C-6 connection. *There will be no audio output if this cable is either missing or improperly installed. (The 780+ and 880+ have no input to output cable).* Figure 1 below details the 78C-6 connection for an 8 X 4 mixer.

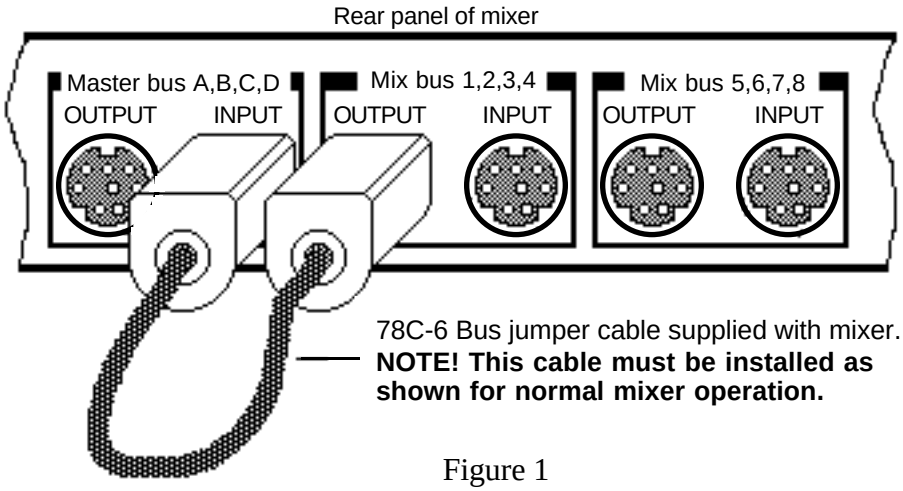


Figure 1

Successful Power Up *(Mixers and Input Expanders)*

The PUP30-32 universal power supply is used with all the mixers and input expanders and provides a broad mains voltage operating range: 100 to 240 VAC, 47 - 63 Hz. Successful powering of the mixers and input expanders can be verified by the lighting of the power indicator LED. When first powered on, the mixers and input expanders initiate a self test which checks a number of their operating parameters to verify proper performance.

Channel Inputs *(Mixers and Input Expanders)*

All channel inputs on all products are electronically balanced. The preamplifier design provides a very low noise input with very high common mode rejection. The input is designed to work with a source impedance of from 50 to 600 Ω and provides excellent RF protection.. The actual input impedance is 2100 Ω.

All channel inputs can accommodate either microphone or line level inputs. Located inside the mixers and input expanders are two switches per channel that allow configuring the gain structure for each input. There is a Mic/Line switch and a 30/50dB gain switch.

The input preamp has a gain of 50dB. With the Mic/Line switch in the Mic position, the input terminals are connected directly to the preamp. With the Mic/Line switch in the Line position, a 30dB pad is inserted between the input terminals and the preamp. The line input has a nominal OdBu input level. The combination of a 30dB pad followed by a 50dB gain stage provides a very high common mode rejection ratio. The greater the common mode rejection, the greater the degree of immunity to common mode noise induced in long cable runs between the mixer and the input connector.

Mic/Line Switch *(Mixers and Input Expanders)*

This switch, located inside the mixers and input expanders, must be set before the units are installed in the rack. It configures the channel input for either a microphone or line level signal. It changes the sensitivity and impedance of the input. In the line position, a 30dB pad is placed in front of the input preamplifier. By maintaining the gain of the preamplifier, the high common mode rejection can also be maintained, as previously discussed. Figure 2 on the following page details the location of the Mic/Line switches inside the mixers and input expanders:

Matrix Mixer and Input Expander Internal Controls

See Following Page for Graphic of Mixer Internal Controls

As shown on the next page, each channel has a high pass (HP) filter, a 30/50dB gain switch, and phantom power switch. These dip switches are located inside the mixer, as are the Mic/Line switches.

It appears that the positioning of the functions on the dip switches inside the mixers do not follow an orderly layout. Actually, the layout is one dictated by electrical necessity to minimize cross-talk, etc. Unfortunately, this layout makes it a little more difficult to follow the functions on the switches, so care should be exercised in setting them up.

30/50dB Gain Switch *(Mixers and Input Expanders)*

The input gain of the preamplifier can be switched between 30 and 50dB of gain, as previously shown. In the 50dB gain position, the input is set to work with standard dynamic microphones. In the 30dB gain position, the input is set to work with microphones that have a higher output level such as condenser microphones. As with the Mic/Line Switch, the gain assignment must be made before the mixer or input expander is installed in the rack.

Phantom Power *(Mixers and Input Expanders)*

Phantom power is available on each input, as Figure 2 shows. The phantom power is +15 VDC for the remote powering of condenser microphones. The voltage must be turned on for each channel that requires phantom power. This is done with the Phantom Power Switch inside the mixers and input expanders. *Again, this assignment must be made before the unit is installed in the rack.*

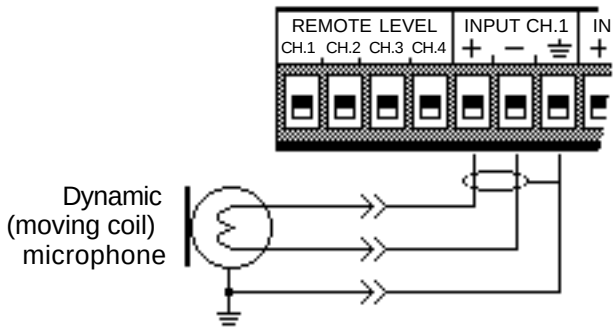
High Pass Filter *(Mixers and Input Expanders)*

There is a high pass filter on each input of the mixers and input expanders (See Figure 2), that can be switched in or out. Units shipped from the factory have this filter switched out, so frequency response is flat.

The High Pass Filter rolls off the lower frequencies. The corner of the rolloff is at 125Hz and has an 18dB per octave slope. The rolloff is designed to reduce p-popping and handling noise of microphones used for vocal applications. The frequency response characteristics of the filter were designed to minimize any affect on the human vocal range.

Connecting a Dynamic (Moving Coil) Microphone *(Mixers and Input Expanders)*

A standard dynamic microphone connects to the mixers and expanders via the Phoenix type connector located on the rear of the units. Pin one of the microphone is connected to the ground terminal on the connector. Pins two (+) and three (-) are connected to their respective terminals. *The Mic/Line switch must be set to the Mic position. The 30/50dB gain switch should be set to the 50dB position.* The setting of the High Pass Filter is optional. The Phantom Power Switch should be set to the OFF position. A typical connection is shown in Figure 3 below:



INTERNAL SWITCH SETTINGS

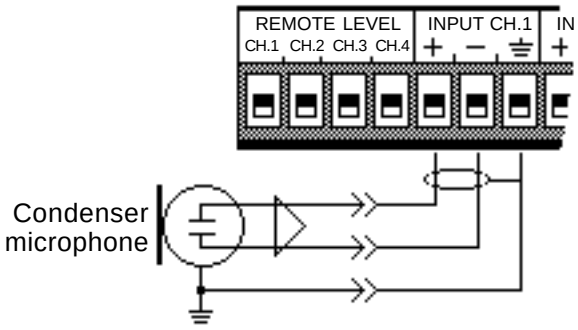
MIC/LINE - **MIC**
30 or 50 dB gain - **50 dB**
HP filter - **OPTIONAL**
Phantom Pwr - **OFF**

Figure 3

Connecting a Condenser Microphone *(Mixers and Input Expanders)*

Condenser microphones connect to the mixers and expanders via the Phoenix type connector located on the rear of the units. Pin one of the microphone is connected to the ground terminal on the connector. Pins two and three are connected to the plus and minus terminals. *The Mic/Line switch must be set to the Mic position. Due to the higher output level of condenser microphones, the 30/50dB gain*

switch should be set to the 30dB position. The setting of the High Pass Filter is optional. The Phantom Power Switch should be set to the ON position. A typical connection is shown in Figure 4 following:



INTERNAL SWITCH SETTINGS

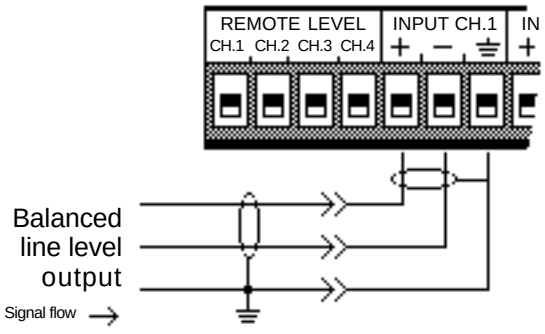
MIC/LINE - **MIC**
30 or 50 dB gain - **30 dB**
HP filter- **OPTIONAL**
Phantom Pwr - **ON**

Figure 4

Connecting a Balanced Line Level to a Channel Input *(Mixers and Input Expanders)*

Line level inputs are connected to the mixers and input expanders via the Phoenix type connector located on the rear of the units. If the shield is to be connected to the input of the unit, it must be connected to the ground terminal. If the shield is connected at the piece of equipment sending the signal, then the shield should *not* be connected at the mixer (or input expander). By connecting the shield at only one end of the cable, potential ground loop problems can be reduced.

The two wires carrying the signal are connected to the plus and minus terminals for that input channel. *The Mic/Line switch must be set to the Line position. The 30/50dB gain switch should be set to the 50dB position.* The setting of the High Pass Filter is optional. The Phantom Power Switch should be set to the OFF position. Figure 5 below details these connections:



INTERNAL SWITCH SETTINGS

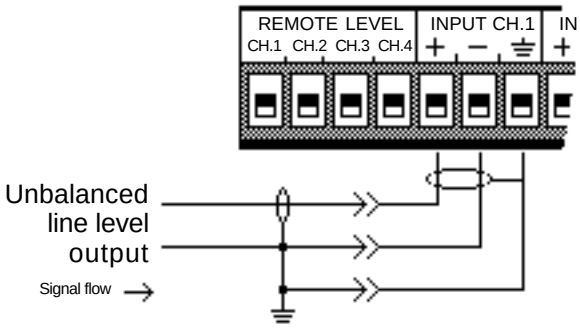
MIC/LINE - **LINE**
30 or 50 dB gain - **50dB**
HP filter- **OPTIONAL**
Phantom Pwr - **OFF**

Figure 5

Connecting an Unbalanced Line Level to an Input *(Mixers and Input Expanders)*

A typical unbalanced line has only two conductors. The signal flows in both conductors. The outer conductor also acts as a shield. Both of these two conductors must be connected to the Euro connector. The center conductor is connected to the plus terminal. The shielded conductor is connected to the minus terminal. Depending upon the installation, connection of the shielded conductor to the ground terminal may be needed in addition to connecting it to the minus terminal. If this additional connection reduces hum on the input, leave it in place. If the additional connection increases or causes hum on the input, remove it.

The Mic/Line switch must be set to the Line position. The 30/50dB gain switch should be set to the 50dB position. The setting of the High Pass Filter is optional. The Phantom Power Switch should be set to the OFF position, as shown in Figure 6 following:



INTERNAL SWITCH SETTINGS

MIC/LINE - **LINE**
30 or 50 dB gain - **50dB**
HP filter- **OPTIONAL**
Phantom Pwr - **OFF**

Figure 6

Master Outputs *(Mixers Only)*

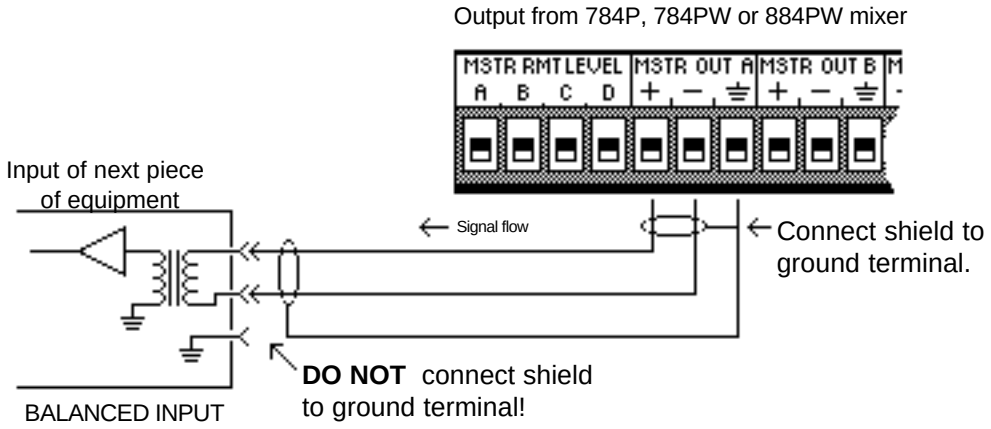
The mixers have four master outputs. They are referred to as master outputs A, B, C and D. When shipped from the factory, outputs A, B, C and D receive their signals from mix buses 1, 2, 3 and 4. *There is a jumper cable (78C-6) on the back of the mixer that connects the output of mix buses 1, 2, 3 and 4 to the master buses A, B, C and D. This 78C-6 cable must be in place for normal operation.* In this configuration, any of the eight inputs can be assigned to any of the master outputs utilizing mix buses 1-4 on the mixers.

On the 8X4 mixers, the A, B, C and D master buses could be connected to buses 5-8 instead of buses 1-4. Typically, buses 5-8 are used only when two or more mixers are "stacked" together to provide a sixteen input by eight output mixer. Please refer to pages 20 and 21 of this manual for more information on stacking mixers. In normal operation, the 78C-6 would be left connected as it was shipped from the factory.

Each master output is transformer balanced to help minimize problems with ground loops. The master outputs can drive a 600 Ω load to +18dBu.

Connecting a Mixer Output to a Balanced Line Level Input *(Mixers Only)*

The output of the mixer should be connected to the next piece of equipment in the audio signal chain via a two-conductor, shielded cable. Care should be taken to make sure the shield is connected at only one end of the cable. This will help minimize ground loop problems. This connection is demonstrated in Figure 7 following:



Good grounding practice dictates that on balanced lines, the shield be connected only at one end of the cable run.

Figure 7

Connecting the Output of the Mixer to an Unbalanced Line Level Input *(Mixers Only)*

The output of the mixer should be connected to the next piece of equipment in the signal chain with a two-conductor, shielded cable. *Care should be taken to make sure that the shield is connected only at the mixer end of the cable.* This will help minimize ground loop problems. Figure 8 below shows this connection in detail:

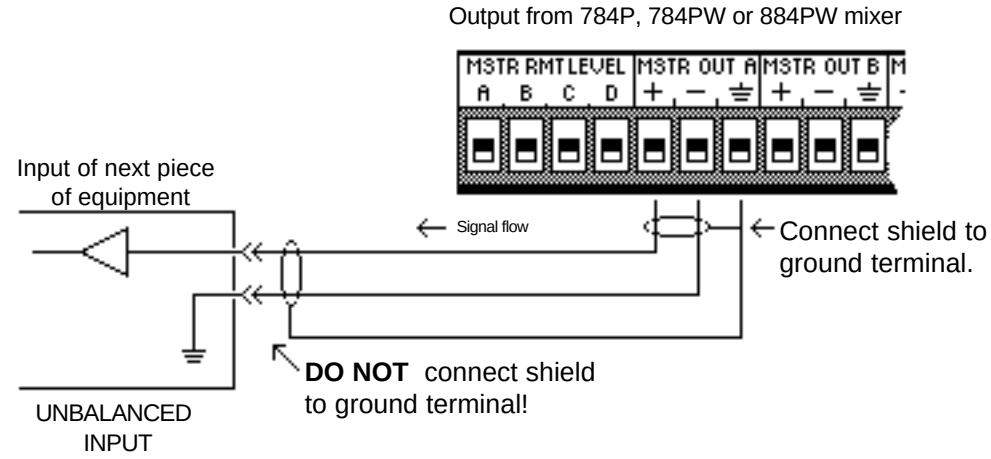


Figure 8

Tape Outputs *(Mixers Only)*

There is a Tape output associated with each of the four Master outputs. The Tape output carries the same program material as its Master output. The output level of the Tape output is *not affected* by the Master output level control. The level of the Tape output *is affected* by the individual channel levels feeding that Master output.

The actual level of the tape output is identical to the level of the Master output when the Master level is set to "full on." Any Gain Trim adjustment (limiting the maximum gain of the Master output) will cause a difference in level between the Master output at full gain, and the Tape output (See page 11 for more information on Gain Trim).

The Tape output is connected through a 604 Ω resistor to the Euro terminal block on the rear of the mixer. This allows Tape outputs to be connected in parallel with other Tape or Direct channel outputs. There will be a 6dB decrease in gain whenever two outputs are connected together.

The corresponding Tape outputs are automatically combined, as well as the Master outputs, whenever the room combining feature within a mixer is activated. There is no drop in output level when combining is done within the mixer. Figure 9 below is a block diagram of the Tape output:

Tape Output Block Diagram

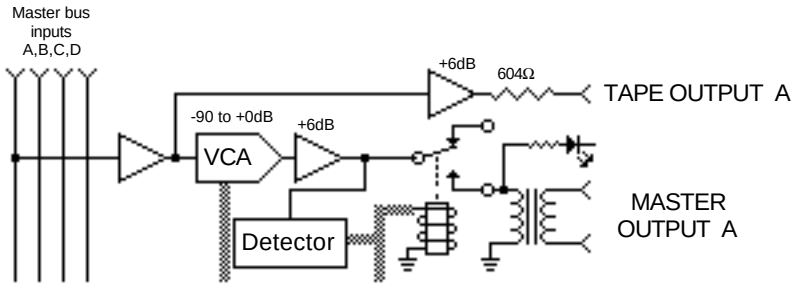


Figure 9

Direct Channel Outputs *(Mixers and Input Expanders)*

There is a Direct output for each individual input channel. This output carries program material from the individual input channel only. If an input channel is not assigned to a mix bus, the only output from that channel would be found at the Direct channel output.

This is a very useful feature. Any input channel, for example, could be used as a microphone to line preamplifier. The line level output would appear at the Direct channel output. A mixer could be configured as eight, 1 X 1 mixers with four sub mixes to the Master outputs. The Direct output is connected through a 604 Ω resistor to the Euro terminal block on the rear of the units. Direct outputs can

be connected in parallel with other Direct outputs when a mix of Direct outputs is required. There will, of course, be a 6dB drop in gain whenever the number of outputs being connected together is doubled. Figure 10 following is a block diagram of a Direct output:

Direct Output Block Diagram

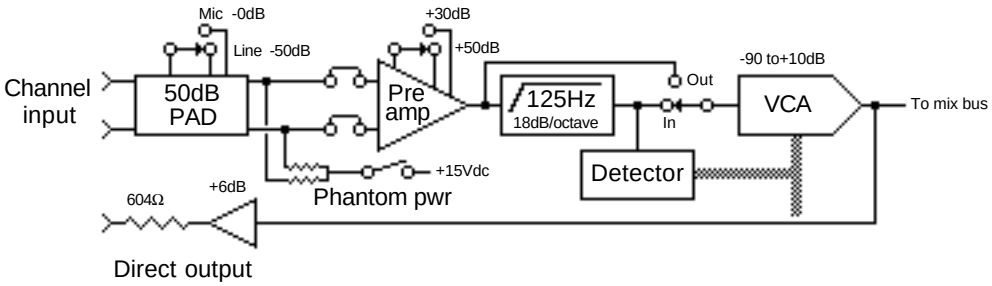


Figure 10

Level Controls *(Mixers and Input Expanders)*

There are eight level controls on the input expanders, one for each input channel. There are twelve level controls on the matrix mixers, eight for the channel inputs and four for the master outputs. All level controls function alike. There are three level functions associated with each control: Level, Range, and Trim (gain trim).

Level *(Mixers and Input Expanders)*

"Level" is the actual, operating, channel level and should be thought of as a common volume control that would be located on the front panel of any standard mixer. The channel Level can be adjusted using a remote control, or via the Audio Net+® software.

The Level control works as an in-line attenuator, and can attenuate signal from 0dB to -100dB. It is very helpful to think of "turning down" the volume as opposed to "turning up" the gain. The gain for the channel is basically set at input preamplifier stage. The Level control acts as an attenuator to the audio signal through that channel.

The Level control follows a standard audio taper as shown in Figure 11 on the following page. The human ear can normally discern changes of 2dB in level, so the beginning increments are in 2dB steps. There are 15 steps of 2dB, and then, as attenuation continues, the taper rolls off more rapidly, as Figure 11 on the following page demonstrates:

Taper Of Level Control

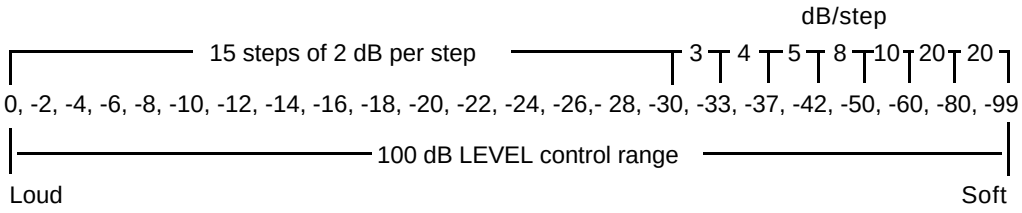


Figure 11

Range Control *(Mixers and Input Expanders)*

As previously noted, the Level control has an adjustment range of 100dB. The Range control allows the limiting, or restricting of the actual operating range of the Level control. The ability to limit the volume control range is very helpful in certain applications. Any type of meeting room with a volume control located in the room can benefit from range limitation. For example, if the volume control range is limited to 20dB, then the sound system can never be turned all the way off. There will always be some sound from the system when somebody talks into or taps on a microphone. They will say to themselves "the sound system is working, but we need to turn up the level." They will not say, "The sound system is not working," but will usually look for the volume control to turn up the level. This can reduce the number of requests by users for someone to come and turn on the sound system.

The Range control limits the range of the volume control. Instead of the full 100dB range, a much smaller range can be selected. The range is referenced from full on, or zero dB of attenuation, to a set number of dB below full on. The range is set by entering the Range menu and setting the control range in dB. The range limit can be set to any point on the Level audio taper.

The Range control has a special feature. It is called "Range Plus Off." A limited control range can be set with the final step in that range being full off, or full attenuation. This gives the volume control a limited range, but still allows for a full off selection.

Trim (Gain Trim) *(Mixers and Input Expanders)*

Trim (gain trim) is a very powerful feature of the matrix mixers and input expanders. It allows the setting of the maximum level, or gain, in a particular channel. The maximum gain through the channel can be limited, so as to prevent feedback from occurring when a channel Level control is turned all the way up. The Gain Trim can also be used to increase maximum gain in a channel. Keep in mind that the majority of gain in the channel is set in the input preamplifier stage. The Trim adjustment allows up to another 10dB of gain to be added to the individual channel. The Trim will also allow up to 100dB of attenuation to permanently limit the maximum gain available to a channel.

Figure 12 on the following page shows the relationship between the Level control, Range control and the Trim control.

Note: The normal full range of the VCA is 100dB (0 to -99). If less than the full range is desired (perhaps only 20dB of range may be wanted), the Range control can be used to limit that range. If less than the full available gain is desired, the Trim control can be used to limit the maximum gain available. If the maximum gain available is limited by the Trim control, the full 0 to -99dB of attenuation is still available below the maximum level that has been set by the Trim control, unless, of course, that range of attenuation has been limited by using the Range control.

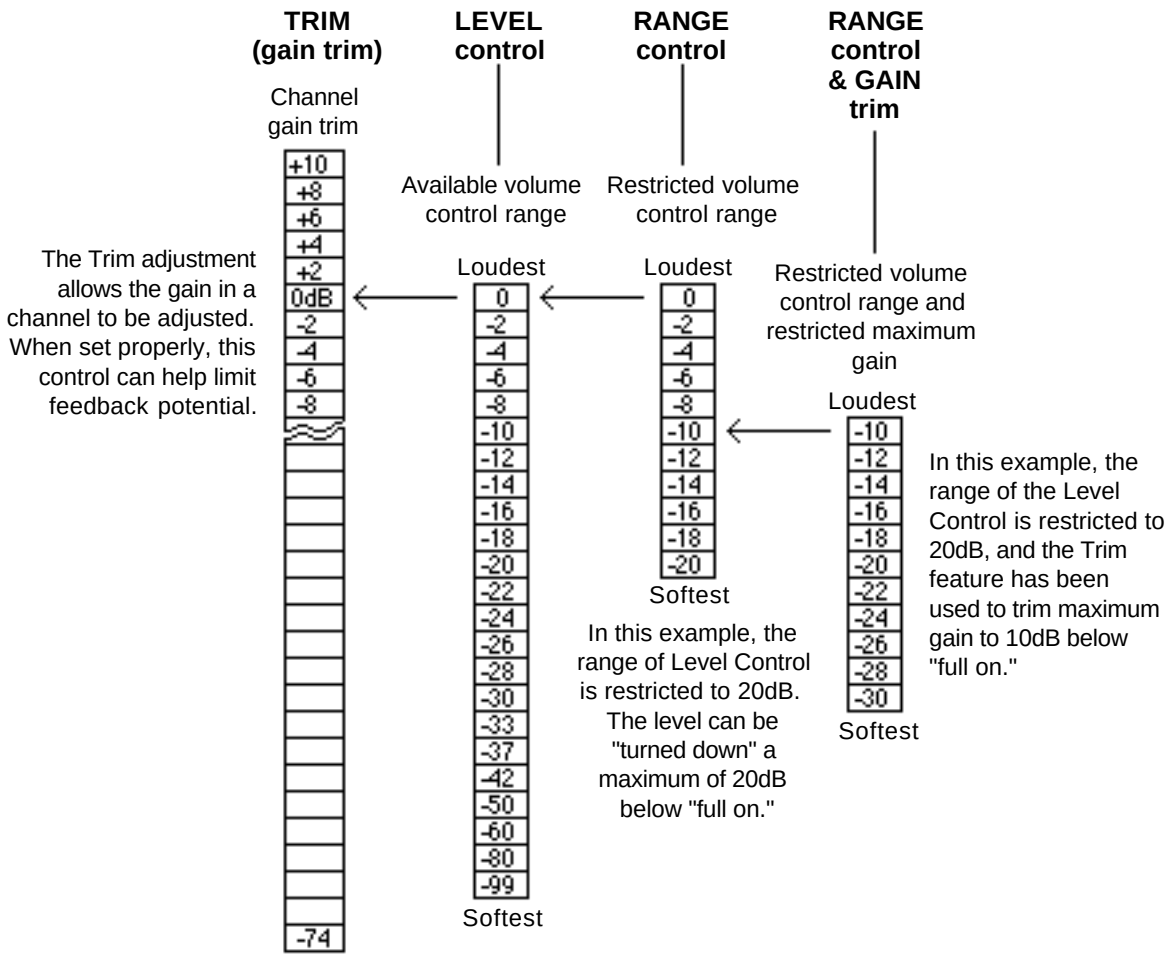


Figure 12

Remote Volume Controls *(Mixers and Input Expanders)*

Please note: *Before any remote control can become active, a mixer or input expander must be programmed to accept input from a specific type of remote control.*

The eight input channel volume controls of mixers and input expanders, and the four master output volume controls of mixers may all be remoted. Two different types of remote controls can be used: a rotary style using a potentiometer, and an up/down push button style. *(The RM-1 and RM-2 up/down push button controls look like pots, not up/down buttons, to the mixers and input expanders.)* All power required by the remotes is supplied by the mixer. *The RMT (remote) voltage supplied by the mixer is current limited and must be used only for the remotes. The RMT voltage cannot be used to power any other type of external device.*

The control range of all remotes follows the programming of the mixer or input expander for channel and master volume controls. The programming can be used to limit the range of the volume controls. A limited range plus "OFF" can also be programmed. For more information on volume control programming refer to the section of the manual titled "Level Controls" on page 10.

Using a Potentiometer Remote *(Mixers and Input Expanders)*

Please note: *Before any remote control can become active, a mixer or input expander must be programmed to accept input from the remote - the software RP Button for that channel must be pressed.*

The Ivie RP-1 (Remote Pot-1) remote volume control is designed for use with the Ivie matrix mixers. It is a potentiometer mounted on single-gang, black anodized, wall plate. The Ivie RP-1S has a DPDT switch mounted on the plate in addition to the potentiometer. The switch is typically used for background music on/off.

A standard 10k Ω, linear taper, potentiometer is connected as illustrated in Figure 13 following. *Note that the potentiometer is connected across the RMT voltage and ground.* The wiper of the potentiometer is connected to the Remote Level channel terminal. As the wiper gets closer to ground, the volume level increases. As the wiper gets closer to RMT+, the volume level decreases. *The RMT voltage cannot be used to power any other type of external device.*

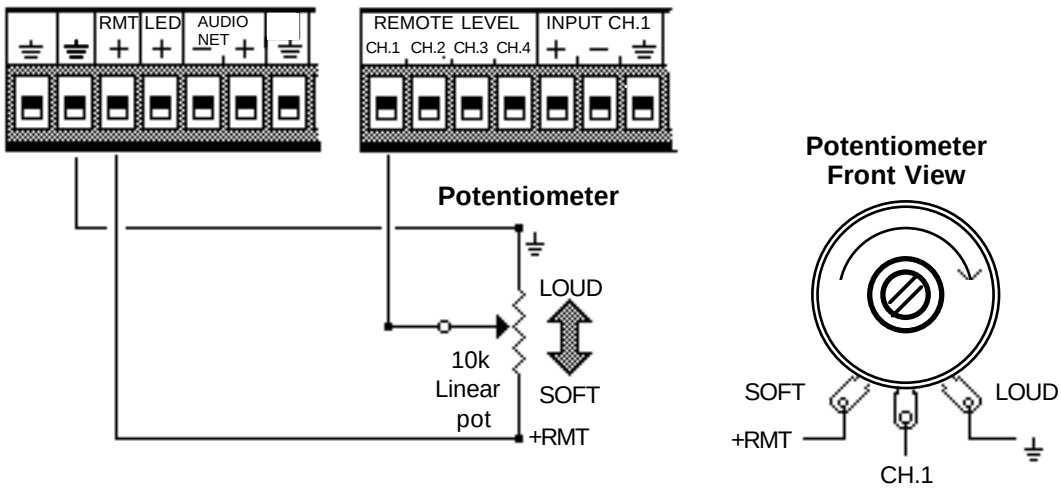


Figure 13

When a limited volume range has been programmed for a channel, the number of steps in that range is divided into the number of degrees of rotation in the potentiometer. Thus, a limited range will be expanded around the rotation of the potentiometer.

Potentiometer style remotes cannot be paralleled together to provide remote control for one channel from several different locations. Software volume control is disabled when a potentiometer style remote is connected and activated by the programming.

Once a matrix mixer or input expander has been programmed for a remote potentiometer, the unit will "look" for a pot connected to the remote level terminal. If it does not detect the pot, it will adjust the level as dictated by the software settings. When a pot is connected, the unit detects the connection and sets the level as dictated by the pot. The unit will automatically switch between the software setting and the remote pot setting. If the pot is disconnected, the mixer will automatically switch back to the software setting.

In some applications, this automatic switching is very advantageous. Day-to-day operations could proceed with the software volume settings. For special events, a plug-in remote control could be used which would override the normal settings. At the conclusion of the event, unplugging the remote would restore the software volume settings.

Ivie manufactures portable remote control panels that are compatible with Ivie matrix mixers. The RMA-10 Remote Control can control 10 channels, and comes with 10 feet of cable and a connector. It is supplied with a matching wall plate and connector assembly. If only 6 channels are needed, an Ivie RMA-6 may be used, which also comes matching wall plate and connector assembly.

Using Up/Down Push Button Remotes *(Mixers and Input Expanders)*

Please note: *Before any remote control can become active a matrix mixer or input expander must be programmed to accept input from a push button remote - the software RB Button must be pressed.*

Ivie manufactures two types of push button remotes that can be used with the matrix mixers and input expanders. The first set of push button remotes consists of the RM-1 and RM-2 microprocessor based remote controls. The RM-1 controls a single channel, while the RM-2 has two sets of push buttons to control two channels. The push button remotes “look like” pots to the matrix mixers, and are connected the same way that potentiometer remotes would be connected. *Since the RM-1 and the RM-2 look like pots to a matrix mixer, the mixer must have its software “Remote Pot” (RP) control enabled in order for these remote controls to work.* The RM-1 and RM-2 have on-board microprocessors, so they must be fed power from the mixers as well as be connected to the remote control ports. They have up/down buttons and an LED bar graph to provide visual indication of level. They are themselves programmable for maximum level, minimum level, and “wake up,” or turn on level. Once programmed for maximum and minimum levels, the LED bar graph will automatically be spread over that range, that is, minimum level will light one LED on the bar graph and maximum level will light all the LED’s on the bar graph. These controls can be factory programmed to track one another automatically in room combining applications.

The second type push button remote is the Ivie RB-1. It is a simple, less expensive, up/down push button. It consists of a single-gang, black anodized, wall plate containing "Up" and "Down" push

buttons, three resistors and a terminal block for wiring connections. Each push of a button increments or decrements the volume level by one step. Several can be paralleled together for remote level control capability from more than one location. *In order for the RB-1 to function, a matrix mixer or input expander must have its software "Remote Button" control enabled.* If tracking of RB-1's is needed for room combining applications, the special RB-1 ROM option is required, since, in the case of RB-1's, the tracking is a function of the mixer or input expander, not the RB-1. Please contact the factory for further information.

Making Custom Up/Down Buttons *(Mixers and Input Expanders)*

As illustrated in Figure 14 below, two push buttons and three 3.3k Ω, 1/4 watt resistors, are needed for a push button remote. The resistors are connected across RMT voltage and ground, forming a voltage divider. All power required by the remotes is supplied by the mixer. *The RMT voltage supplied by the mixer is current limited and must be used only for the remotes. The RMT voltage cannot be used to power any other type of external device.*

Connection of Push Button Remote Volume Controls

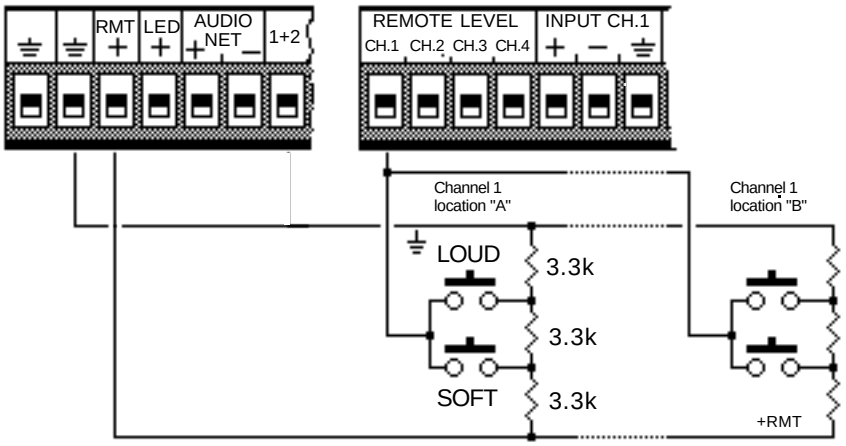


Figure 14

Hardware Activation of Presets *(Mixers and Input Expanders)*

Presets can be activated remotely on all the matrix mixers and input expanders via a switch connected to the appropriate terminals on the rear panel. Presets can also be activated via Audio Net+®. Activation via Audio Net+® is covered in Section II of this manual.

In order for a preset to be activated remotely via a contact closure (switch or relay), connections across two terminals on the rear of the mixer or input expander are required. There is a terminal for each of the

presets. When a preset terminal is connected to ground, it activates that particular preset. Euro terminal connections for presets are shown in Figures 15 and 16 below:

A preset is activated when a preset terminal is grounded. Presets may be programmed to be momentary or alternate (toggle). Please refer to the section titled "Programming Presets"

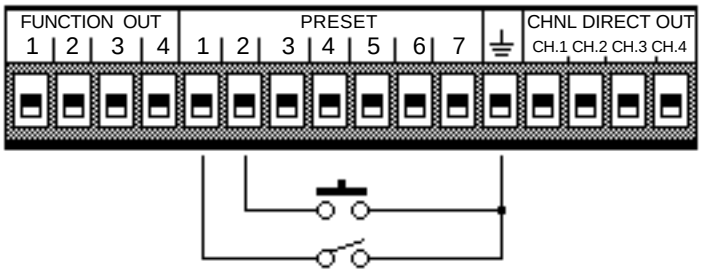


Figure 15

Figure 16 below shows how to activate more than one preset simultaneously, using a single switch. As illustrated, isolation diodes should be used:

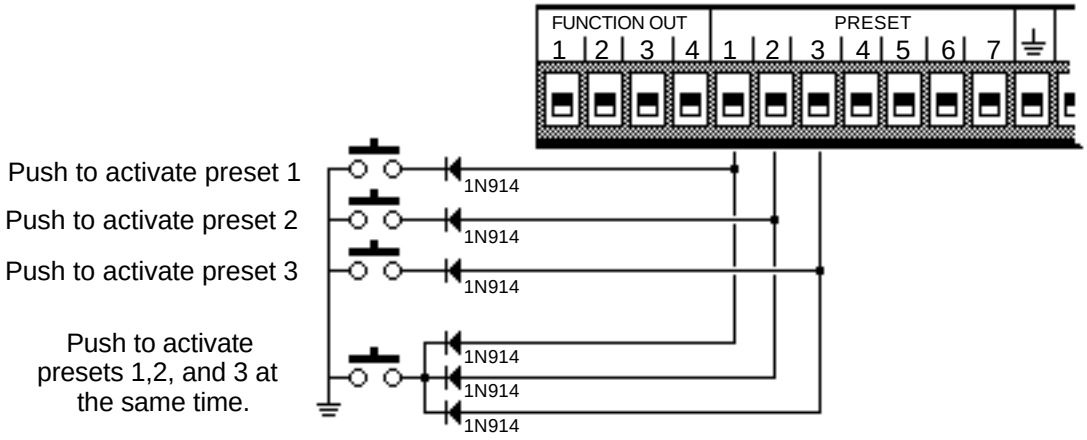


Figure 16

Activating Presets on Two Units at the Same Time *(Mixers and Input Expanders)*

Presets on two different mixers and/or input expanders may be activated at the same time. Diodes must be used to isolate one mixer from the other. A standard 1N914 diode can be used, as shown in Figure 17 on the following page:

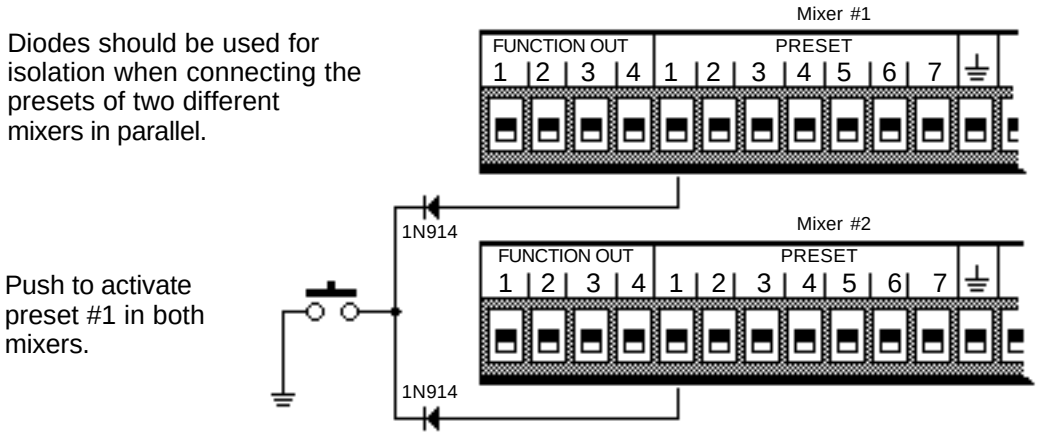


Figure 17

Combining Mix Buses *(Used For Room Combining)*

All of the mixers and input expanders have a built-in room combining feature. Any, or all of the eight mix buses may be combined together. Combining is handled internally in these units, under microprocessor supervision. The units do not actually combine outputs, but accomplish room combining by assigning input channels to the involved outputs. For example: In a setup where input channels 1 and 2 are assigned to mix bus 1 feeding output A, and input channels 3 and 4 are assigned to mix bus 2 feeding output B, if the 1+2 combining terminal were activated (this terminal combines mix buses 1 & 2, not inputs 1 & 2), inputs 1 and 2 would be fed to both outputs A and B. Likewise, inputs 3 and 4 would be fed to both outputs A and B.

There are 5 combining terminals located on the rear of a mixer or input expander. The combining feature is activated by connecting one or more of these terminals to a ground terminal. Figure 18 following depicts 2 combining switches connected to the rear panel Euro connector. If both switches are closed at the same time, all three rooms (A+B+C) are combined together. These two switches allow four different combinations. 1) All rooms independent. 2) Rooms A and B combined. 3) Rooms B and C combined. 4) All rooms combined (A and B and C). *(When required, some terminals normally used for preset activation can be reassigned to provide additional room combining terminals. See Appendix IV, page 71, titled* ***Customized Room Combining****.)*

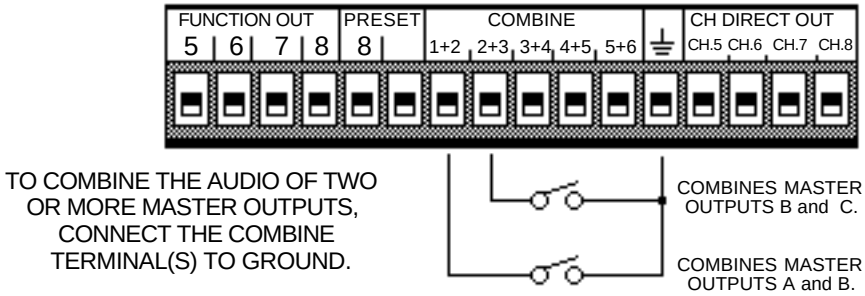


Figure 18

Function Output *("Channel Active" Output on 880+ and 884+)*

Each Function Output on the matrix mixers and input expanders can be used for controlling external devices, typically LED's. A Function Output is not an audio output. It is a low voltage, DC, solid-state switch. This switch is activated by the internal microprocessor. It can be used to light an LED, or with an external (installer provided) power supply, could be used to control an external relay.

One purpose of the Function Outputs on the *780+* and *784+* is to provide a status indicator for the Preset and Combine switches. This is very useful when duplicate Preset switches are located at several different locations. The Function Output can be used to illuminate an LED that is part of the Preset switch. On the *880+* and the *884+*, the Function Output normally provides "channel active" status, although it is possible to program the *880+* and *884+* so that the Function Outputs latch with Preset and Combine activation, instead of latching upon channel activation.

The activation of the Function Outputs on the *780+* and *784+* follows the activation of the Preset and/or Combine switches. As shipped from the factory, the Function Output switches are activated whenever their associated Preset is activated. Function Output number 1 is paired with Preset number 1. Function Output number 2 is paired with Preset number 2, and so on.

On the *880+* and *884+*, Function Output number 1 is a channel active indicator for input number 1, Function Output 2 is channel active indication for input 2, and so on. *(When required, Function Outputs can be reprogrammed to follow other events such as preset or room combine activation. See page 36 titled* ***Function Output Assign****.)*

The Function Output is the open collector type. The collector is normally open until the Function Output is activated at which time the collector is brought to ground potential. Please understand that *NO* voltage is supplied by this switch. It is *ONLY* a switch to ground. Since this is a transistor, polarity, maximum voltage and current limits must be observed when connecting external devices.

Maximum current through the switch should not exceed 200 mA. Voltage across the switch should not exceed 30 VDC. Figure 19 below shows an LED connected to the Function Output. The mixer has a built in power supply voltage for use with LED's, and *it must not be used for any other purpose than LED's.* Figure 19 also shows the proper connection of an LED:

Two LEDs are connected between the FUNCTION out and LED+ terminals. They are activated according to the programming of the mixer. The LEDs can be placed at different locations.

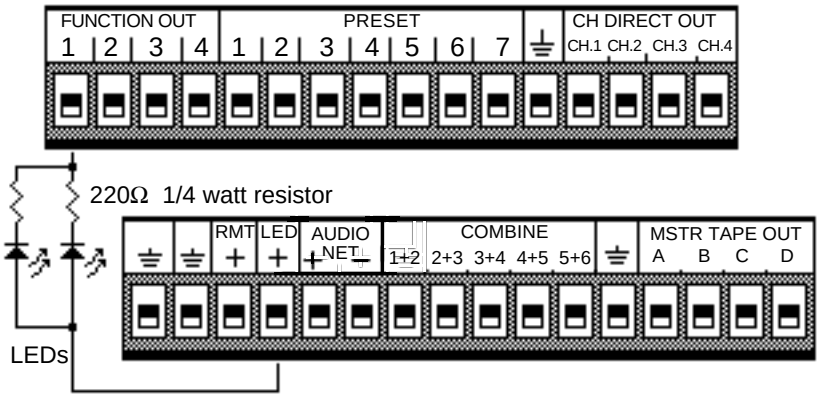


Figure 19

Master Mute *(Mixers Only)*

Certain types of sound system applications dictate the need for the sound system to be muted or "shut down" by some type of supervisory system. Typically, the sound system must be muted whenever a life safety system is activated. The Master Mute terminals on the rear of the each of the matrix mixers provide this function. *For convenience in manufacturing, there are Master Mute terminals on the 780+ and 880+ Input Expanders, but they are inoperative. They have no outputs to mute.*

On matrix mixers, master outputs A, B, C, and D are muted whenever the MASTER MUTE terminals are connected together. *There is no audio output on any of the master outputs when the Master Mute terminals are shorted together.*

There is a direct, "hardware" connection between the Master Mute terminals and the output relays that mute the audio. The master mute terminals override all other commands to the relays. No software commands can override the Master Mute terminals. Please note that the connections are made via a screw terminal block and not via a plug-in style connector. This provides additional security that the Master Mute function will be available when needed.

Since the Master Mute function is activated when the two terminals are connected together, this may be accomplished with a switch or a relay contact closure. *No voltage should be applied to these two terminals.* Multiple mixers may be activated by wiring the Master Mute terminals of all mixers in parallel as shown in Figure 20 below:

Master Mute Connections

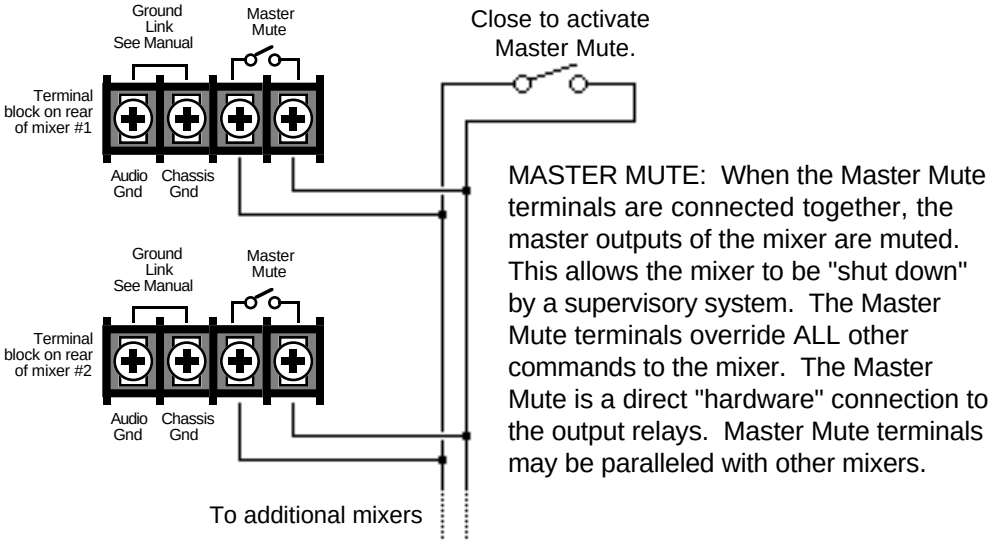


Figure 20

Mix Bus Inputs And Outputs *(Mixers and Input Expanders)*

There are eight mix buses inside the matrix mixers and input expanders. Any input channel can be assigned to any or all of the eight mix buses. The channel to mix bus assignment is made using a Windows® based computer and Ivie ANSW+ software.

There are mix bus input and output (I/O) connectors on the rear of the mixers. There is a set of I/O connectors for mix buses 1, 2, 3 and 4 and a set for mix buses 5, 6, 7 and 8. These I/O connectors allow the multiple mixers to be "ganged" or "stacked" together. For example, two mixers can be stacked together to make a sixteen input by eight output mixer.

The output of the mix buses must be connected to the Master Bus input connector in order for any audio signal to appear at the output of the mixer.

On mixers shipped from the factory, mix buses 1, 2, 3 and 4 are connected to Master Outputs A, B, C and D. A cable is connected between the Mix bus 1, 2, 3 and 4 Output connector and the Master Bus A, B, C and D Input connector. The cable supplied with the mixer is an Ivie part number 78C-6. This cable ***MUST*** be in place if the mixer is to have any output. Please note that on the rear panel of the mixers, there is a white line drawn between the Mix bus 1, 2, 3 and 4 Output connector and the Master bus A, B, C and D Input connector. This line indicates that these two connectors should have an Ivie 78C-6 cable connected between them. Figure 21 following shows the factory set configuration for a single, 8 X 4 mixer:

Mix Bus to Master Bus Connections

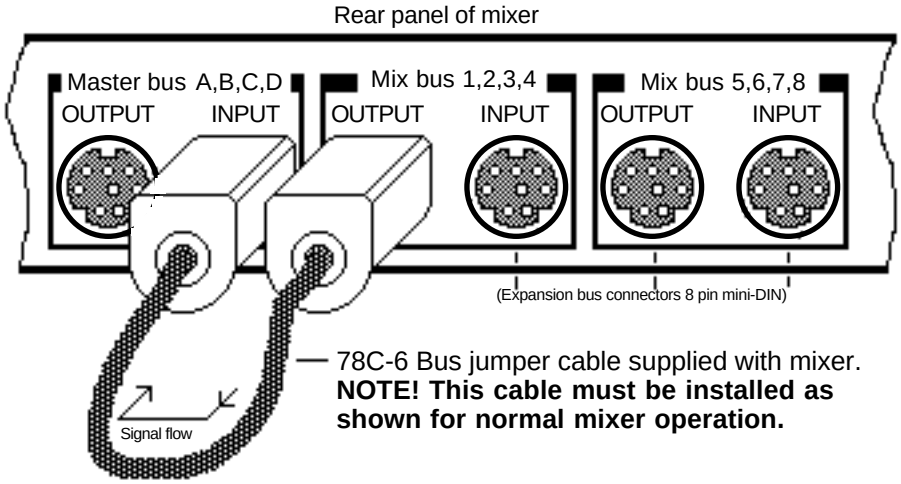


Figure 21

Although there are eight mix buses inside the mixers, only buses 1, 2, 3 and 4 are utilized in a single 8 X 4 mixer application. This is due to the fact that there are only four master outputs available to access the eight mix buses. So what good are eight mix buses if only four can be used? The answer becomes apparent when two 8 X 4 mixers are "stacked" together. The four outputs of the first mixer are connected to mix buses 1, 2, 3 and 4. The outputs of the second mixer are connected to mix buses 5, 6, 7 and 8. Two mixers connected together in this configuration provide sixteen inputs assignable to any

or all eight outputs. It requires two 8 X 4 mixers to take advantage of all eight mix buses. Additional mixers and/or input expanders can be "stacked" to provide even more inputs, but the number of output channels is limited to eight because there are only eight internal mix buses.

There are six, 8-pin, mini-Din connectors on the rear panel of the 8 X 4 mixers and four mini-Din connectors on the back of the input expanders. These connectors allow the mixers and input expanders to be stacked together in various combinations. Take a moment to examine Figure 22 below. As can be seen, there are two 8 X 4 mixers connected together via mix bus I/O connectors. Mix buses 1, 2, 3 and 4, and 5, 6, 7 and 8 are brought from mixer number one to mixer number two, via two 78C-6 cables.

The outputs of mix buses 1, 2, 3 and 4 are then connected to Master buses A, B, C and D of mixer number one via 78C-6 cable. Mix buses 5, 6, 7 and 8 are connected to Master buses A, B, C and D of mixer number two. Figure 22 following details these connections:

Two Mixers "Stacked Together" To Make a 16 X 8 Mixer

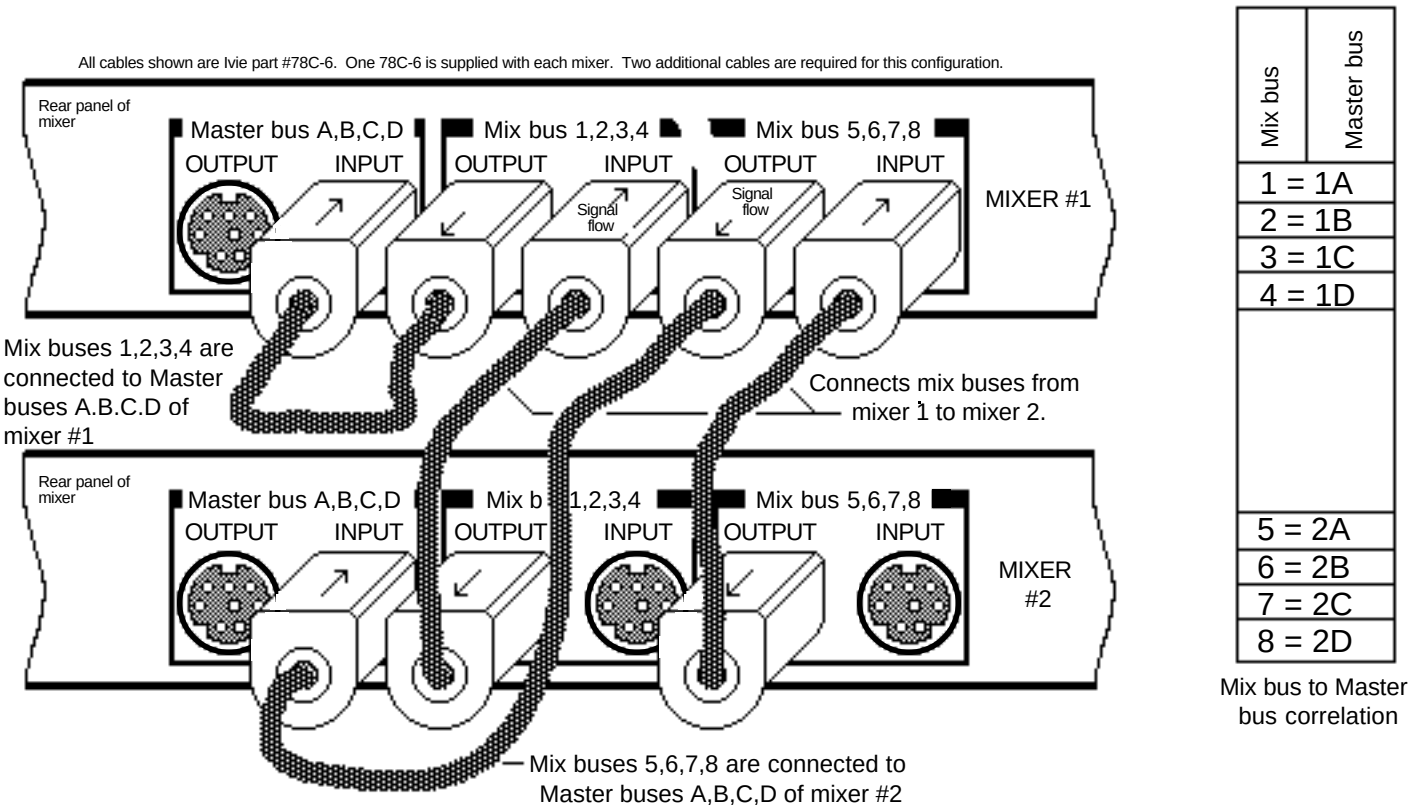


Figure 22

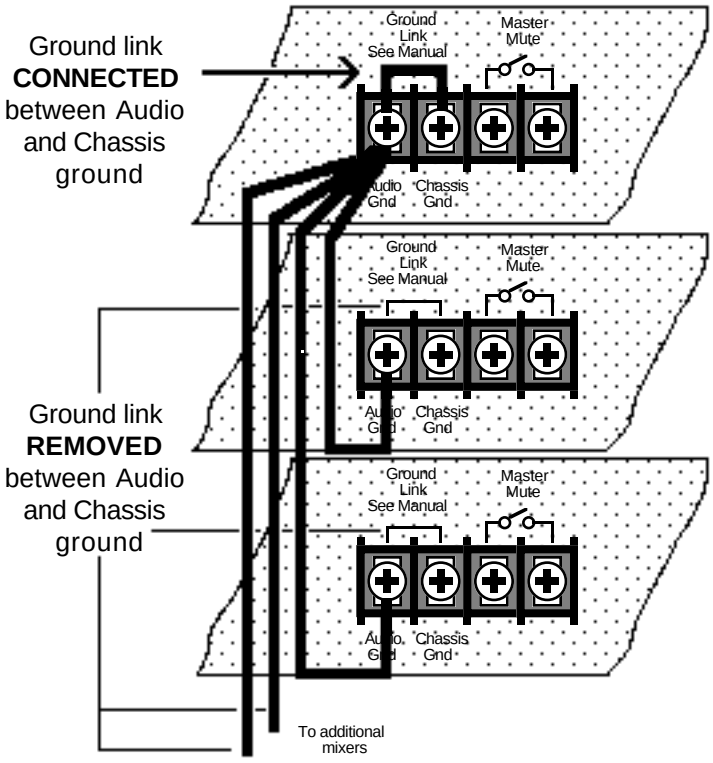
Note: *For more information on stacking multiple mixers, consult Appendix I, page 44 of this manual entitled,* ***How Inputs are Routed to Mix Buses and Mix Buses are connected to Master Outputs.***

Ground Link Connections *(Mixers and Input Expanders)*

The four position terminal block located on the rear of mixers and input expanders has two terminals to assist in properly grounding the units. The two terminals are labeled "Audio Gnd" and "Chassis Gnd."

There is a jumper connected between the "Audio Gnd" and "Chassis Gnd" terminals. This jumper should remain in place unless more than one unit is being connected together. If more than one unit is being used, they should be wired together in a "star ground" configuration. Figure 23 below shows this method of grounding. "Star grounding" greatly reduces noise problems due to ground loops between units.

Please note that the first mixer in this "star ground" scheme has its jumper installed between the audio and chassis ground terminals. All other units have their jumpers removed. A large gauge wire, 16AWG or greater, is connected between each successive unit's Audio Gnd terminal and the Ground terminals on the first mixer. *This separate wire run all the way from each unit back to the first mixer is important. Do not daisy chain from one Audio Gnd terminal to another and then to the first mixer.* When connecting to the first mixer, the Chassis Gnd terminal can be used if the Audio Gnd terminal gets too congested.



Recommended Grounding Procedure For Multiple Mixers

This grounding procedure will help minimize noise due to ground loops between mixers.

1) Make certain that the first mixer in the system has a jumper between Audio and Chassis ground terminals on the rear of the mixer.

2) Remove the Audio to Chassis ground jumper on all other mixers.

3) Use 16 AWG wire or greater and connect the Audio Ground of each successive mixer to the Audio Ground of the first mixer. There should be a"home run" wire from each mixer to the first mixer. Do not "daisy chain" them

Figure 23

Section II: Using Audio Net+® Software

Introduction

The ANSW+ (Audio Net+® Software) is compatible with all Audio Net+® products from Ivie, and is a Windows® 95/98 application requiring a Windows® based computer - the faster, the better. When a software command is issued to an Audio Net+® product via the ANSW+ software, the command is executed by the product instantaneously, after which it sends its updated status back to be displayed by the software. While this occurs instantaneously, the software will take a moment to display the updated status. The slower the computer, the longer this update will take to be displayed.

A hard disk drive is required for overall operation of the system. A mouse or similar pointing device is required to use this program. There are over 100 controls on the software screen, thus necessitating the convenience of a mouse. A serial port is required for the connection to a mixer, either directly through the RS-232 port on the mixer front panel (an RS-232 port is not yet available on all Audio Net+® products), or through a 78-232A cable that connects the computer to a mixer or input expander through the Audio Net+® RS-485 port on the mixer front panel. ***The serial port must be unattached from any other device (it cannot have any other software tying it up)*** - mouse driver software, or modem software running in the background, for example. If the port is <u>not</u> unattached when trying to run ANSW+ software, the ANSW+ software will detect this state and display a error message explaining that the port is not available. The only way to solve a "Cannot Find 78-232A" error message is to choose another port, or deactivate whatever software is tying up the port.

Please note: The computer used must have either one serial port and one mouse port, or two serial ports - one for the mouse and one for the mixer connection

There are several software control screens in the ANSW+ software. One screen is for programming matrix mixers and input expanders. The software recognizes the unit identity (*784+*, *884+*, *780+*, *880+*, *784PW, 884PW etc*.) and displays the proper screen for that unit. There is a separate screen for programming the 728PW and 730PW equalizers and another screen for programming the 802A Automixer. Switching between screens is a simple, pull-down menu function.

In addition to the ANSW+ software, the ANSW+ software disk contains another program called 485MON. This program is an RS-485 monitoring program that is a tremendous help is programming external control devices such as touch screens. An Audio Net+® Programming Manual is available from Ivie - either by request to the factory, or by downloading from Ivie's web site at www.ivie.com. This manual is intended for those programming external controls (such as AMX® or Crestron® touch panels) and provides both the HEX and ASCII code for controlling all functions of the Audio Net+® products. Use of the 485MON program when programming is very beneficial and highly recommended.

RS-232 Connections Between Mixers and Computers *(Mixers Only)*

On their front panels, the *784+* and *884+* have both an RS-485 Audio Net+® input and a RS-232 Audio Net+® input. As discussed above, connecting a computer to the RS-485 Audio Net+® input requires a 78-232A adaptor. However, a direct connection between a computer RS-232 port and the RS-232 Audio Net+® input is possible using a "straight RS-232" cable (a null-modem cable will not work). Only three of the cable conductors are required for this connection. Figure 24 on the following

page details the pin-out and signal flow of this cable connection:

RS-232 Connections Between a Computer and an Ivie Matrix Mixer

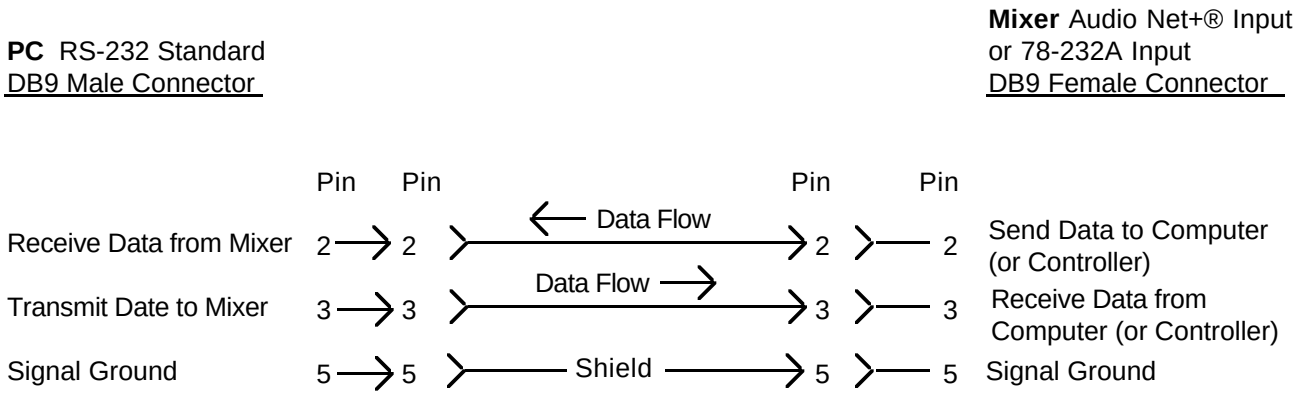


Figure 24

Front Panel Audio Net+® LED's *(Mixers Only)*

The *784+* and *884+* have both an RS-485 Audio Net+® input and a RS-232 Audio Net+® input, as previously discussed. When plugging directly into the front panel RS-232 input, the mixer protocol is still Audio Net+®, so a 78-232A (RS-232 to RS-485 adaptor) has been built into the mixers. The communications LED's of the built-in 78-232A are located on the front panel near the RS-232, DB9 input (LED's are also provided on a 78-232A adaptor). These LED's provide the following information: Yellow LED - Indicates data generated from the unit and sent to the Audio Net® bus. Red LED - Indicates date received from a PC or other controller such as AMX or Crestron touch panels which is sent to the bus. Green LED - Indicates activity on the bus.

The yellow LED, which is labeled ACK, illuminates whenever its unit generates data to the Audio Net® bus. For example, the yellow ACK LED will flash when sending data to the Audio Net® software in response to a query. If the unit has been placed on line, the yellow ACK LED will also flash as the unit generates data to the bus due to a gating event, remote level change event, preset event etc.

Important Note: If two or more units have their yellow ACK LED's flashing simultaneously, the units have identical addresses. This is a problem than must be fixed. Each unit must have its own unique address.

Important Note: If yellow ACK LED's are illuminating from time to time and the Ivie Audio Net® software is not controlling the system, this should be of concern. Some or all of the units have not received a global "sleep" command (@Z) and they are still on line. This can lead to collisions on the network which can cause all sorts of strange things. It can corrupt addresses or cause other problems.* To solve this problem, connect a PC to the system and poll the system via the Audio Net® software. Then Exit the software. Upon exiting, the software issues the global *@Z sleep command. To avoid the problem in the future, always exit the software before disconnecting the PC from the network.

The red and green LED's are both labeled DATA. The red LED flashes to indicate data is being received via the RS-232 DB9 front panel connector and sent to the Audio Net® bus. The red LED, then, provides *positive* indication that a computer is successfully sending data to the mixer through a

properly wired cable. Since it is an indicator of the DB9, RS-232 signal pathway, the red LED will *not* illuminate when data is sent to a mixer via the RS-485 port.

The green LED will flash whenever there is *any* activity on the Audio Net® bus. It will flash in conjunction with the yellow LED. It will even flash when data is sent to the Audio Net® bus via the RS-485 port because it is a bus activity indicator. There is one condition when it does not flash as data activity occurs on the bus. If data is received on the bus from the RS-232, DB9 connector, the green LED associated with *that unit* will not flash when the red LED flashes. The green LED's of all other units on the network will flash.

Question: Is the mixer successfully receiving data from the computer? *Answer:* If the computer is connected to the RS-232 port and the red LED flashes with each poll - connection is successful. If the computer is connected to the RS-232 port and the green LED flashes with each poll - connection is successful. *If no red or green LED flashing occurs, a proper connection has not been made.*

Software Conventions

Strong effort has been made in writing the software to maintain the "look and feel" of common, Windows® based software, so that the ANSW+ software will be very intuitive to experienced Windows® users. Whenever possible, standard conventions are used, such as pull-down menus, pop-up windows, and use of the tab key to advance from one heading to the next when entering data.

Getting Started:

Software Installation

ANSW+ software is shipped with all Audio Net+® products from Ivie. An install shield is provided with the software to help install it properly. Additionally, the latest version of the software is always available for downloading from Ivie's web site at www.ivie.com.

*The **standard** ANSW+ software is for Windows® 95/98 operating systems*. To install the software, insert the ANSW+ software disk into the floppy drive and follow these instructions:

1. Click the **Start** button in the task bar.
2. Select **Run**.
3. Type **a:\SETUP**, then click **OK** (where "a" is the letter denoting the floppy drive).
4. Follow the on-screen prompts to fully install the ANSW+ software.

Selecting a Com Port

During the installation process previously described, two shortcuts will be created, with their associated icons, by which the software can be started. One shortcut will start the program with communication on Com 1 and the other will start the program with communication on Com 2. Currently, the software allows communication only on Com 1 or Com 2. A laptop computer will typically use com port 1.

Computer To Mixer Connections *(Mixers and Input Expanders)*

Audio Net+® *(78-232A Optional)*

Before a matrix mixer can be controlled, it must be connected to a computer. This can be accomplished using the Ivie 78-232A adaptor, or, with some Audio Net+® products, through the RS-232 port on their front panel. If there is no RS-232 port on the product front panel, a 78-232A adaptor must be used. This adaptor connects between the computer RS-232 serial communications port (Com port) and the Audio Net+® connector on the front panel of the mixer. The RS-485 Audio Net+® connection can also be made at the Audio Net+® plus and minus terminals located on the rear of the mixer. Audio Net+® units can be "daisy chained" together using the rear panel Audio Net+® connectors such that they are all on the network and access to one unit allows access to all units. *The ANSW+ software looks for a 78-232A during the network polling process. The 78-232A can either be an external one, or the one that is built-in behind the front panel on units that have a DB9 connector on their front panel. If it can't find a 78-232A, an error message appears. There is no point in continuing until communication to the 78-232A is established.* Messages to and from Audio Net+® computer controlled matrix mixers and input expanders occur over a multi-dropped, RS-485, Half Duplex network. The network operates asynchronously at 9600 Baud, 8 Bit, No Parity with 1 Stop bit. Figure 25 below details a typical Audio Net+® system configuration when using a 78-232A RS-232 to RS-485 adaptor. *(The older model 78-232 was for DOS software only and will not work with the ANSW+ software).* The 78-232A is connected to the computer using the supplied adaptors as required. The stereo plug end of the 78-232A is plugged into the Audio Net+® connector on the front panel of the mixer or input expander, as shown in Figure 25 below:

Typical Audio Net+® System Configuration

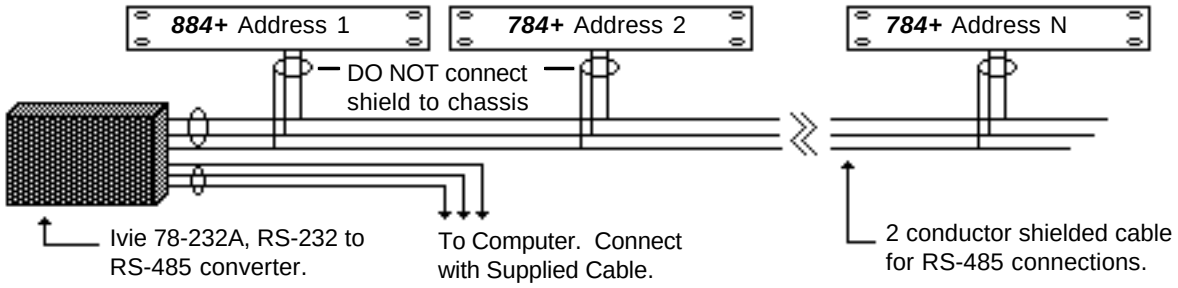


Figure 25

Assigning a Unit Number *(Mixers and Input Expanders)*

The default unit assignment number for mixers is unit number 1 and for input expanders, number 2. Sometimes the factory may assign sequential unit numbers to units when more than one unit is ordered. ***Important Note:*** *When several units are on the network simultaneously, each must have a different unit address. When assigning unit numbers, units may not be tied together on the network, or all will happily accept the same unit number - obviously causing problems.* Assigning unit numbers is covered in detail on page 38 of this manual in the section titled **Set Unit Address**.

The ANSW+® Software Control Screen *(Mixers and Input Expanders)*

The software control screen is intuitive and easy to operate. Figure 26 below is an excerpt from the ANSW+ software control screen. It shows two inputs and two outputs, with their associated Level Adjustment Buttons, Remote Control Buttons etc. It also shows two rows of the Bus Assignment Switches which are used to assign an input to a specific mix bus - *any input may be assigned to any or all mix buses.* Figure 26 also shows the Automix Assign Button and the Software Gating Switch (*880+* and *884+* only). The software recognizes which model Audio Net+® product it is addressing, and if a unit other than an *880+* or *884+* is being addressed, the Software Gating Switches and the Automix Assign Buttons will disappear. Notice that the mix buses which feed the outputs are graphically represented - mix bus 1 feeds output A, and mix bus 2 feeds the next output, output B. A full ANSW+ software control screen would show mix bus 3 feeding output C and mix bus 4 feeding output D. Mix buses 5, 6, 7 and 8 are not feeding an output, and are normally unused in single mixer applications. However, when two 8 X 4 mixers are tied together, the second mixer would have its outputs fed by mix buses 5 through 8. The eight mix buses allow a true 16 X 8 matrix when two mixers are ganged together. When viewing the software control screen for the second mixer, the software would show its outputs being fed by mix buses 5 through 8:

The Controls and Windows of the ANSW+ Software Control Screen

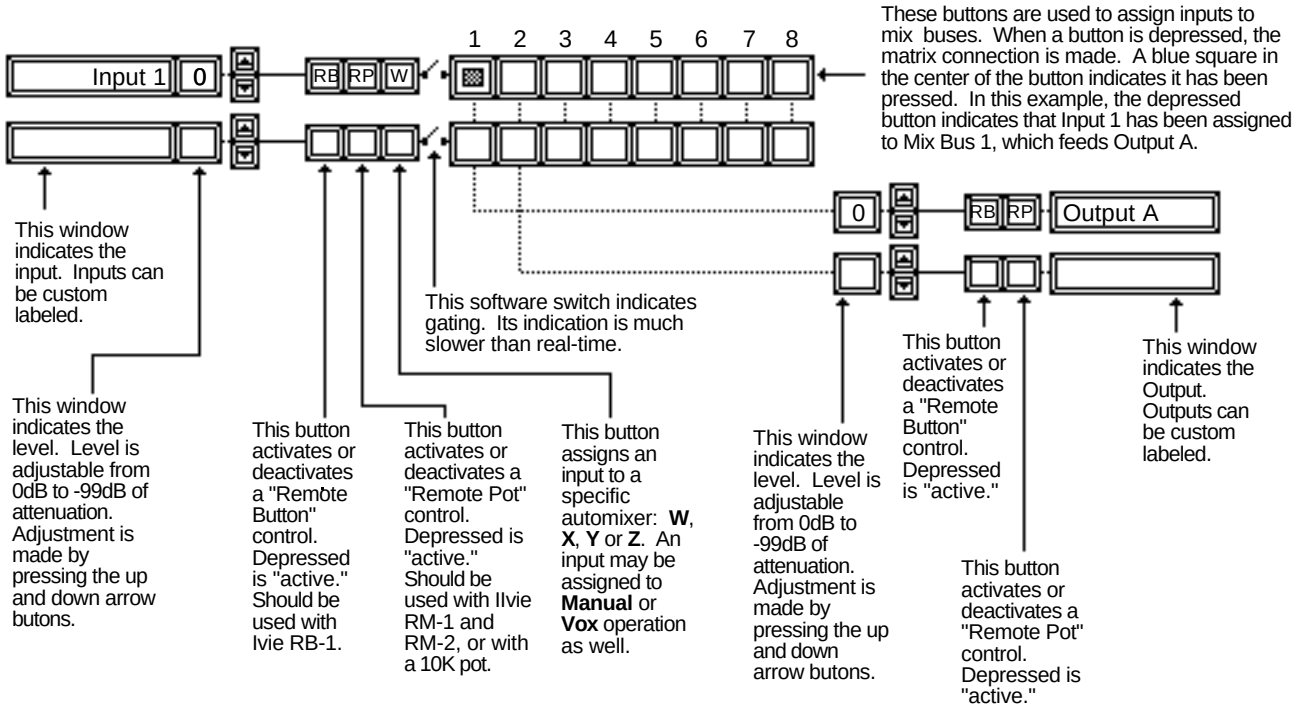


Figure 26

While the ANSW+ software control screen is intuitive, a couple of “not-so-obvious” features should be pointed out. Following is an outline of these features.

Custom Labeling Inputs and Outputs *(Mixers and Input Expanders)*

As indicated in Figure 25, inputs and outputs can be custom labeled. To do this, simply place the cursor arrow over an Input or Output Window and double click. A pop up label form will appear which allows both the inputs and outputs to be labeled. Double clicking over either the Input or Output Window will bring up the same pop up label form. Using the keyboard, type in the desired labels and then click OK to close the pop up label form. *Input expanders use the same pop up label form, even though they have no outputs to label.*

RB and RP Remote Control Buttons *(Mixers and Input Expanders)*

This set of buttons on both inputs and outputs provides for activation or deactivation of remote controls. When a Remote Control Button is pressed, its associated remote control is active. If the Remote Control Button is not pressed, the associated remote control will not be active even though it may be properly wired up. This is a very useful feature. In paging applications, for example, a remote control could be overridden during a page so the page could come through at the proper level. After paging, the remote control would become active again. This can all be done as a part of a programmed preset. Programming presets is covered in detail in **Appendix II** of this manual titled **Programming Presets**, beginning on page 48.

The RB and RP Remote Control Buttons cannot both be pressed simultaneously. Their function is to both indicate the type of remote control being used and to activate it or deactivate it. The RB button should be used with the Ivie RB-1 and RB-2 remote controls. The RP button should be used with either a 10k pot as a remote control, or with Ivie RM-1 and RM-2 up/down button remote controls. Even though the RM-1 and the RM-2 remote controls have up and down button with an associated LED bar graph, they "look like" pots to mixers and input expanders, so the RP remote control button should be used with them.

A computer running Ivie ANSW+ software, a remote touch panel, and remote button (Ivie RB-1) controls can work together at the same time. They are momentary commands that increment or decrement the level of the channel to which they are assigned. Basically, they are connected in parallel through the hardware and software.

The remote pot (RP) control cannot be used simultaneously with any other control. The Remote Pot will automatically override the computer screen. When a remote pot is activated and detected, the background color of the level windows on the ANSW+ software control screen will change from gray to blue indicating that the level is being remotely controlled. *Caution: If a remote pot is connected to a level control and the RB button is pressed, the results will be the same as if someone were continually pressing one of the up/down buttons.*

Automix Assign Button *(880+ and 884+ Only)*

The Automix Assign button is functional only with *880+* Input Expanders and *884+* Automatic Matrix

Mixers. Pressing the button will cause it to toggle through the available selections. There are four automixers inside the *880+* and *884+*, automixers W, X, Y and Z. Assigning inputs to automixer X, for example, will cause those inputs to be associated together for gating, NOM and threshold - in other words, it will associate them together as an automixer separate from automixers W, Y and Z. In addition to being able to be assigned to a particular automixer, an input could be assigned to VOX operation (indicated by a V on the Automix Assign button), or manual operation. Manual "Off" can be selected (indicated by an M on the Automix Assign button with the button not depressed), or Manual "On" can be selected (M on a depressed Automix Assign button). When an input is assigned to Manual "On," the Software Gating Switch will be closed.

The operation of the Automix Assign Button is detailed in Figure 27 below:

The Seven Positions of the Automix Assign Button

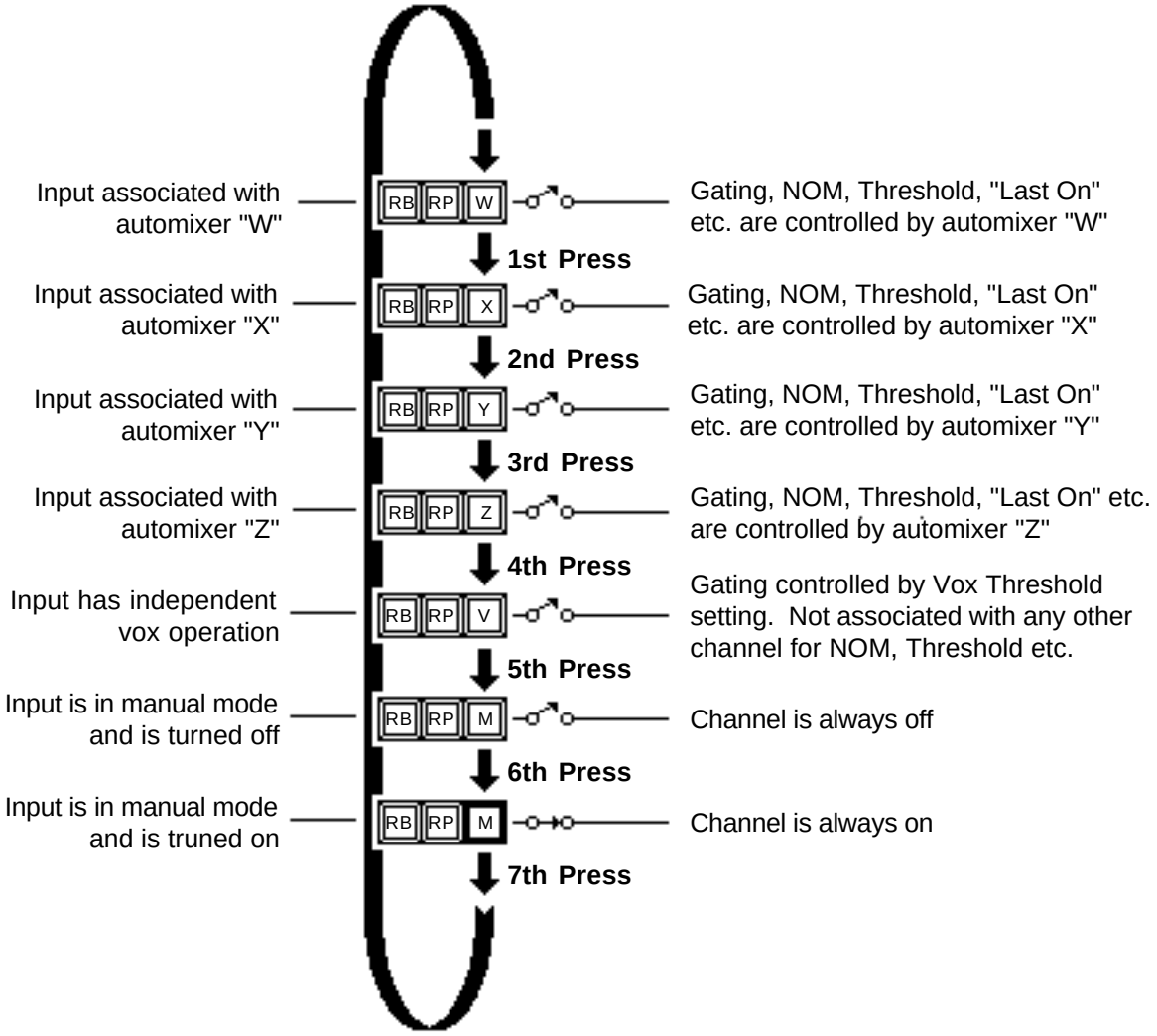


Figure 27

The *880+* Input Expander and *884+* Automatic Matrix Mixer can actually be up to four separate automixers in one box. There are eight inputs, and on the *884+* there are four outputs. Ivie has provided separate automixing capability such that any input can be associated with any other input and

any output in an automixer configuration. Up to four of these automixers can be configured in one box. These four automixers are labeled W, X, Y, and Z.

When inputs 1 and 2, for example, are assigned to the "W" automixer, those two inputs are associated with one another. The microprocessor "looks" only at signals from these two inputs when deciding which of these two inputs should be gated on. When both of these inputs are gated on, the NOM (Number of Open Microphones) attenuator is activated. If inputs 3 and 4 are assigned to automixer "X" while inputs 1 and 2 remain assigned to automixer "W," the result is two separate, independent automixers in operation at the same time. If inputs 1 and 2 are assigned via the matrix to output "A," the audio output of automixer "W" will appear on output A. The W, X, Y and Z designation is just a way of showing (labeling), on screen, the inputs that are associated with each other for automatic mixing. By contrast, if a single eight by one automixer is required, all inputs would be assigned to the same automixer, automixer "W," for example. Via the matrix, all eight inputs would be assigned to mix bus "1" (mix bus "1" feeds output "A"). In this same manner, custom size automixers can be configured with a single *880+* or *884+*. The limitation is the maximum of eight inputs and four outputs.

880+'s and *884+*'s can, of course, be ganged with other *880+*'s and *884+*'s to provide more inputs. *However, automixers cannot "cross boxes."* That is, the automatic mixing functions, gating, NOM and threshold, remain inside their respective boxes. If, for example, one *884+* and one *880+* are ganged together to provide a 16 X 1 mixer configuration, there will not be one, large 16 X 1 automixer, there will actually be two 8 input automixers mixed into a single output. All the inputs in the *880+* will be associated together for gating, NOM, threshold etc., but they will not be associated with the inputs of the *884+* for those functions.

Important Note: If two automixers in the same box become tied together by a room combine command, they will not become one automixer. Use of a preset in conjunction with a room combine is required to make them become one automixer.

Audio Net+® Window *(Mixers and Input Expanders)*

Figure 28 below details the Audio Net+® Window found in the ANSW+ Software Control Screen:

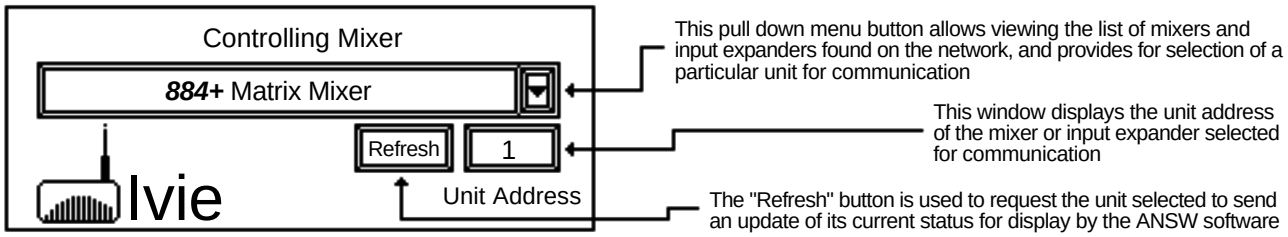


Figure 28

"Controlling Mixer" Pull Down Menu *(Mixers and Input Expanders)*

The "Controlling Mixer" Pull-Down Menu displays the type and the unit address of the mixer or input

expander currently selected for communication. Clicking on the pull-down arrow causes a pull-down menu to appear which lists all the mixers and input expanders on the network. "Dragging" the cursor down the listing allows selection of any mixer or input expander on the network for communication.

Refresh Button *(Mixers and Input Expanders)*

The Refresh Button sends a query to the mixer or input expander selected requesting that it answer back with its current status. That status is then displayed by the ANSW+ software control screen. The Refresh Button would normally be used when a question arises as to whether the unit under control and the ANSW+ software are communicating. Pressing the Refresh Button provides a quick confirmation of proper communication.

Level, Trim and Range Buttons *(Mixers and Input Expanders)*

The Level, Trim and Range Buttons are located just above the Audio Net+® window on the ANSW+ software control screen, as Figure 29 below shows:

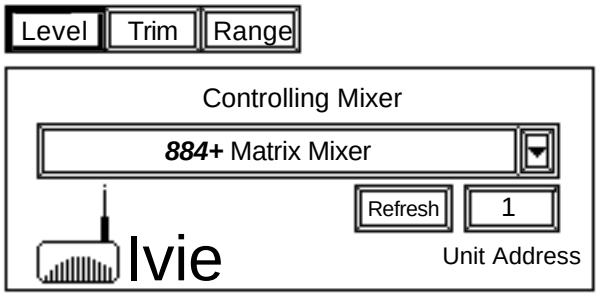


Figure 29

The Level Button has been pressed in Figure 29, which is the default configuration. Pressing the Trim Button would select the Trim function, and likewise, pressing the Range Button would select that function. A complete explanation of the functions of the Level, Trim (gain trim) and Range is given on pages 10, 11 and 12 in **Section I** of this manual.

The ANSW+ software control screen changes colors from its normal, blue background when either the Trim or Range Buttons are depressed. The area behind the Input and Output Windows will change from blue to white, with red, diagonal strips. This is to provide quick, visual indication that Level has not been selected, and either Trim or Range has been selected. The up and down cursor arrow buttons next to the Input and Output Windows (see Figure 26 on page 27) are used to adjust Level, Trim and Range.

Range Plus Off: It may sometimes be desirable to limit the range of a remote control, but still give it full-off capability. The Range Plus Off feature does just that. After the range has been set to the desired limits, activating the Range Plus Off feature will add one more step to the remote control, -99dB, or full-off. Figure 30 on the following page details activation of the Range Plus Off Feature:

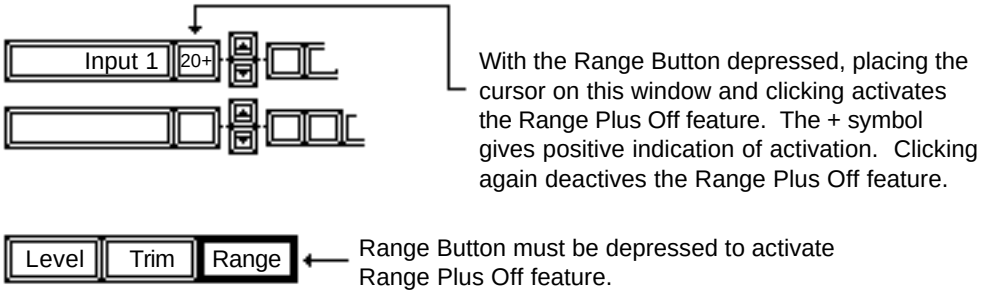


Figure 30

Range example: If the Range Button has been depressed, and the number in the Level Window for an input reads -20, then the range for that input channel has been set to 20dB. If the Level Button is then pushed and the level of the input run up and down, it will be found that the level will go all the way up to 0dB (no attenuation, or maximum gain), but in turning the level down, it will not go below -20 (the range that has been set). *This is true whether the software is used to run the level up and down, or whether a remote control is used* This is a very nice way to limit the range of a remote control. Of course, if the Range Plus Off feature has been activated, the level will follow it normal taper down to -20dB, and the next step will drop it to -99dB, or full-off.

Trim example: If the Trim Button has been depressed, and the number in the Level Window for an input reads -10, then the maximum gain available for that input channel has been reduced by 10dB. If the Level Button is then pushed and the level of the input run up and down, it will be found that the level will go all the way up to 0dB (no attenuation, or maximum gain), and all the way down to -99dB (unless, of course, the Level function has been used to limit how far down the channel can be adjusted). However, when the channel is turned up to 0dB, or maximum gain, that maximum gain will be 10dB down. *This is true whether the software is used to run the level up and down, or whether a remote control is used.* Both the range of a remote control and its level when it is "full on" can be adjusted using the Range and Trim functions.

Preset and Room Combining Buttons *(Mixers and Input Expanders)*

The top right area of the ANSW+ software control screen contains the Preset and Room Combining programming buttons as shown in Figure 31 on the following page:

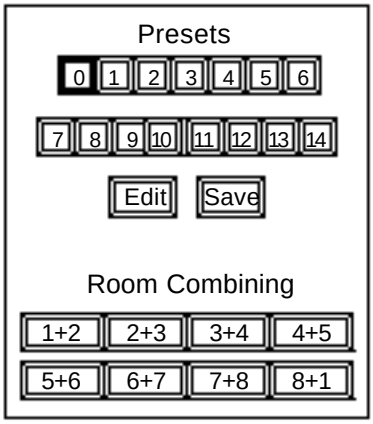


Figure 31

There are 15 presets (0 - 14) available in the mixers and input expanders. Presets 1 through 14 can be activated or deactivated by the ANSW+ software, a contact closure (14 sets of contacts are available on the rear panel of the mixers and input expanders), or a control code string via the Audio Net+® Network. Preset 0, on the other hand, is always active. Once it is programmed, it cannot be turned off. Preset 0 is the "base" configuration of the mixer or input expander, and reflects the way the unit normally operates. All the other presets "overlay" Preset 0 and "amend" or change it as they become activated. This means that Preset 0 should be programmed first. Several or all presets may be active simultaneously and will live together quite happily unless there is a conflict in instructions. If, for example, one active preset instructs "turn on background music," and another active preset gives the conflicting instruction "turn off background music," the higher numbered preset will win the battle. The presets have a built-in priority corresponding to their number. Preset 3, for example, has priority over presets 2, 1 and 0. Likewise, preset 5 has priority over presets 4, 3, 2, 1 and 0. Obviously, preset 14 has the highest priority of all. Having these built-in levels of priority is a real benefit when programming, allowing maximum flexibility. It is incumbent on the programmer to use higher numbered presets for those instructions that are intended to take priority. **Appendix II** of this manual titled **Programming Presets**, beginning on page 48, contains detailed information on this subject.

The Room Combine Buttons on the ANSW+ software control screen are used to effect room combining. The first button, for example, combines mix bus 1 and mix bus 2 (see Figure 28 on the preceding page). Normally, Output A is fed by mix bus 1, and Output B is fed by mix bus 2. Pressing the mix bus 1+2 button, therefore, would combine Outputs A and B. *(Note: The mix buses are not actually "combined," or "mixed together." Combining is effected by copying the bus assignments from one bus to the other, and vise versa.)* When two 4 output mixers are combined to provide 8 outputs, mix buses 5, 6, 7 and 8 also feed mixer outputs. Room combining for these mix buses is handled in exactly the same manner, as shown by the labeling on the buttons.

The room combine instruction given when a Room Combine Button is pressed is a "smart" instruction. The mixer or input expander looks to see if there are any presets active that affect either mix bus. If there are, as part of the room combine instruction, any presets affecting one bus are copied over to affect the other bus in precisely the same manner. This provides exactly what would be expected in a room combining situation. If, for example, background music is "on" in a "stand alone" room, and this room

is combined with another room where background music is not on, the combining instruction would send background to both rooms, just as would be expected.

Room combining, just like preset activation, can be done via the ANSW+ software, contact closures or code instructions over the Audio Net+® network. For a complete discussion of custom room combining, please refer to **Appendix IV** of this manual titled **Custom Room Combining** on page 71.

ANSW+ Software Pull-Down Menus (Mixers and Input Expanders)

Many of the functions of the ANSW+ software have been concentrated in pull-down menus. Many of these menus have pop-us screens associated with them. These pop-up screens are quite self explanatory and intuitive in their operation.

File Pull-Down Menu *(Mixers and Input Expanders)*

The File pull-down menu contains five menu items. These include: Load mixer settings FROM disk, Save mixer settings TO disk, Poll, Print Matrix Screen and Exit.

Load *(File Pull-Down Menu)*

It would be most helpful to read the information below titled **Save** ***(File Pull-Down Menu)*** before continuing with these Load instructions...

Selecting Load from the File pull-down menu facilitates uploading mixer settings (presets, etc.) from disk storage to a mixer or input expander. Once a mixer or input expander has been fully programmed, it is prudent to download its programming to disk. When this is done, it is a simple matter to reprogram the unit from disk, should this become necessary.

The Load utility provides a pop-up window with instructions for uploading data. The List files of type: pull-down window allows selection of the type of files being considered for uploading. Ivie Audio Net+® units will not accept files for uploading that are not specifically intended for them. A *784+*, for example, will not accept *884+* files, nor will *784+* units accept 784PW files, and *vice versa*. The same follows for *884+* versus 884PW files etc. Plus (+) unit and non plus unit files are not compatible.

When uploading to an *884+*, for example, files with the extension .84P must be selected. After .84P files have been selected using the List files of type: pull-down window, all files of that type will be displayed. Of course, the other two pull-down windows of the Load utility pop-up window must be used to select the drive and the folder where files have been stored. Once the files have been displayed, the desired file can be chosen by double clicking on the appropriate file name, or by typing its name in the File name window. Clicking the OK bar will then upload the file to the unit under control.

Save *(File Pull-Down Menu)*

The Save utility can be used to save mixer settings to disk. Once a mixer's programming has been saved to disk, it is easy to reprogram the unit from disk, should this become necessary, using the Load utility.

The Save utility provides a pop-up window with instructions for downloading data. First, a file name must be given to the file by typing the name into the File name window. The default file name is the unit address (Unit_3, for example) and unit type (.74P, for example, is the unit type designator for a *784+*). The unit address portion of the file number can be changed as desired to better identify a unit. Up to eight characters can be used. *However, the unit type designator (file extension) must remain as part of the file name. Files for a 784+, for example, must be saved as .74P files because only .74P files can be uploaded into a 784+.* A desirable file name for a *784+* matrix mixer might be BallrmA.74P, identifying the mixer as the Ballroom A mixer, and as a *784+*. The file name is an 8 + 3 format: up to eight characters of choice plus the required, three character, unit type designator.

The unit type designators (file extensions) for other products are as follows: .784 for the 784PW, .884 for the 884PW, .780 for the 780PW, .880 for the 880PW, .74P for the *784+*, .84P for the *884+*, .70P for the *780+* and .80P for the *880+*. The "Save file as type:" pull-down window lists the Ivie Audio Net+® products with their unit type designators for use in naming files.

Pull down windows are also provided to allow selection of a drive and a folder for file storage. Once the file has been given a proper name and a location for storage has been selected, clicking the OK bar will save the file.

Note: *Units will accept files for uploading only of their specific type. An 884+ file cannot be uploaded to a 784+, for example, nor can files from plus (+) units be uploaded to non-plus units.*

Poll *(File Pull-Down Menu)*

Selecting Poll from the File pull-down menu commands the software to poll the network and list all the Audio Net+® units it finds on the network. A pop-up window appears while polling is in progress, and a pull-down window is provided which lists all the units on the network. After this pop-up window has been closed, a listing of Audio Net units on the network is still available from the pull-down menu in the Audio Net+® window of the ANSW+ software (See Figure 28 on page 30).

Print Matrix Mixer Screen *(File Pull-Down Menu)*

This is simply a "screen dump" print command, and can be used to document programming by printing a hard copy of the software control screen after programming has been completed. Of course, the mixer or input expander must be "on line" in order to "see" it configuration on screen.

Exit *(File Pull-Down Menu)*

This command exits the ANSW+ software program. *It is very important to exit the software before unplugging from the network.* Exiting the software issues a "quieting" command to all units on the network to prevent them from generating unnecessary network traffic.

Edit Pull-Down Menu *(Mixers and Input Expanders)*

The Edit pull-down menu contains six menu items. These include: Switch Type, Function Out Assign, Room Combine Set-up, Mixer Name...Enter/Edit Lock Unit with Password, UNlock Unit with Password Set Unit Address, and ReAssign Unit Address.

Switch Type *(Edit Pull-Down Menu)*

There are 19 contact closures available on the mixers and input expanders for use in actuating presets and room combines. The usual configuration is 14 closures for presets and 5 for room combines. Additional closures may be "stolen" from presets to use for room combines (see **Appendix IV** titled **Custom Room Combining** on page 71), but the total number of closures available is still 19. The switches providing the contact closure can be either momentary or alternate action switches, but the unit needs to know what kind of switch is being used. When shipped from the factory, all units are set for momentary switches.

Momentary type switches activate and maintain a preset or room combination *only* as long as the switch closure is maintained. The alternate action switch is a latching action. A temporary closure of the switch will cause the preset or room combination to occur and be maintained (latched) until another temporary closure of the switch occurs, unlatching the preset.

Important note: *The terms Momentary and Alternate refer to how the mixer acts upon sensing a switch contact closure, and not to the type of switch actually used. Perhaps it is best to think of Momentary as Non latching and Alternate as Latching.*

A switch that has a momentary action (the contact closure is maintained only while the switch is physically pressed) could be used with either a Momentary or Alternate setting on the mixer. For example: a momentary action switch with the Momentary setting in a mixer or input expander is typically used with a paging microphone. The page will be active only while the switch is pressed.

A momentary switch used with the Alternate setting in the mixer or input expander would work well for a background music switch. The first press of the switch would turn the music on. The second press would turn the music off. An advantage of the Alternate action is that multiple switches can be wired in parallel and activated from several locations.

Selecting the Switch Type utility provides a pop-up window with labeled push button controls for all 14 presets and the first 5 room combines. Using these push buttons, individual selection of either momentary or alternate action is possible.

Caution: Paralleling an alternate action switch to two or more units for use in preset or room combine activation will not insure tracking of the units.

Function Out Assign *(Edit Pull-Down Menu)*

Ivie matrix mixers and input expanders have eight function outputs which are electronic switches (open collectors to ground). On the *784+* mixer and the *780+* input expander, function outputs are set at the factory to activate in conjunction with presets 1 through 8 (activating preset 1 activates function output 1, preset 2 activates function output 2, etc.). The *880+* input expander and *884+* automatic matrix mixer function outputs are set at the factory to activate on channel gating. At times it may be desirable to have the function outputs activate in conjunction with the room combining switches. The Function Out

Assign utility allows function outputs to be reassigned.

There are three possible choices:

1. Function outputs latch with channel gating (possible with the *880+* and *884+* only).

2. Function outputs latch with preset activation.

3. Function outputs latch with a combination of presets and room combines. Presets 1, 2 and 3 latch function outputs 1, 2 and 3 respectively, and room combines 1+2, 2+3, 3+4, 4+5 and 5+6 latch function outputs 4, 5, 6, 7 and 8 respectively.

The pop-up window of the Function Out Assign utility allows a quick choice of one of the above three options. *Note: The front panel LED's of the 880+ and 884+ will continue to reflect gating (channel active) regardless of the function output option chosen.*

Note: The Room Combine Set-Up screen under the Edit pull down menu allows custom programming of the function outputs. This is a very powerful feature. Consult ***Appendix IV*** *of this manual titled* ***Custom Room Combining*** *on page 71*
.

Room Combine Set-up *(Edit Pull-Down Menu)*

A extremely powerful feature of the Audio Net+® matrix mixers and input expanders is their built-in ability to handle complex room combining tasks. While the control screen of the Audio Net+® directly supports serial room combining, the Room Combine Set-up utility allows the mixers and input expanders to be programmed to handle complex room combining requirements. Up to eight rooms with up to eleven combining nodes can be combined in various, custom configurations. Rooms do not need to be adjacent to be combined.

The pop-up window of the Room Combine Set-up utility provides programming examples and a dozen templates that can be selected if the template matches the room layout required. Selecting a template automatically programs the unit for that particular room layout.

In addition, the Room Combine Set-up utility allows custom configuration of the Function Outputs to follow specific room combinations.

Appendix IV of this manual, titled **Custom Room Combining** (beginning on page 71), details the use of the Room Combine Set-up utility.

Mixer Name...Enter/Edit *(Edit Pull-Down Menu)*

The Audio Net+® software allows each mixer or input expander on the network to be given a meaningful name. This makes it much easier and very intuitive to select and work with individual units. A mixer with the name "Exhibit Hall" is much more meaningful than *884+* at address number 5.

To give a mixer or input expander a name, select the Mixer Name...Enter/Edit menu item. A box will appear with the current or default name of the unit. Type in the desired name and press the OK button. To exit without making any changes, press the cancel button.

Lock Unit with Password *(Edit Pull-Down Menu)*

Password security is available for mixers and input expanders. The Lock Unit with Password utility provides a pop-up window which allows the entry of a four digit password for locking units. *Characters cannot be used as part of a password, only digits between 0 and 9. All four digits must be used.*

UNlock Unit with Password *(Edit Pull-Down Menu)*

To unlock a unit with a password, the UNlock Unit with Password utility provides a pop-up window which allows the entry of a four digit password. *Characters cannot be used as part of a password, only digits between 0 and 9.*

Set Unit Address *(Edit Pull-Down Menu)*

Whenever there is more than one Audio Net+® unit on the network simultaneously, each unit ***must*** have its own unique address. Mixers and input expanders are shipped from the factory with a default unit address of 1 or 2. The factory may, however, sequentially number units when several are shipped together.

In any event, it may become necessary to give a unit a unit address. The Set Unit Address utility facilitates that requirement. A pop-up window allows a new unit address to be assigned via the computer keyboard. In allocating unit numbers, it is important to remember that Ivie dual channel equalizers require two unit numbers, one for each channel.

Important: *When assigning a mixer or input expander a unit number, it must not be connected to the network at the time of assignment.* To assign unit numbers, each Audio Net+® product must be connected to the computer individually and assigned its unit number, before it is connected to the network. If a unit is connected to the network when a unit number is assigned, all of the units on the network will happily accept the same unit number, and *this will absolutely cause problems.*

Re-Assign Unit Address *(Edit Pull-Down Menu)*

If several mixers and/or input expanders are on the network simultaneously, it is possible, using the Re-Assign Unit Address utility, to change the unit number assignment of any one of the units without disconnecting that unit from the network. The Re-Assign Unit Address utility is different from Set Unit Address because it allows selective setting of unit addresses. The Set Unit Address utility, by contrast, sends a universal command that will cause every unit on the network to be set to the same address.

Re-Assign Unit Address can only be used if each unit on the network has a different address. If there is more than one unit on the network with the same unit address, the duplicate addressed unit must be removed from the network and given a new address using the Set Unit Address utility. *Also, if a unit for some reason has lost its address, the Re-Assign Unit Address utility may not be used to restore that lost address.* Restoring a lost address requires that the unit be removed from the network. The Set Unit Address utility can then be used to restore the lost address.

Automix *(880+ and 884+ Only)*

The Automix pull-down menu contains three items. These include ALC Levels, ALC and Vox Sens and

Automix Parameters.

ALC Levels (*Automix Pull-Down Menu)*

The *880+* Input Expander and the *884+* Automatic Matrix Mixer have Automatic Level Control (ALC) available on each of their inputs. It is switchable, so it can be turned either on or off for each input individually. The ALC Levels utility provides the ability to adjust the ALC. Two channels at once can be displayed and the action of their ALC's viewed on a chart recorder display. Any two channels can be selected for display.

The ALC and its setup is discussed at length in **Appendix III**, page 67, of this manual titled **Automatic Level Control Setup**. Please consult that section of this manual for complete information.

ALC and Vox Sens (*Automix Pull-Down Menu)*

As mentioned above, Automatic Level Control can be turned on or off for each input channel of an *880+* or *884+*. The ALC and Vox Sens utility provides a pop-up window with controls for turning on or off the ALC for each channel. In addition, there are controls provided to adjust the Vox Sensitivity for any or all input channels, should any or all channels be set to Vox operation.

Vox Sensitivity adjusts the level above ambient required to gate on a Vox mic. A Vox input is not associated with any other input. A Vox microphone compares its level to Vox Sensitivity in determining when to gate. It gates on when a signal of sufficient level is present at the microphone.

Vox Sensitivity is set for each of the eight input channels separately. If set too low, the mic will gate on randomly. If set too high, the mic will be difficult to gate on under normal operation. Vox Sensitivity is adjustable between 1 and 100, and is factory preset at 40, which is sufficient for most applications. If Vox sensitivity must be changed, the ALC and Vox Sens utility allows changes to be made.

Automix Parameters (*Automix Pull-Down Menu)*

The Automix Parameters utility provides a pop-up window that allows control of four automix parameters and one Vox parameter. These parameters include activation of the Last On feature, Difference adjustment (an ease of gating parameter), Off Attenuation, NOM Attenuation, Release time for automixers and Vox operation and Sensitivity adjustment for automixers.

Last On is an Ivie innovation which enhances the operation of automatic mixers. It is very useful when tape recording the output of the mixer, or when the mixer signal is being fed to some other room, or is being broadcast. When activated, the Last On feature prevents the annoying "pumping," that is, background noise being heard, and then not heard as the system changes from all microphones being gated "off" (no background noise), to at least one microphone being gated "on" (background noise present). When the Last On feature is activated for a particular automixer (W, X, Y or Z), that mixer will always have at least one microphone gated on at all times. Simply stated: The last microphone gated on will remain on until another microphone is gated on to take its place. To activate Last On, click the appropriate box in the Automix Parameters pop-up window.

Difference adjustment allows fine tuning the amount of level difference (in dB) which must be present at a given microphone - when compared to the level at other microphones - before that microphone is gated

on. The "Difference" default setting is 6dB. The recommended range is from 4dB to 8dB. The factory setting of 6dB should work well for most automixer installations. *The difference setting is global. There is one setting for all four automixers, not a separate adjustment for each automixer.*

Off Attenuation provides the ability to set the level of channels that are gated off. How far down from "on" is "off?" The Off Attenuation parameter allows a global setting from 0dB to -99dB in 1dB steps.

NOM Attenuation allows setting a global NOM (number of open microphones) attenuation of either 3dB or 0dB. If NOM attenuation is set for 3dB, the level will be attenuated by 3dB each time the number of open microphones doubles (one mic active = no attenuation, 2 mics active = 3dB of attenuation, 4 mics active = 6dB of attenuation etc.).

Release Time on an automatic mixer has a purpose of providing smoother, more seamless operation of the automatic mixer. If an input channel were gated off immediately when the signal at microphone became too low, it would sound very distracting. The channel would gate off between syllables, and in some cases, even during a syllable. Consequently, all automatic mixers have some amount of release time delay built-in in order to slow the gating off of a channel. Ivie automixers have an adjustable release time. The release time is adjustable over a range of 1 second to 10 seconds in 1 second steps.

The *880+* and the *884+* have a release timer built into all four of their internal automixers, "W," "X," "Y" and "Z." It is also a function of Vox operation. The release timer keeps a channel gated on for a period of time after the automatic mixer has determined that the signal on the channel is low enough in level to require gating the channel off. Ivie recommends a release time of about three seconds, which is the factory default setting. If it is necessary to change that setting, the pop-up window of the Automix Parameters utility provides that capability for both the automixers and Vox.

Sensitivity for the automixers is very similar to Vox Sensitivity. It adjusts the level above ambient that must be present at one of the automixer mics before the automixer will begin to gate. It is especially important when only one microphone is connected. Each automixer (W, X, Y and Z) can be individually set. The function of Amix (automix) Sensitivity is to prevent random gating in ambient conditions. Like Vox Sensitivity, if it is set too low, random gating will occur, and if set too high, gating will not occur under normal operating conditions. Amix Sensitivity is adjustable between 1 and 100, and is factory preset at 40, which is sufficient for most applications.

View *(Mixers and Input Expanders)*

The View pull-down menu contains six items. These include: 625/626 DSP, 730/728 Equalizer, 784/884 Matrix Mixers, 802A Mixer, Sinewave Oscillator and Audio Net Listing.

625/626 DSP *(View Pull-Down Menu)*

When selected, the 625/626 DSP menu item displays the ANSW+ software control screen for the 625 and 626 Digital Signal Processors.

730/728 Equalizer *(View Pull-Down Menu)*

When selected, the 730/728 Equalizer menu item displays the ANSW+ software control screen for the

728PW and 730PW 1/3 Octave Equalizers.

784/884 Matrix Mixers *(View Pull-Down Menu)*

When selected, the 784/884 Matrix Mixer menu item displays the ANSW+ software control screen for the *784+, 884+*784PW and 884PW Matrix Mixers and the *780+, 880+,* 780PW and 880PW Input Expanders.

802A Mixer *(View Pull-Down Menu)*

When selected, the 802A Mixer menu item displays the ANSW+ software control screen for the 802A Automatic Mixer.

Sinewave Oscillator *(View Pull-Down Menu)*

Ivie matrix mixers and input expanders have a built-in sinewave generator that can be placed under software control. The oscillator operates over a frequency range of 20Hz to 20kHz. The Sinewave Oscillator utility provides a pop-up window that allows control of the oscillator. There are three sets of controls that can be used to select frequency, and a button to turn the oscillator off. It automatically turns on when any of the three sets of controls are used to assign a frequency.

The first set of controls allows the selection of discrete frequencies of 20Hz, 20kHz, or any octave frequency center of the ten ISO standard octaves of the audio band (31.5Hz, 63Hz, 125Hz etc.).

A second set of controls allows the selection of a frequency, either up or down, in 1Hz steps.

The third set of controls allows the selection of a five digit frequency by decade. There is a separate set of controls for selecting each decade, 1's, 10's, 100's, 1000's and 10,000's.

Audio Net Listing *(View Pull-Down Menu)*

When selected, the Audio Net Listing utility provides a pop-up window with controls that allow interrogation of the Audio Net+® network to see what Audio Net+® units are on the network. A Poll button is provided along with windows that list unit types and unit names (if units have been given names).

Also provided is an Audio Net List pull-down window that allows selection of any unit on the network for control. Once a unit has been selected, the ANSW+ software recognizes the unit type and automatically displays the proper ANSW+ software control screen for that unit.

After the Audio Net Listing utility pop-up window has been closed, a listing of units on the network is still available through the Controlling Mixer pull-down window located in the Audio Net+® window of the ANSW+ software (lower, left-hand side of the ANSW+ software control screen).

Modem *(Mixers and Input Expanders)*

The Modem pull-down menu contains only one item titled Display Modem Window.

Display Modem Window *(Modem Pull-Down Menu)*

Ivie matrix mixers and input expanders have the capability to be addressed via a modem. The Display Modem Window utility facilitates that capability. The Display Modem Window utility provides a pop-up window for modem interface. Typing "ATD" followed by the desired phone number will establish communication. Then, follow the instructions in the modem window. *There must be an Ivie 78-232M modem connected to the mixer or input expander at the other end.*

Once communication is established, the Display Modem Window can be closed for operation of the ANSW+ software. When finished, open the Display Modem Window again and press the "Hang Up" button to disconnect the telephone line.

Options *(Mixers Only)*

The Options pull-down menu contains only one item, Positive Gain Trim on Outputs.

Positive Gain Trim on Outputs *(Options Pull-Down Menu)*

When selected, the Positive Gain Trim on Outputs adds positive gain capability (up to +10dB) to the output Trim (Gain Trim) function. The inputs already have positive gain available (their trim range is +10dB to -74dB), but the normal output trim range is +0 to -74dB. It is generally not a good idea to have positive trim on an output because it increases the noise as well as the signal level.

However, in some special situations, it may be desirable to have positive trim available on an output, so the Positive Gain Trim on Outputs utility provides that capability. It should be selected with great care, however, because it will increase noise as well as increasing signal level.

Help *(Mixers and Input Expanders)*

There Help pull-down menu contains eight items: Anet+ Start-Up Options, Controls, Initial Combine Level/Volume Tracking, Modem, Programming Presets, Programmable Room Combining, Stacking Mixers and About.

Anet+ Start-Up Options, Controls, Initial Combine Level/Volume Tracking, Modem, Programming Presets, Programmable Room Combining, Stacking Mixers *(Help Pull-Down Menu)*

These Help Menu items are self explanatory. Selecting one will provide information on that subject.

About *(Help Pull-Down Menu)*

The About menu item from the Help pull-down is informational. It identifies the version of ANSW+

software currently being run on the computer, and explains how the latest release of software may be obtained from Ivie's web site at www.ivie.com.

The About pop-up window also identifies the version of firmware running in the Ivie Audio Net+® unit under control. The version of the firmware is important information and should always be provided to the factory when phoning or faxing the factory for assistance.

Appendix I
Reference Information

Software Updates *(Mixers and Input Expanders)*

The ANSW+, Windows® based software is constantly being updated. New features are added and capability expanded. While every effort is made to keep the software compatible with older versions of firmware, this is not always possible, especially when new features are added. The latest version of software is always available directly from the factory, or at no cost from Ivie's web site at *"www.ivie.com."* If there are any questions regarding backward compatibility, please contact the factory. Have the firmware version of the product of concern available when calling or faxing (for a description of how to get the firmware version information, see the section of this manual titled **About *(Help Pull-Down Menu)*** on page 42).

Application Notes *(Mixers and Input Expanders)*

In addition to the information found in this appendix, there are a number of application notes available from Ivie that show the features of the matrix mixers being employed in interesting and innovative ways. Such subjects as paging, room combining, court room application and remote control are shown in various configurations, with helpful information provided. To request application notes, simply write, Email at *"ivie@ivie.com",* fax at 801-224-7526, or call us at 801-224-1800.

Programming Manual *(Mixers and Input Expanders)*

For those interested in, or needing to "talk" to matrix mixers (or other Ivie Audio Net+® products) via Ivie's RS-485 based, Audio Net+® network, Ivie has an Audio Net+® programming manual available. This manual provides all the Hex and ASCII code required to address all functions of the mixers through the Audio Net+® network.

The Audio Net+® programming manual will prove especially valuable to those programming touch panels, such as those available from AMX® or Crestron®. Level control, room combining, input/output assignments, preset activation, paging functions and many other features of the matrix mixers can be accessed through a programmed touch panel. To request an Audio Net+® programming manual, simply write, Email at *"ivie@ivie.com",* fax at 801-766-8282, or call us at 801-766-7600.

Input and Output Connectors *(Mixers and Input Expanders)*

Figure 32 below details the DIN connector inputs and outputs on mixers and input expanders:

784+ & *884+* Mix Bus I/O Connections

(The pattern is the same for *780* + & *880+*, except they have no outputs.

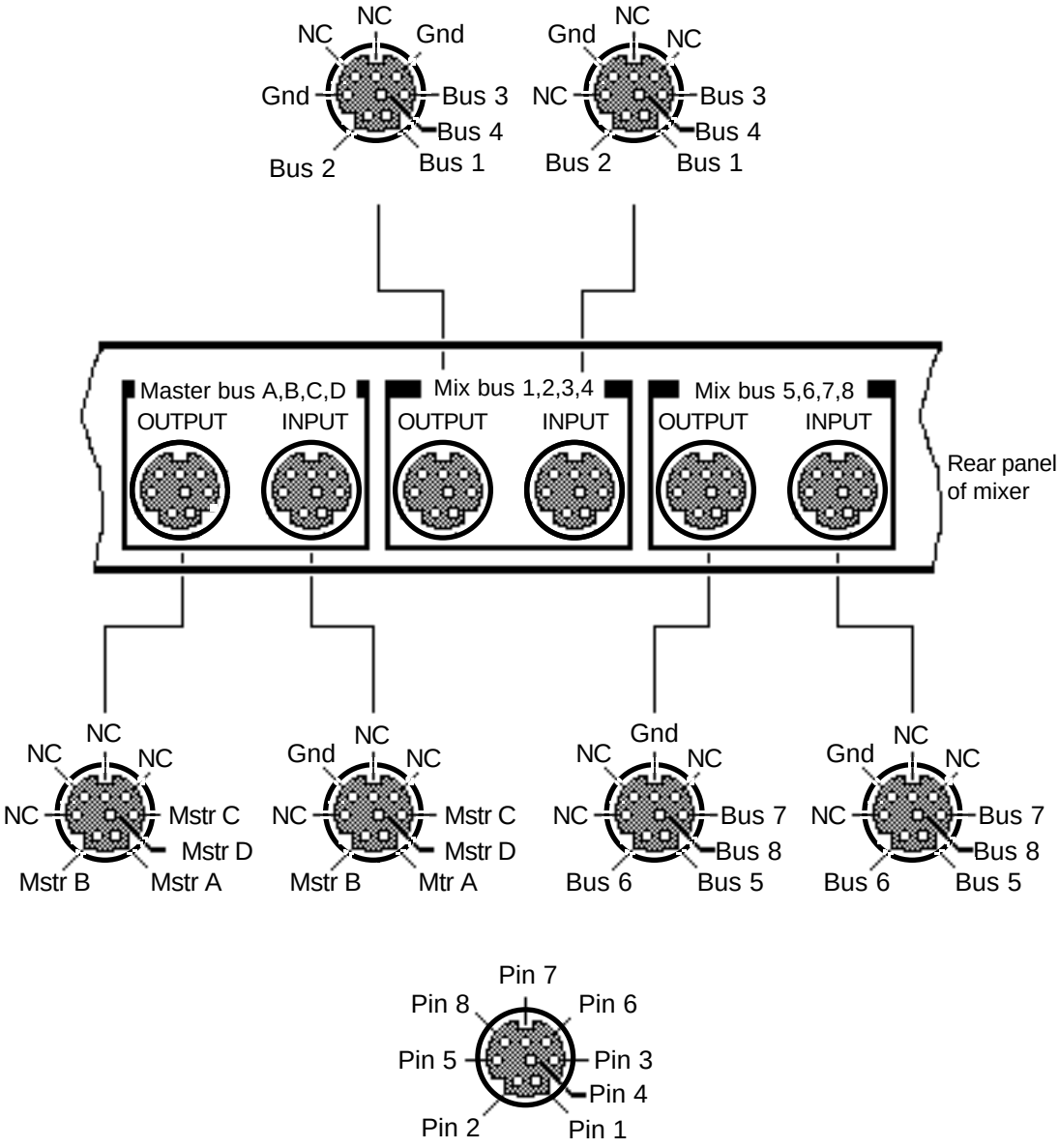


Figure 32

Ganging Mixers and/or Input Expanders

78C-6 Jumper Connections for 5 Mixers (40X8 Matrix)

Diagram for 2 Mixers and 3 Input Expanders would be identical except bottom 3 units (input expanders) would have no Master Bus A, B, C and D DIN connectors.

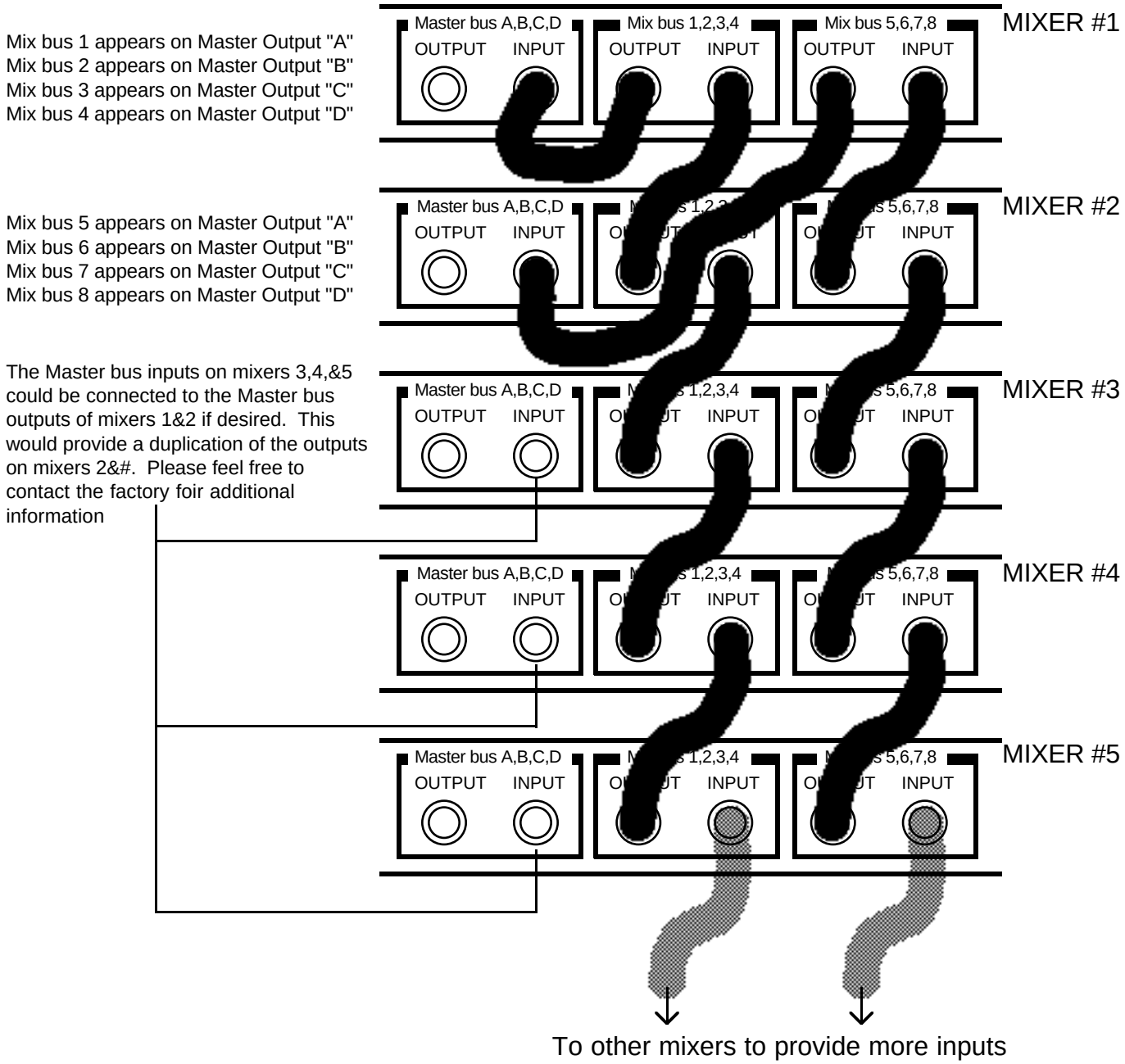


Figure 34

Appendix II
Programming Presets

What's a Preset?

Ivie matrix mixers and input expanders have 15 presets numbered 0 through 14. Conceptually these presets are similar to presets on a lighting control console.

So what is a preset? A preset is a convenient name for a memory - a memory that contains the settings of almost all the functions of the mixer. "Saving" a preset writes into memory the levels of all inputs, outputs, remote pot assignments, remote button assignments, mix bus assignments etc. A "saved" preset is a "snapshot" of the current mixer configuration. When a preset is "stored," the current setup is memorized and assigned a preset (or memory) number. Different mixer configurations can be "stored" as different presets, and these various configurations can be recalled by activating the appropriate presets. Preset activation can be effected by a number of different methods: Contact closure, ANSW+ software, and code over the Audio Net+® network from an external device such as an AMX® or Crestron® touch panel.

Some mixer and input expander parameters can be stored as part of Preset 0, but cannot be stored and part of Presets 1 through 14, so a discussion of Preset 0 is in order.

Preset 0 *(Zero) (Mixers and Input Expanders)*

Preset 0 is the "base" preset. Imagine for a moment a matrix mixer where presets 1 through 14 do not exist. There is only one program memory (Preset 0). To install this computer controllable mixer, after physically putting it in a rack, a computer would be connected to on of its Audio Net+® ports in order to program it, or set it up for the installation. Using the computer, various inputs would be set to either automatic or manual operation. The matrix would be programmed to route the inputs to the proper outputs. Levels would be set and, if remote controls were being used, the mixer would be "told" to recognize them. After the mixer was configured and setup was complete, the computer could be disconnected because the mixer would be ready for operation. *By definition, Preset 0 would have just been programmed.*

Programming Preset 0 would do no good unless the programming were "saved" to the mixer's memory, so the last thing that would have to be done before disconnecting the computer would be to save the programming. This would be done by pressing the "Save" button in the Preset and Room Combine window of the ANSW+ software control screen. Pressing the Save button would cause a prompt to appear on screen asking if saving the preset is really wanted. To complete the Preset 0 saving process, clicking the "OK" button would be necessary.

Once programmed and saved, Preset 0 is always active. It is the base configuration of the mixer, the normal mode of operation when no other presets are active. The other presets, 1 through 14, "overlay" Preset 0. They make changes or alterations to Preset 0 operation. This is why Preset 0 must be programmed first, before additional presets can be programmed. All other presets are really modifications to Preset 0.

Every time the mixer is turned on, it will "come up" configured to the settings of the preset or presets that were last active when it was turned off, just as would be expected. If no preset was active, the unit will come up configured to the base preset, Preset 0. *All preset memories within the mixer are stored in non-volatile memory. No batteries are required to maintain the memory of the unit.*

Presets 1 - 14 *(Mixers and Input Expanders)*

Presets 1 through 14 overplay Preset 0. More than one preset can be active at the same time, and as long as there are no conflicting instructions, they will all live together happily. What happens if two presets are active and they have conflicting commands (one says “turn on background music,” and the other says “turn off background music”) for example? The answer is: The higher numbered preset will win the conflict. Presets 1-14 have built-in priority. Preset 14 has the highest priority. Preset 5 has priority over Presets 4, 3, 2, 1 and 0, etc. When programming presets, it is important to use higher numbered presets for those events requiring priority over other possible conditions.

A simple example of preset priority would be an “All Page.” Normally, an All Page is intended to take priority over whatever else is going on at the time. It would override other pages (local ones, for example), turn off or “duck” background music, override any remote controls that had level turned down, and so on. Whatever was happening in any room, or zone of the system, the All Page would take precedence and come through loud and clear. Preset 14 would be a good preset to use for an All Page function. Nothing else could have a higher priority.

While Preset 0 is always active, Presets 1 - 14 are active only temporarily. They can be turned on or off at will. They can be activated by a contact closure, the ANSW+ software, or by command string over the Audio Net+® network. A common use of presets is in paging applications. For example: Preset 1 could be programmed to turn on a microphone and route it to one or more outputs (zones). As long as the switch connected to the preset terminal is active (depressed), the paging microphone will be active. When this paging Preset 1 is activated, it "overlays" the base configuration (Preset 0). As long as the paging button is depressed, Preset 1 will be active as well as Preset 0. As soon as the paging button is released, Preset 1 would no longer be active. More than one preset can be active at a time, thus "layering" one preset upon another. Each preset is independent from all other presets as long as there are no conflicts among them.

The Preset and Room Combining Window

The Preset and Room Combining Window is alway active in the ANSW+ software control screen. The buttons in the window are used to save the programming of Presets 0 through 14 and to edit the programming of Presets 1 through 14. The Room Combining Buttons are used to activate or deactivate room combining. The Preset and Room Combining Window with its various buttons is illustrated in Figure 35 on the following page:

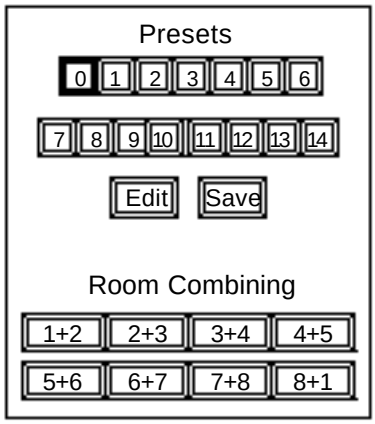


Figure 35

The Preset and Room Combining window allows the programmer to observe the status of presets, to activate presets with the mouse cursor, and to program presets. The Edit and Save Buttons are used only during the programming of presets. The Edit Button need not be depressed when programming Preset 0, but it must be pressed when programming any preset higher than Preset 0. The Preset and Room Combining Window is reviewed again in Figure 36 below, with its functions identified:

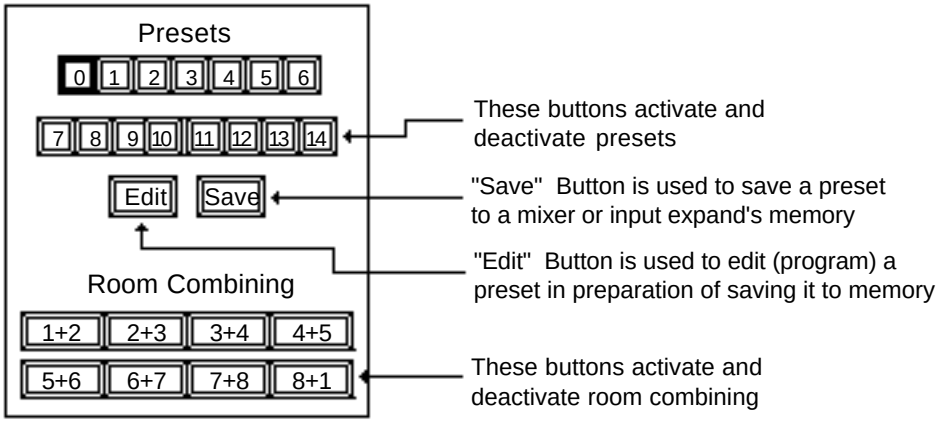


Figure 36

Programming Preset 0 *(Zero) (Mixers and Input Expanders)*

Before Presets 1 through 14 can be programmed, the base configuration of the mixer or input expander must first be programmed. As previously stated, Preset 0 is always active, once programmed, and all other presets overlay it. The easiest way to illustrate programming any preset is through the use of a specific example. Even complex requirements can be easily programmed, once a few basic concepts are

understood. Figure 37 below illustrates a three meeting room application with requirements for paging, room combining, background music and remote control:

Three Meeting Room Application

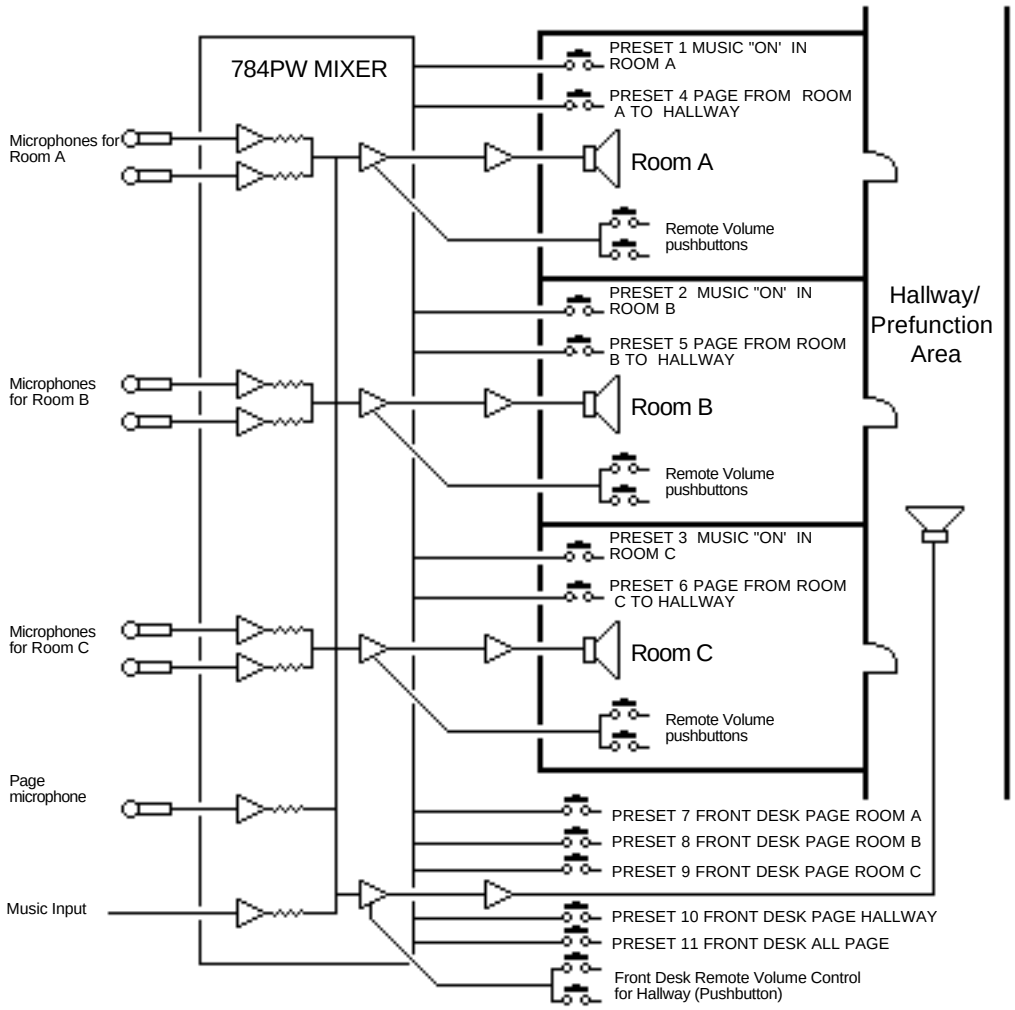


Figure 37

In this particular application, there are three meeting rooms and a hallway (prefunction area) that require sound reinforcement. Each meeting room has two gated microphone inputs. There is a local page function inside each room that, when enabled, sends both microphone inputs not only to the room, but also into the hallway. Each room has a background music on/off switch for turning on background music (which, in this application, also turns off the microphones in the room) and there are two room combining switches for room combining (A+B and B+C). The last input is a paging microphone that allows the front desk to page each area individually, or page all areas (zones) simultaneously.

Finally, there is an Ivie RM-1 up/down push button level control in each room, and the same type control at the front desk which sets the background music level in the hallway. All of these requirements can be handled nicely by a single *884+* Automatic Matrix Mixer.

The first question to ask when programming Preset 0 is, "What is the normal operating configuration of the system?" Once the answer to this question is known, programming becomes simple. For the system in Figure 37, the normal operating configuration is as follows: Each room has both of its microphones active. Background music is off. The rooms are not combined. The master level control for each room room is active. In the hallway, background music is on and the level control for the hallway is active.

Figure 38 below shows the ANSW+ software control screen and demonstrates the programming that has been done to set up the mixer for its base configuration (Preset 0):

Programming Preset 0 (2 Mics Each into 3 Rooms, Background Music on in Hallway)

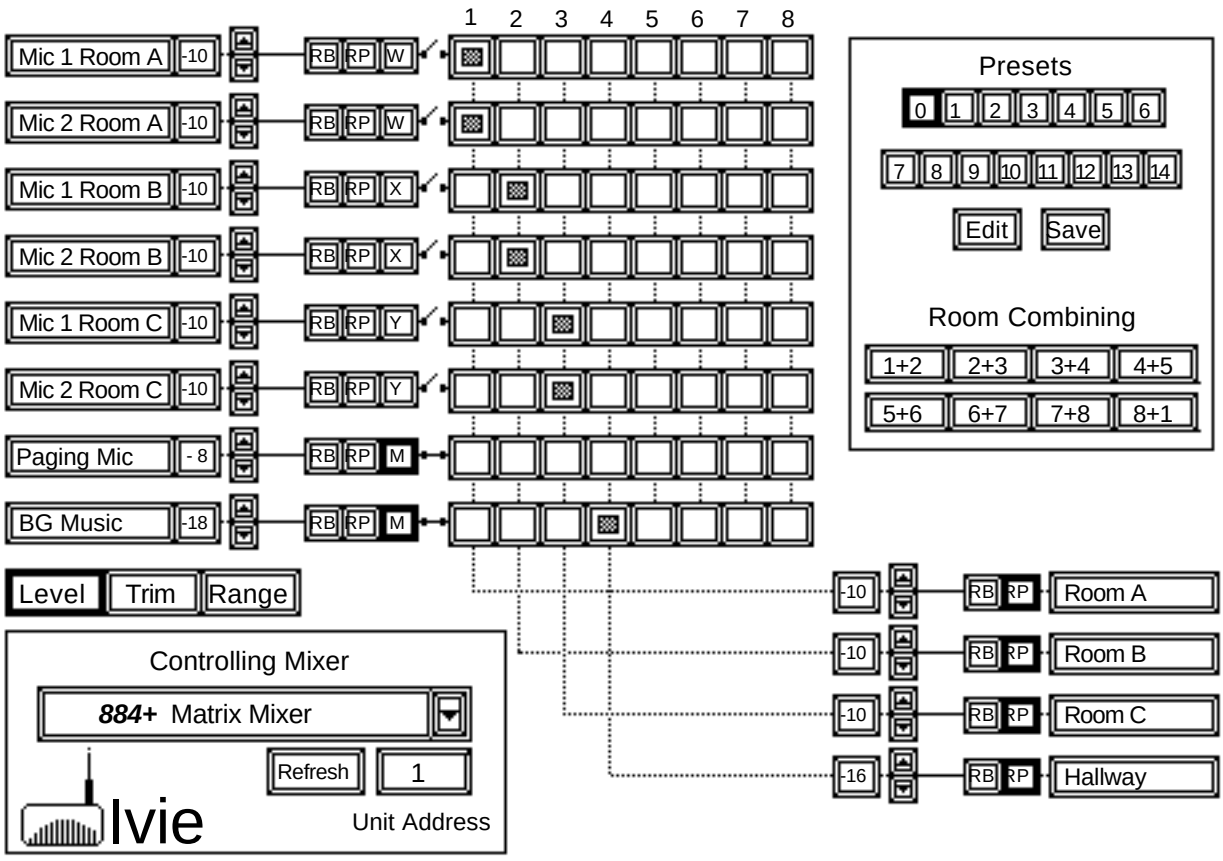


Figure 38

The best way to understand what has been done to arrive at the programmed screen in Figure 38 above is to go through the steps one by one.

1. Using the "File" pull-down menu, "Poll" has been selected to establish communication with the *884+*.

2. By placing the cursor over the input label area (ie. placing the cursor over the window that

says “Mic 2 Room A”) and double clicking, a pop-up window allows custom labeling all the inputs and outputs, as has been done.

3. Using the Automix Assign Buttons, the first two inputs have been assigned, or grouped together as automixer “W.” The next two inputs have been assigned to automixer “X” and the next two inputs have been assigned to automixer “Y.” In this configuration, there are three, independent automixers in the box - one for each room. The Paging Mic input has been assigned to “Manual On,” as has the Background Music input (notice the “M” for “Manual” and the depressed button for “On.” Notice also that the Software Gating Switch is latched showing that these two inputs are always on.

4. Inputs have been assigned to mix buses (which feed the outputs as shown). An input to output assignment is made by placing the cursor on a Bus Assign Button and clicking. As the cursor passes over a button, the input to output assignment that would be made is highlighted as a programming aid. Once a button is pressed, positive indication is given by the blue square in the center of the grey button. As can be seen in Figure 38, the two Room A mics have been assigned to the Room A output. The same is true of the microphones for Rooms B and C. No assignment has been made for the Paging Mic which is normally not active (it is activated by a contact closure). The normal configuration for Background Music is that it is on in the Hallway, but not in the meeting rooms, so that assignment has been made, as can be seen.

So far, inputs and outputs have been custom named, inputs have been grouped as separate automixers, or set to manual as necessary, and input to output assignments have been made. However, more programming has been done.

5. Since there are remote controls on each output, they have been activated by pressing the RP buttons (The Remote Potentiometer button is used because the RM-1 “looks like” a pot to the mixer). *If the RP Button were not pressed, the remote control would be deactivated.*

6. Lastly, the levels have been set for all the inputs and outputs. This, of course, would have been done while the system was operating. If limiting the range of the remote controls, or setting some gain trim to limit maximum gain had been necessary, this also could have been done using the Range and Trim functions.

7. The final step would be to save the programming as Preset 0. This would be done by pressing the Save button. A prompt would come up allowing a chance for a change of mind. If all is OK, clicking the “OK” button would save the programming as Preset 0.

What happens if Preset 0 is programmed, and a change is desired? No problem. Make the change and press Save again. The new programming will entirely overwrite what is in memory, replacing what was previously saved as Preset 0.

Once Preset 0 has been completely programmed to satisfaction, other presets can be programmed.

Programming Presets 1 - 14 *(Mixers and Input Expanders)*

The most difficult part of programming Presets 1 through 14 is deciding what should happen when a preset is activated. Once that decision has been made, programming a unit to execute the desired action

is easy. In the application shown in Figure 37, one of the system requirements is a switch in each room that turns on background music and kills the mics. This switch would activate a preset. The programming of that preset (Preset 1) for Room A is shown in Figure 39 below:

Programming Preset 1 (Room A Background Music On, Local Mics Off)

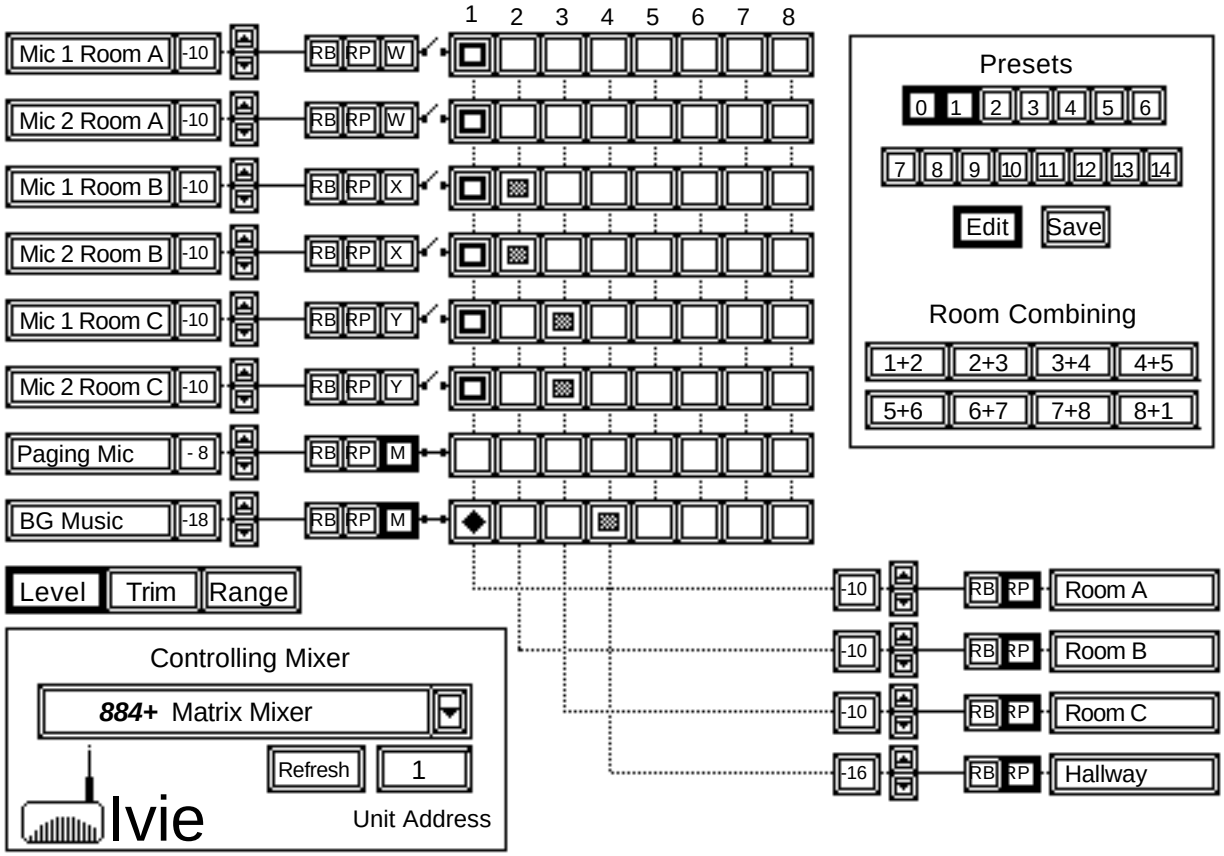


Figure 39

The steps in programming Preset 1 as show above are:

1. Press the Edit Button. The Edit Button must be used with Presets 1 - 14. This notifies the unit being programmed that a preset different from Preset 0 is being programmed.

2. Press the desired preset button. Notice in Figure 39 that the Edit and Preset 1 Buttons have both been pressed. This indicates that Preset 1 has been selected for programming. *The Edit button must be pressed first.* Pressing the Preset 1 Button first would give an "Activate Preset 1" command instead of an edit or "Program Preset 1" command.

3. Program the changes to Preset 0 that Preset 1 will cause when activated. Next, press the Save Button to save the programming of Preset 1 and then press the Edit Button once again to exit the Edit (Preset Programming) Mode.

Step #3 demonstrates an important concept. Preset 1 "overlays" Preset 0 and causes only specific

changes. Notice that the rest of the Preset 0 programming is still visible on the screen. The blue squares still show the bus assignments made which feed Rooms B, C, D and the Hallway. The remote controls for all areas remain active. The only thing that has been changed is the signal being fed to Room A The intent of Preset 1 was to feed background music to room A and turn off the microphones in Room A. The first Bus Assign Button that has been pressed is the Background Music input connecting this input to Mix Bus 1, which feeds Room A. As a programming aid, when in the "Edit" mode, the colors and, in some cases, the shapes of indicators change. Notice that the indicator for the "Background Music to Mix Bus 1" Bus Assign Button is a red diamond.

In the Edit mode, all programmed information shows up in red on the ANSW+ software control screen. This provides positive indication that the Edit mode has been selected. In programming Preset 0, a Bus Assign Button is either pressed or not pressed (blue square inside grey button, or no blue square, respectively), but in the Edit mode, three states are possible: 1) Definitely On, 2) Definitely Off, and 3) Unchanged. When modifying Preset 0 (which is exactly what Presets 1 - 14 do) three possibilities exist for a change to a Bus Assign Button: Let it stay the way it was in Preset 0 (Unchanged), or, make the bus assignment (Definitely On), regardless of whether it was made or not made in Preset 0, or definitely do not make the bus assignment (Definitely Off), regardless of whether it was made of not made in Preset 0. Figure 40 below demonstrates these indicators on the Bus Assign Buttons:

Bus Assign Status Indicators

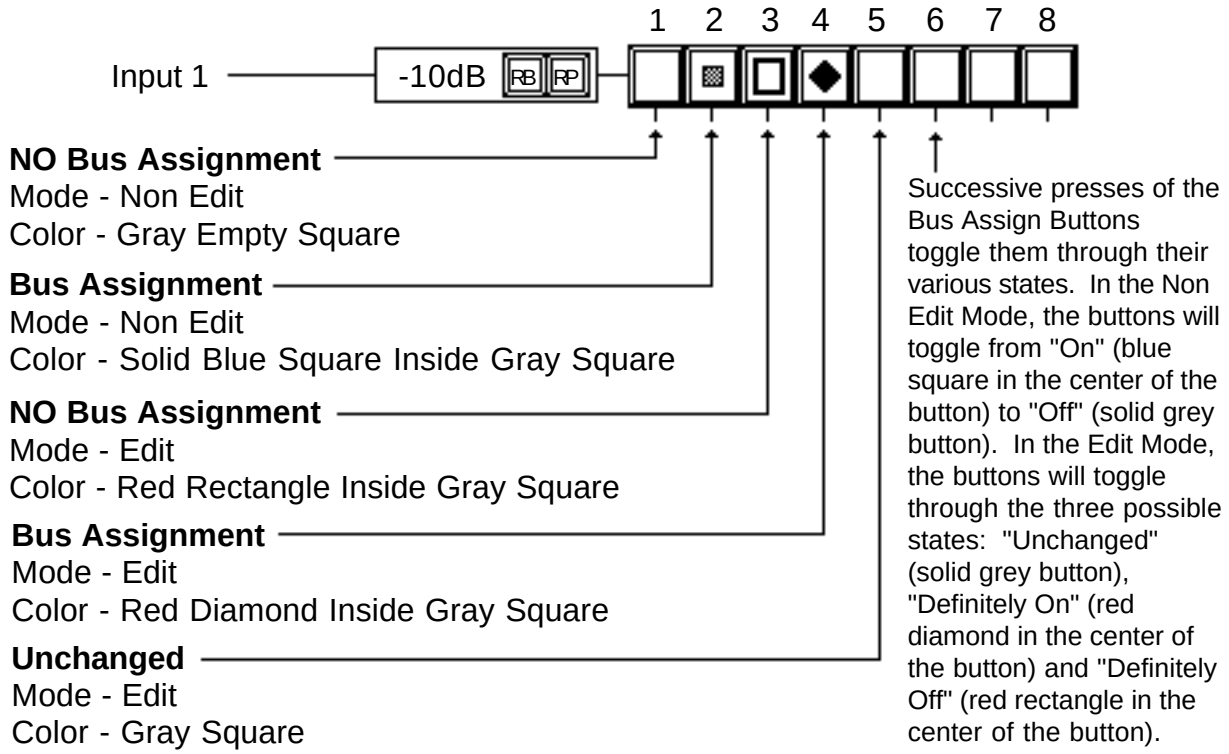


Figure 40

As shown in Figure 40, a "Definitely On" bus assignment shows up as a red diamond, while a "Definitely Off" bus assignment shows up as a red rectangle. Referring back to Figure 39, the microphone inputs for Room A have been turned off as part of Preset 1. *However, notice that the*

microphone input to rooms B and C are also turned off. Although these microphones do not normally feed Room A, they *would* feed A in a room combining situation. If Rooms A and B were combined, the Room A microphones would feed both rooms and the Room B microphones would feed both rooms. The precaution has been taken, in programming Preset 1, of turning off any microphone that may be feeding Room A when background music is turned on. Figure 39 illustrates this clearly.

In the same manner that Preset 1 turns on background music in Room A, Preset 2 could be used to turn on background music in Room B, and Preset 3 could be used to turn on background music in Room C. The programming steps would be identical, except that Mix Bus 2 would be changed in Preset 2 and Mix Bus 3 would be changed in Preset 3.

Preset 4 could be used for the page function in Room A. When active, Preset 4 would send the Room A microphone inputs to the Hallway as well as to Room A. Figure 41 below illustrates the programming of Preset 4:

Programming Preset 4 (Page from Room A Mics to Room A and Hallway)

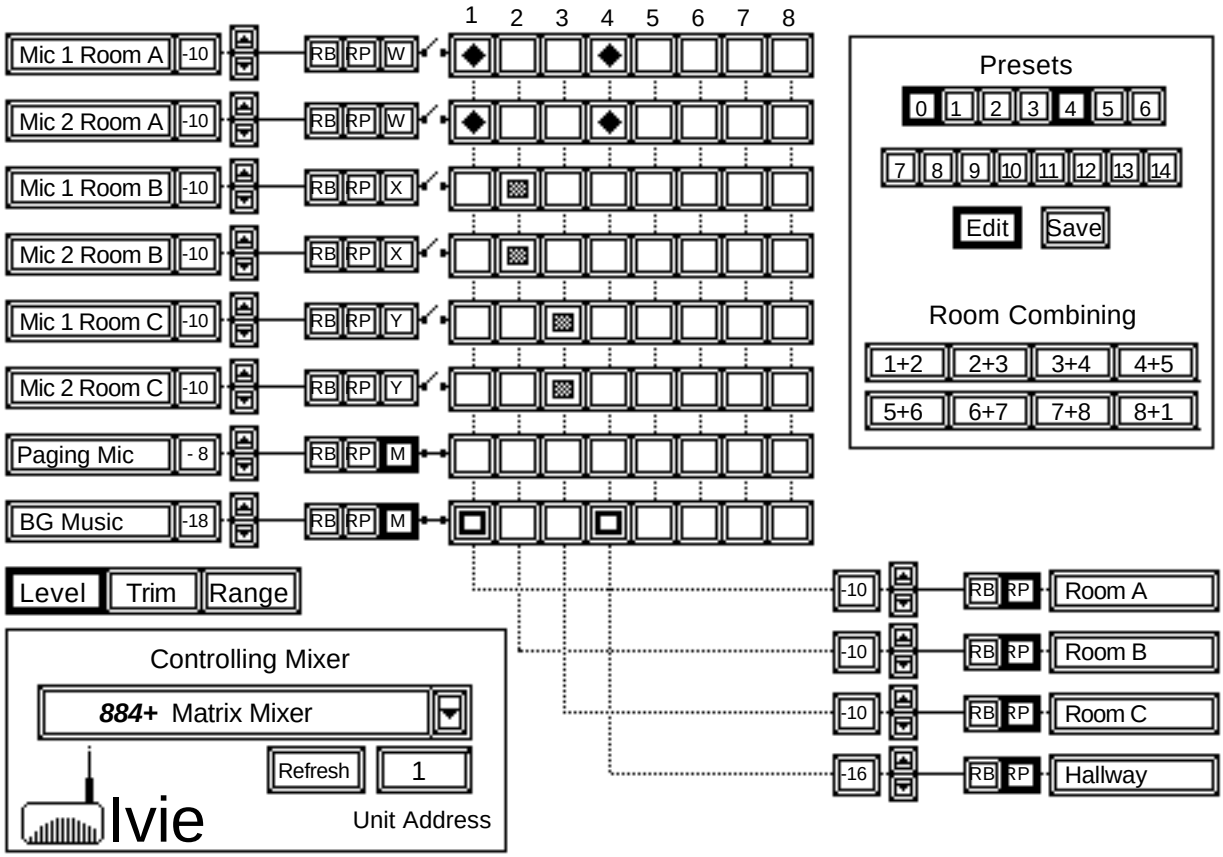


Figure 41

The steps for programming Preset 4 are as follows:

1. Press the Edit Button to enter the preset programming mode.

2. Press the Preset 4 Button to select Preset 4 for programming.
3. Assign (Definitely On) Room A microphones 1 and 2 to Room A (Mix Bus 1) and to the Hallway (Mix Bus 4) The red diamond indicates this has been done.

4. Turn off (Definitely Off) Background Music to the Hallway and to Room A. The red rectangles indicates this has been done.

5. Press the Save Button to save the Preset 4 programming to the *884+* mixer's memory.

6. Press the Edit Button again to exit the preset programming mode.

Important Note: Notice that as part of Preset 4's programming, Microphones 1 and 2 were assigned to feed Room A. In Preset 0, which is always active, these two microphones already feed Room A. Turning them on again as part of Preset 4 may seem redundant, so why do it? This is part of the foresight and planning that must occur before programming begins. Consider the following scenario which demonstrates the necessity of reassigning the two microphones to Room A as part of Preset 4.

Consider, for example, that a meeting in Room A had adjourned briefly for a break. Preset 1 had been activated to turn on background music during the break. When it was time to conclude the break, the person conducting the meeting would press the page button to address those in Room A and those still on break in the Hallway to announce the resumption of the meeting. The problem is that the microphones in Room A would be dead. They would have been turned off as part of Preset 1 activation when background music was turned on.

Solution: Preset 4 is a higher number preset than Preset 1. It therefore takes priority when a conflict occurs. If Preset 4's programming dictated that the microphones be on, it would, when activated, over ride Preset 1's dictum that they be off. As soon as the conductor of the meeting released the page button, the microphones would die and background music would resume. To begin the meeting, the conductor would have to turn off the background music (deactivate Preset 1) which would automatically turn the microphones back on. With a little experience in programming presets, these kinds of considerations become easier and easier to identify.

Just as Preset 4 was used to activate local paging for Room A, Preset 5 could be used for activating the local page function for Room B, which would send Room B's microphone output to the Hallway as well as to Room B. The programming would follow the pattern of Preset 4 shown in Figure 41. Likewise, Preset 6 could be programmed to perform the local page function for Room C.

The next phase of programming the *884+* for the three meeting room application would be programming the presets to allow paging from the front desk to the hallway, any room individually, or do an All Page. When doing paging, there are some level considerations that must be addressed. What happens, for example, if the front desk wants to page Room A, but Room A has background music on, and the remote master level control is turned down so low that the page cannot be properly heard?

These considerations are covered in detail in Figure 42 on the following page where Preset 7 programming is shown (Page Room A from the Front Desk):

Programming Preset 7 (Page Room A from Front Desk)

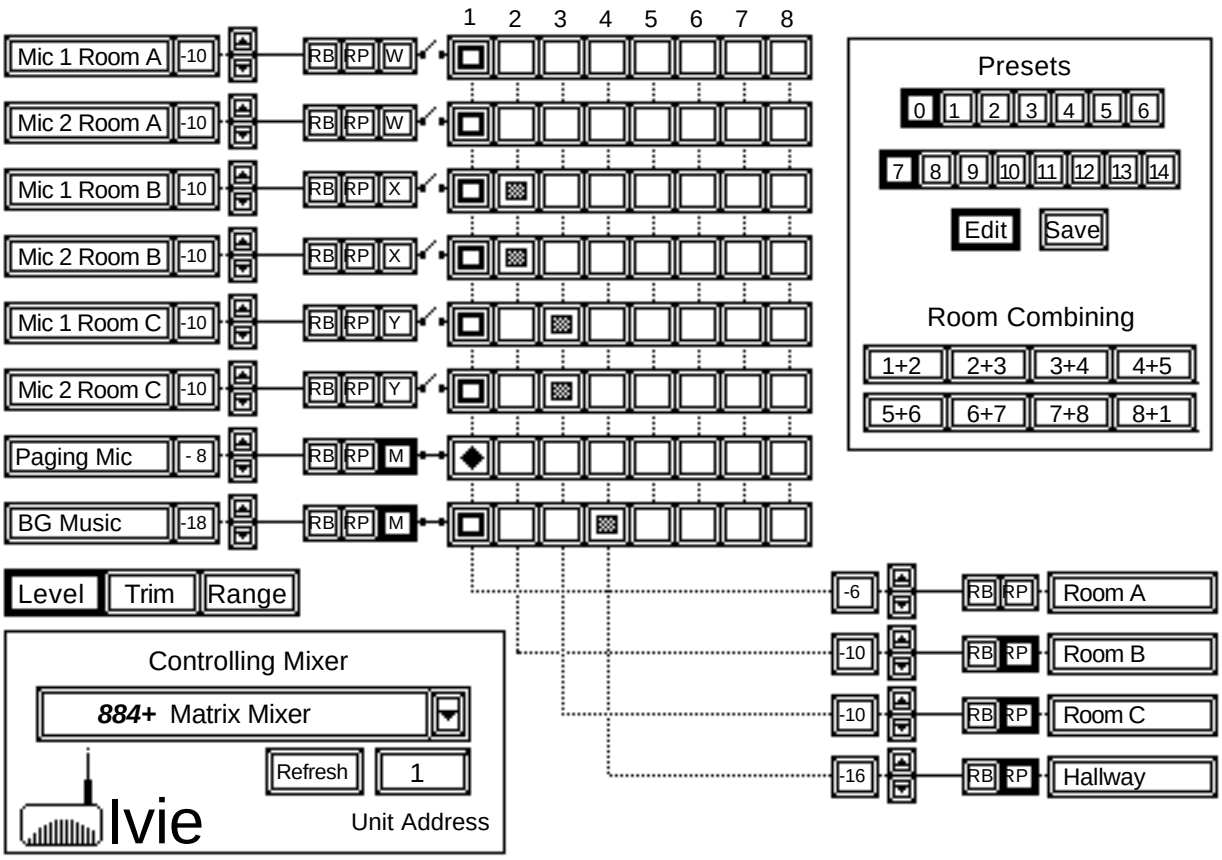


Figure 42

The steps for programming Preset 7 are as follows:

1. Press the Edit Button to select the preset programming mode.

2. Press the Preset 7 button to select Preset 7 for programming.

3. Assign (Definitely On) the front desk Paging Microphone to Room A using the appropriate Mix Bus Assign Button (the red diamond on the Mix Bus Assign Button shown in Figure 42 above demonstrates that this has been done).

4. Turn off every other input that could possible feed Room A (the red rectangles on Mix Bus Assign Buttons shown in Figure 42 above demonstrate that this has been done).

5. Turn off (disable) the remote control for Room A (notice the RB button for Room A output is no longer depressed. Clicking on this RB Button releases it. While the graphic in Figure 42 shows the button released, it cannot show that the button would be red in color. On the ANSW+ software screen, it *would* be red to indicate it had been changed as part of the preset.

6. Using the up/down level buttons, adjust the page level coming into Room A. (Notice that the Room A output level has been turned up from -10dB to -6dB in Figure 42 on the preceding page.) (Again, the changed level would be red in color.)

7. Press the Save Button to save the programming of Preset 7 to the *884+* mixer's memory.

8. Press the Edit Button again to exit the preset programming mode.

When activated, Preset 7, which has just been programmed, does a host of things. It turns on the page mic and routes it to Room A. It turns off any other signal than could be present in Room A. It disables the remote master level control in Room A so the page cannot be turned down by the remote control. Finally, it sets the level of the page into Room A at the desired level. All of these changes are part of the preset, and take effect as long as the preset is active.

The final preset that would need to be programmed for the three meeting room application is the All Page (Preset 11). The highest numbered preset used in this application is Preset 11 because the All Page has the highest priority of any condition of the sound reinforcement system. The programming of Preset 11 is demonstrated in Figure 43 below:

Programming Preset ("All Page" from Front Desk)

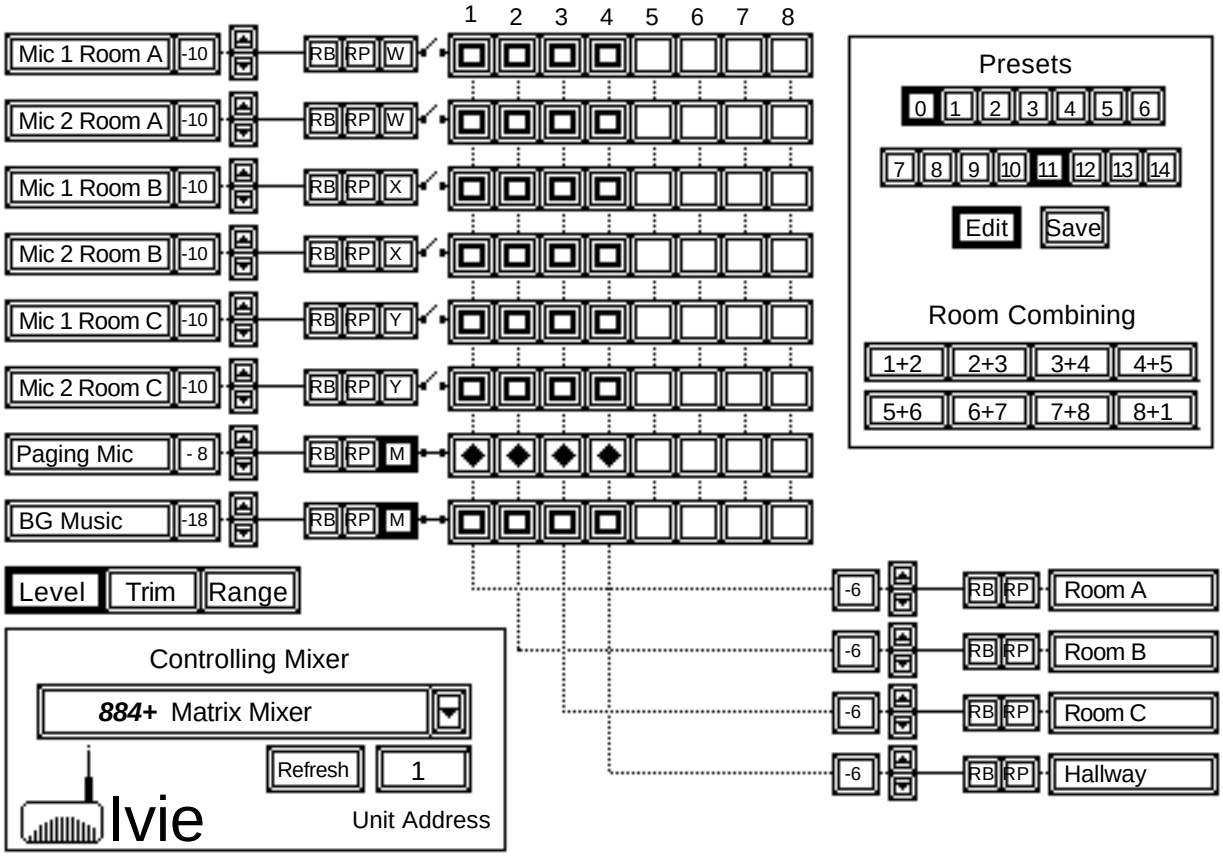


Figure 43

The steps for programming Preset 11 are as follows:

1. Press the Edit Button to select the preset programming mode.

2. Press the Preset 11 button to select Preset 11 for programming.

3. Assign (Definitely On) the front desk Paging Microphone to all zones using the appropriate Mix Bus Assign Buttons (the red diamonds on the Mix Bus Assign Buttons shown in Figure 43 on the preceding page demonstrate that this has been done).

4. Turn off every other input that could possible feed any zone (the red rectangles on Mix Bus Assign Buttons shown in Figure 43 demonstrate that this has been done).

5. Turn off (disable) the remote controls for all zones (notice the RB buttons for all zone outputs are not longer depressed. Clicking on an RB Button releases it. While the graphic in Figure 43 shows the buttons released, it cannot show that they would be red in color. On the ANSW+ software screen, they *would* be red to indicate they had been changed as part of the preset.

6. Using the up/down level buttons, adjust the page level coming into each zone. (Notice that each zone's output level has been turned up from -10dB to -6dB as Figure 43 on the preceding page shows.) (Again, the changed level would be red in color.)

7. Press the Save Button to save the programming of Preset 11 to the *884+* mixer's memory.

8. Press the Edit Button again to exit the preset programming mode.

This would complete the programming of the *884+* for the three meeting room application. Eleven presets were used to provide a number of features. A reasonably complex and demanding set of requirements were routinely handled without the use of relays, diodes, or external logic of any kind. The nice thing about it is that it was all done with only one mixer. Of course, had additional inputs been needed, an *880+* Input Expander could have been used, or a *780+* Input Expander if the addition inputs did not need to be automatic. The versatility is apparent.

After presets have been programmed, they can be activated by the ANSW+ software, a contact closure, or by command strings coming from such things as AMX® or Crestron® touch panels. When a mixer or input expander receives a status change command, it executes the command and then reports back its updated status. This updated status is displayed by the ANSW+ software. If monitoring a unit via the ANSW+ software, the ANSW+ software screen will be updated to reflect any changes in status commanded by any control source. For example, if a remote control is active and someone is making level changes, the software will reflect those changes as they occur. If a touch panel request a room combine, the unit will execute the room combine as instructed, and the ANSW+ software will visually display the updated status.

This is a useful feature because it can be a nice programming aid. If, after programming a preset, one desires to check his programming accuracy, he can save the preset and then exit the Edit Mode. Now, by pressing the preset button of the preset he has just programmed, he can activate that preset. The unit will then display the preset active information and the programmer can consult the control screen to make certain everything that was supposed to happen, did in fact happen. This is especially useful when multiple presets are active. The resultant composite will be shown on screen and the programmer can check to make sure the preset priorities are maintained as he wants them to be. He can also assure by successive activation that each preset does what it is supposed to do.

An examination of this feature may prove helpful. For the three meeting room example just programmed, Figure 44 shows Preset 0 only active and Figure 45 shows Presets 0 and 11 active:

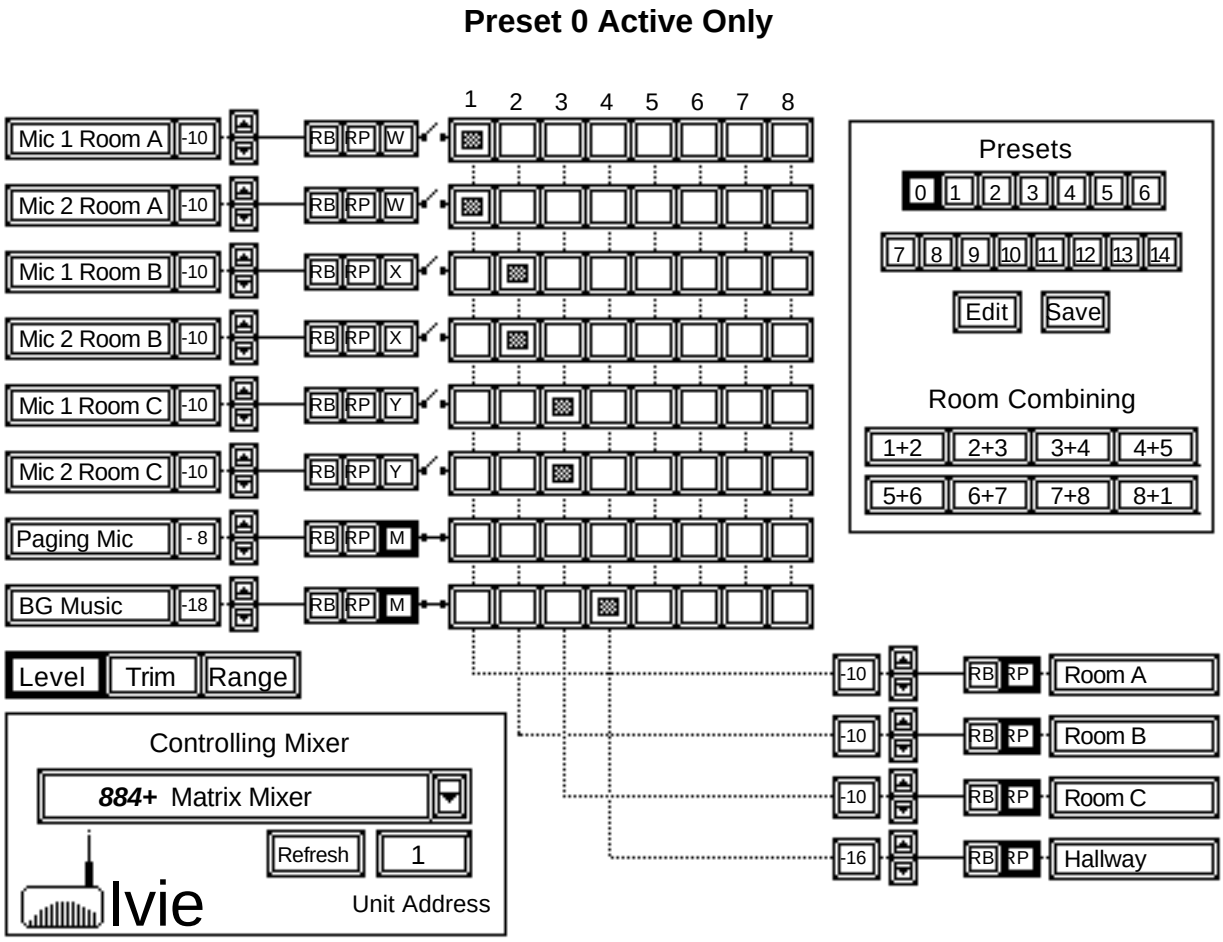


Figure 44

Preset 0 remains just as it was programmed earlier for the three meeting room application. The Preset and Room Combining Window shows that only Preset 0 is active. None of the buttons for Presets 1 through 14 are pressed.

No room combining is active. The Preset and Room Combining Window shows that none of the room combining buttons have been pressed.

The remote controls for all the outputs are active. Notice that the RB Button has been pressed for each one of the zones: Room A, Room B, Room C, and the Hallway.

Rooms A, B and C have their level set at -10dB, as the Level Window shows. The Hallway has its level set at -16dB.

Remember that the changes for Preset 11 would affect the bus assignment for all the zones because the only signal allowed to come through would be the All Page. Remember also that the remote controls would be turned off by Preset 11, and the levels into each zone would be set to a nice, loud level so the All Page would come through clearly. Figure 45 on the following page shows what the ANSW+ software screen would display if Preset 11 were activated by any method (pressing the Preset 11 Button

on the ANSW+ software screen, closing the contact associated with Preset 11, or by a software command coming from an external source). Regardless of how Preset 11 were activated, the ANSW+ software control screen would reflect the changes shown in Figure 45 below:

Preset 0 and Preset 11 Active

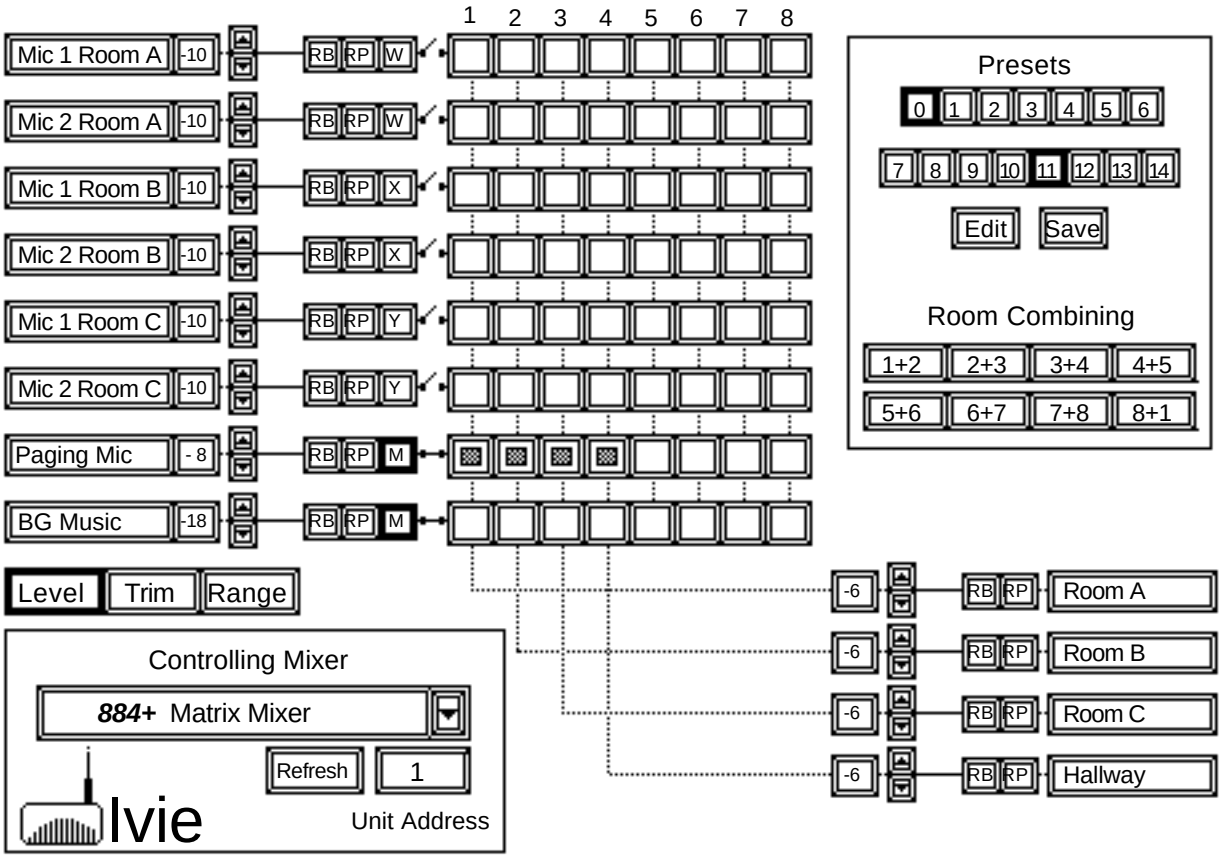


Figure 45

Notice the following:

1. The Preset 0 and Preset 11 Buttons are depressed indicating that Presets 0 and 11 are active.
2. The only signal going to any output is the Paging Microphone input.
3. The remote controls for each zone have been disabled. None of the RB Buttons is pressed.
4. The output level for each of the zones has been reset (turned up) to -6dB.

Preset 11 did everything it was supposed to do. It must have been correctly programmed.

Room Combining Commands

Room combine commands may also be programmed as part of a preset. In the preceding application, no room combine commands were necessary as part of a preset. However, room combining was a feature that was provided. Room A could be combined with Room B, or Room B with Room C, etc.

When a room combine command is given, all active presets affecting either room are copied over to both rooms. The room combine command is a "smart" preset. Figure 44 shows only Preset 0 active. If a "Combine Room A with Room B" command were given, (ie. pressing the 1+2 Room Combining Button), the changes shown in Figure 46 below would occur :

Combining Room A with Room B

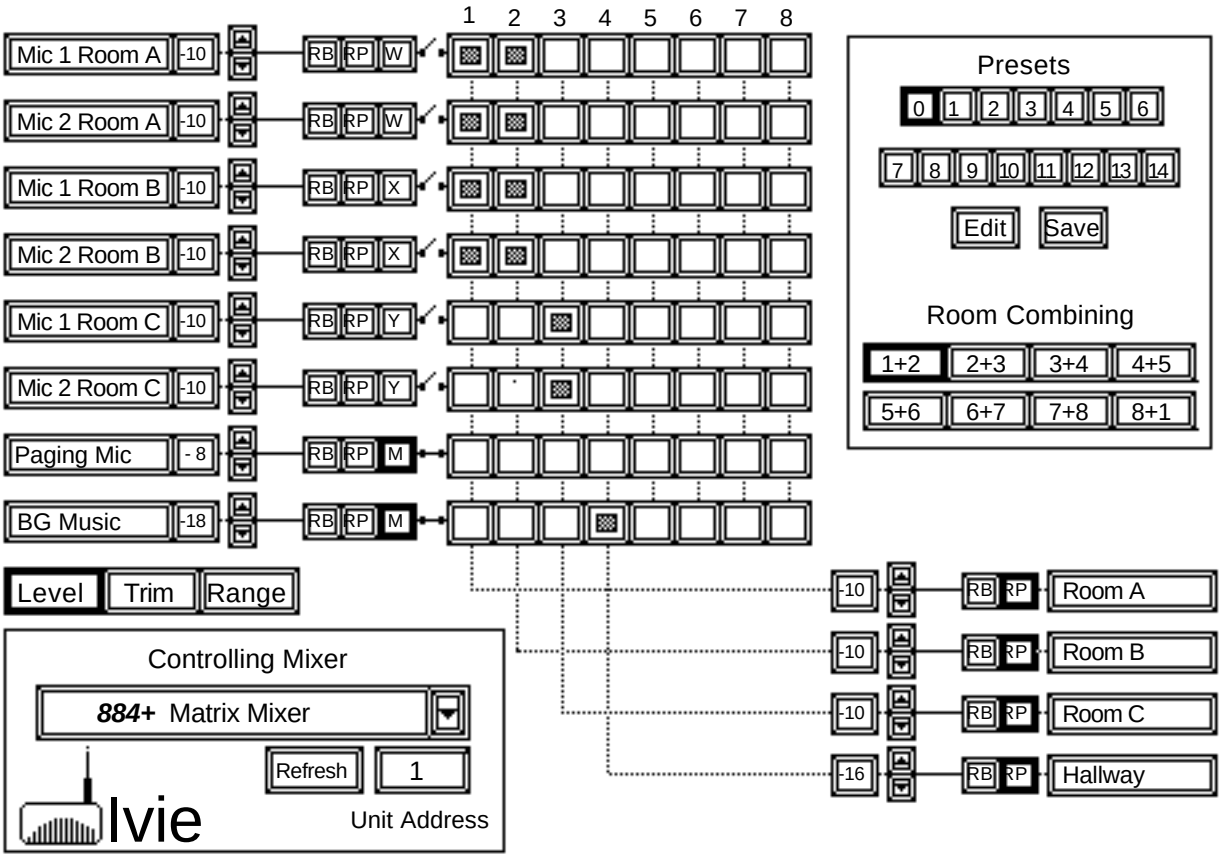


Figure 46

Notice the following:

1. The 1+2 Room Combine Button has been pressed to combine Rooms A and B.

2. The Room A microphones now feed both Room A and B, and the Room B microphones now feed both Room A and B, just as would be expected in a room combine situation.

When rooms are combined, remote controls, if desired, can tie together and track one another such that volume for both rooms can be adjusted from either room. On the other hand, if no tracking is wanted, so that each room or zone has its own volume control, this option is possible as well. The RM-1 and RM-2 from Ivie have a built-in tracking feature that can be used, or not used as desired. The RB-1 up/down button control from Ivie (which was the one used in the three meeting room application) does not track on its own with the older 784PW and 884PW product, but it does track with the newer Audio Net®+ products (*784+, 884+, 780+ and 880+*). This is a nice feature that has been added.

When rooms are combined, preset activation normally affecting only a single room, will affect all combined rooms, just as would be expected. In the room combining example above, if background music were turned on in Room A by activating Preset 1, background music would appear in both rooms and all microphones would be turned off. Figure 47 below demonstrates this:

Combining Room A with Room B with Preset 1 Active

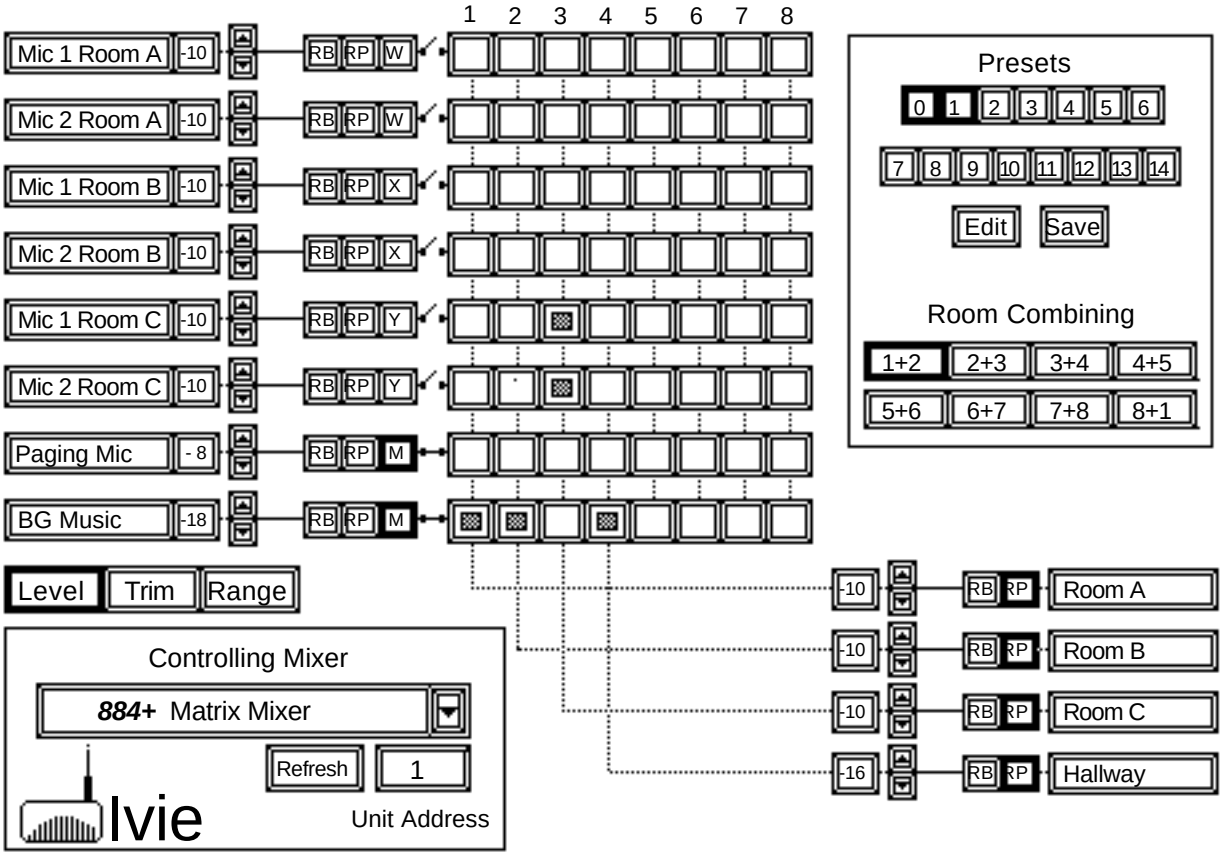


Figure 47

Notice the following:

1. The 1+2 Room Combine Button has been pressed to combine Rooms A and B.

2. Preset 1 is active which normally feeds background music only to Room A, but when Rooms A and B are combined, background music is fed to both rooms, just as would be expected.

Room Combining Using More Than One Mixer

When mixers and/or input expanders are ganged together, a preset or room combining requirement that "crosses boxes" must have presets or room combines in all affected "boxes" actuated simultaneously. As an example, consider two 8 X 4 matrix mixers ganged together using Ivie 78C-6 cables to form one 16 X 8 mixer. Mixer A has two inputs each assigned to Mixes Buses 1, 2, 3 and 4. Mixer B has two inputs each assigned to Mix Buses 5, 6, 7 and 8. (That uses up all sixteen inputs and all eight outputs.)

If a room combining command is needed to combine mix buses 5 and 6, a situation arises that "crosses boxes." Mix Bus 5 gets its inputs from mixer (or "box") A, while mix bus 6 gets its inputs from mixer B. The only way to effect a complete "room combine" is to execute the 5+6 room combine command in both mixers simultaneously. This can be done by a remote device sending software control code, first to one mixer and then to the other. It can also be done with a single contact closure. The 5+6 room combine contacts from both mixers can be brought to a single switch. *Diode isolation on the lines coming to the contact closure should be employed.* Figure 48 below shows how the wiring for this example would be done:

Activating More Than One Preset or Room Combine With One Contact Closure

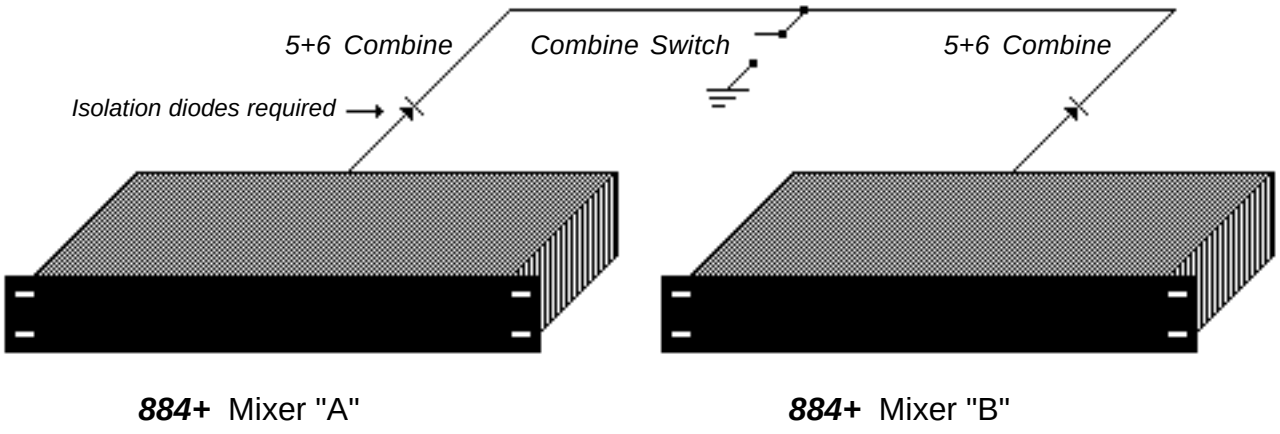


Figure 48

Figure 48 above shows the proper placement of diodes when wiring for activating a preset or room combine from more than one unit using a single contact closure. More than two mixers or input expanders could be handled in exactly the same manner.

For more detailed wiring diagrams, and further information on executing multiple presets and room

combines simultaneously, consult the section of this manual titled **Hardware Activation of Presets** on pages 15, 16 and 17. By programming presets and using room combining, the mixers and input expanders can be easily adapted to a wide variety of applications and venues. Flexibility is almost unlimited. Technical support is always available from the factory. Please do not hesitate to phone with any questions that may arise.

Programming Tips and Tricks *(Mixers and Input Expanders)*

Plan ahead in programming. The priority of events triggered by preset activation should be carefully considered so that the proper priority occurs as desired.

The most intensive communication that occurs on the network takes place when a preset is saved. If there is any problem with the network (insufficient power to drive an Ivie 78-232A, RS-232 to RS-485 converter, for example) it will show up in the preset saving process. Always check the saving process by activating the preset that has just been programmed. Make sure that every change wanted is properly reflected on the software control screen.

The Trim and Range functions can be saved as part of Preset 0*, but they cannot be changed as part of Presets 1 through 14.* Once set and saved as part of Preset 0, they are universal and are carried through in all the rest of the presets. Every other function, *including Automix Parameters,* can be saved as part of Presets 1 through 14.

When two automixers inside one box are combined by a room combine command, they do not become one automixer. Their NOM, threshold etc. remain individual. Automixers *can* be reconfigured as part of a preset. If requirements necessitate combined rooms being one automixer, a preset dedicated to making them one automixer can be programmed. As long as the two commands occur simultaneously (the room combining command and the preset command reconfiguring the two automixers as one automixer), the desired results will be obtained. A preset command and a room combining command can be executed simultaneously by a single contact closure. *Diode isolation on the two lines coming to the contact closure should be employed.*

Avoiding network collisions saves a lot of grief. All Ivie Audio Net+® products are designed to live on the Audio Net+® network together such that all units may be addressed by plugging into any unit. Since the Audio Net+® network is a two-wire RS-485 network, both send and receive information occur over the same twisted pair. Whenever a unit receives a command, it executes the command and then echoes its updated status back over the network. That updated status is displayed by the ANSW+ software. It is important that only one unit at a time reports its status. The ANSW+ software handles this by putting all units to "sleep" (non echo mode) except the one being controlled. The command of polling and selecting a given unit for communication automatically executes the "sleep" command to all other units. When the software is exited, it puts all units to "sleep." *When plugged into the front of a unit to address it via the ANSW+ software, it is not a good idea to unplug before exiting the software.* This leaves the unit "awake." If several units are left "awake" in this manner, an event on the network could cause them all to try to report their status at once. Collisions and possible problems could be the result. *Always exit the ANSW+ software before unplugging from the network.* This will put all units on the network to "sleep" and prevent any unnecessary collisions.

Appendix III
Setting Auto Level Control *(884+ and 880+ Only)*

Theory of Operation

All inputs of the *880+* Input Expander and the *884+* Automatic Matrix Mixer are equipped with an automatic level control (ALC) which can be activated or deactivated. Each channel is independent of the others, so ALC can be active for one channel only, or for any combination of channels.

When properly set up, *the ALC will absolutely not send a system into feedback.* Conversely, it is not magic. *It cannot increase system gain before feedback.* Perhaps the ALC is easiest thought of as a remote control that has had its gain trimmed such that it can turn a system up to just below feedback, but not any higher. No matter how hard the operator of the remote control tries, he cannot turn up the system far enough to go into feedback. He can turn down microphone users that are too loud into the room, and turn up those who are too soft. However, he cannot turn up someone who is soft spoken any louder than just below system feedback. That is the maximum gain available - the system limits have been reached - and if the person is still not loud enough, he must speak up or get closer to the mic, or both.

The Ivie ALC functions in exactly this manner. It replaces the operator of the "remote control" in the analogy above. The "level control" the ALC uses is the input VCA. When the Automatic Level Control is turned on, it will try, by controlling the VCA, to hold a constant level (target level) at the output of the VCA. The ALC has a maximum range of 32dB and the mixer or input expander can automatically adjust level over that range in order to keep the sound level in the room constant. The ALC factory default setting turns the level down (attenuates) at a rate of 10dB per second (20dB in two seconds). It turns the level up at a rate of 2.5dB per second (20dB in eight seconds). These rates work very well over the vast majority of application requirements, but can be altered when necessary using the Automix Parameters pop-up window under the Automix pull-down menu.

Figure 49 below details the functional range of the ALC:

ALC 32dB Operating Range

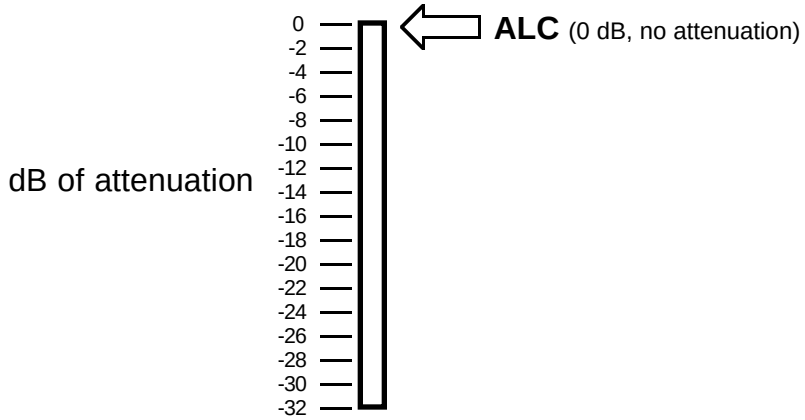


Figure 49

As can be seen is Figure 49 above, the maximum range of the ALC is 32dB. It operates over that range to try to keep a constant "target" level at the output of the VCA. The signal which the target level

maintains can actually be measured at the "Channel Direct Out" (see the inside of the rear cover of this manual to locate the Channel Direct Out terminals). The target level is adjustable from -10dBu to +4dBu in 1dB steps. The factory default setting for the target level is -10dBu.

Think of the ALC target level as the "loudness" in the room. There is a relationship between the target level and the level appearing at the Master output. If the Master Gain Trim and Master Level Control are both set at 0dB of attenuation, the level measured at the Master output will be the same as the target level measured at the Channel Direct Out. If, for example, the ALC is maintaining a target signal level of -10dBu, the Master Output level will measure -10dBu as well.

In *Figure 49* on the preceding page, the arrow shows the position of the ALC at 0dB of attenuation. Consider again the analogy of the sound system with a remote control. The system was adjusted for maximum gain before feedback. The remote control (the ALC in the analogy) is turned up as loud as it can go. The system is just below feedback, but the person speaking can be nicely heard. The ALC (remote control in the analogy) is at the top of its range. Now, imagine someone with a very loud voice comes to the microphone and begins to speak. The person operating the remote control will need to turn him down. Consider Figure 50 below:

ALC 32dB Operating Range

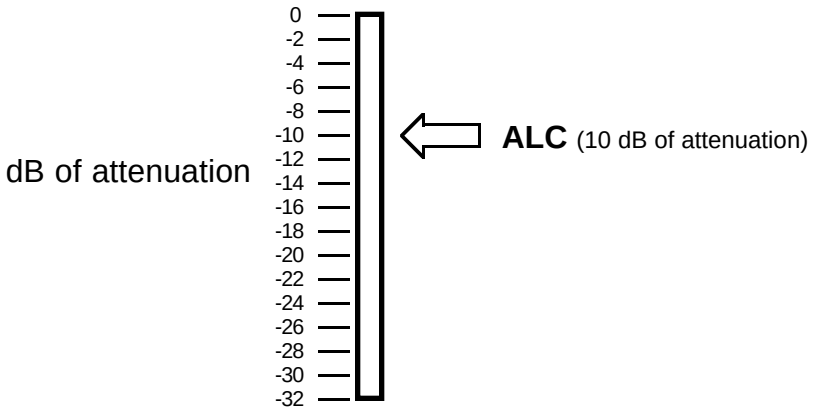


Figure 50

In Figure 50 above, the ALC (remote control) has turned down the person speaking by 10dB in order to maintain the same "loudness" in the room. The ALC still has another 22dB of attenuation available and it can now "turn up" the system by as much as 10dB before the system comes back again to the "edge of feedback."

This is exactly how the ALC works. It is capable of automatically turning down signals that are too loud into the room, or turning up signals that are too soft until it hits its maximum level (0 dB of attenuation) at just below feedback. *For best results, the ALC should be set up such that the top of its range is just below feedback.* The following procedure explains how this is done.

Automatic Level Control Setup Procedure

The ALC setup procedure begins as though there were no ALC available. With the ALC for each channel turned off, the system gain structures should be adjusted according to standard installation procedures to achieve maximum gain before feedback. Input and output levels should be set, along

with any Range and Trim setting desired, and gain structures through the rest of the system finalized. *To maintain good signal to noise, the input and output levels on a matrix mixer (input levels only on an input expander) should be set in the upper quarter of their range (not more than 1/4 below "full on").* Once the system has been balanced, the ALC for each input channel can be set. ALC can be turned on via the "ALC and Vox Sens" menu item under the Automix pull-down menu.

To properly setup the ALC, a pop-up window is provided under the menu titled "ALC Levels," found under the Automix pull-down menu. This pop-up window provides the ability to monitor and set any two channels of ALC. The channels can be selected and the action of their associated ALC's viewed on a chart recorder display. One channel is shown in blue and the other in red. Figure 51 below details the the controls and strip chart recorder display of the ALC Level pop-up window:

The ALC Level Pop-Up Window

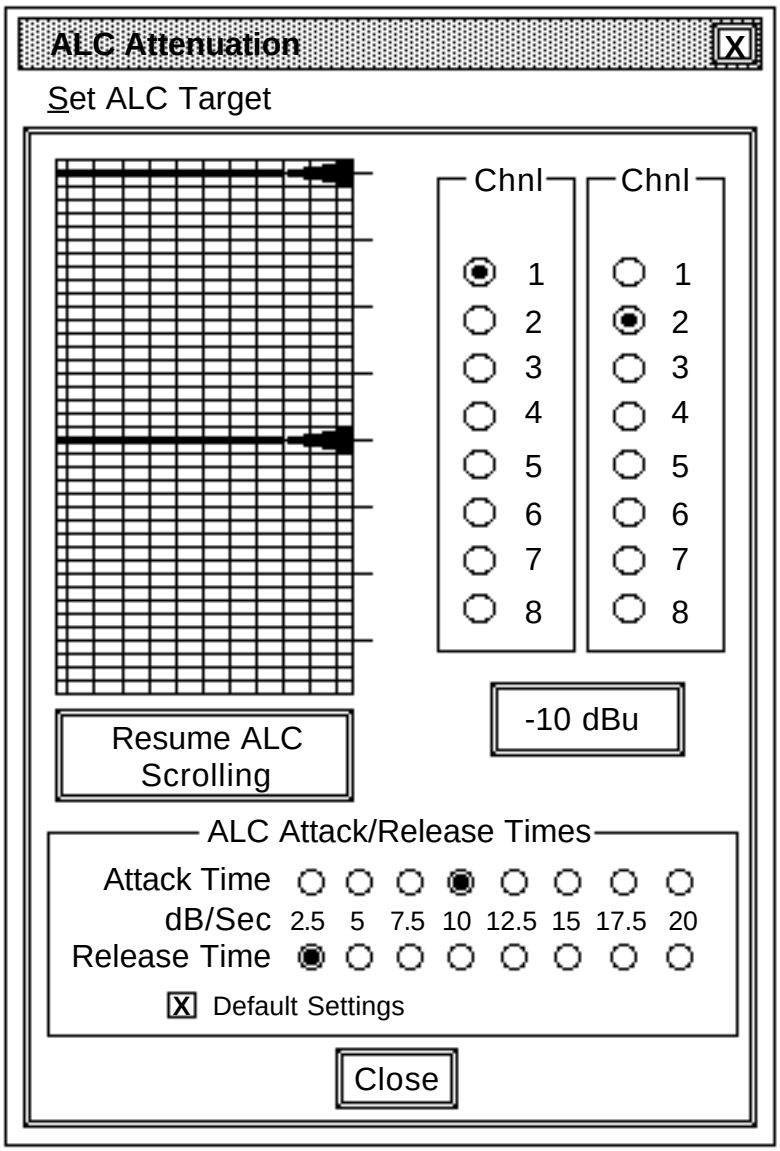


Figure 51

The Automatic Level Control can be set using the following steps:

1. With ALC off, balance the system for maximum gain before feedback.

2. Turn on ALC via the "ALC and Vox Sens" menu item under the Automix pull-down menu.

3. Open the ALC Level pop-up window under the Automix pull-down menu. Select the inputs for ALC setup (one in red and one in blue).

4. Choose one channel (only one channel at a time can be set). Begin speaking into the microphone for that channel The microphone should be spoken into at the level expected under normal operating conditions.

5. Watch the action of the pen in the strip chart recorder for the channel being set up. There should be visible action of the ALC. If there is not, the gain structure will need to be changed a little by turning up the input and turning down the output at the system amplifier, still maintaining maximum gain before feedback.

6. Notice where the pen is riding as the microphone is being spoken into. The recorder pen indicates the action of the ALC. If the input signal is greater than the target level that has been set, the ALC will reduce the input level until it matches the target. The recorder pen will indicate how much attenuation the ALC is providing.

7. Listen to the sound level in the room to see if it is loud enough. If it is not, adjust the target level upward using the "Set ALC Target" pull-down menu. (In earlier versions of ANSW software, the Set ALC Target menu was called "Set ALC Threshold." The functions are the same, only the name has changed). *The ALC Target becomes the room volume control.* Set the target at a level where the sound is as loud as wanted with normal voice input. The attenuation the ALC is providing will be reduced in direct proportion to the number of dB the target is being increased. If, for example, the pen shows ALC attenuation of 10dB, and the target is increased by 6dB, the ALC attenuation will change to 4dB. Once the ALC attenuation reaches 0dB of attenuation, it will do no good to turn the target up any higher. The ALC will not be providing any attenuation, so the limits of system gain before feedback will have been reached. In other words, *if the ALC is not attenuating, the system is as loud as it can get.*

8. Adjust ALC for the rest of the inputs. *Once the target level has been set, moving it to accommodate a second input would only counteract the first input setup.* When the target level has already been set to provide the desired sound level in the room, it is generally best to leave it. If a "hot" input is forcing the ALC to attenuate more than desired, this can be countered by turning the input level down. This will reduce the ALC attenuation and allow the ALC to "ride" closer to the top of its range.

9. Adjust Attack and/or Release Time, *only if necessary*. The default settings of 10dB/sec. for Attack Time and 2.5dB/sec. for Release Time are best for most applications. Adjusting Attack and Release Times too much can cause more problems than it solves.

Note: While viewing the action of the ALC on screen, making any adjustment to the mixer or input expander will cause the strip chart recorder to stop scrolling. ALC monitoring is network intensive and stopping the scroll helps prevent too much traffic on the network. To resume the scrolling action, simply press the Resume ALC Scrolling Button.

Appendix IV
Custom Room Combining

An extremely powerful feature of the Audio Net+® products is the addition of custom room combining capability. In addition to custom room combining capability, the "plus" products allow the software assignment of the Function Outputs to follow various functions. As previously discussed, the factory default setting for the Function Outputs on an *884+*, for example, is that the Function Outputs follow channel gating. If the application requires the Function Outputs to follow room combine or preset activation, this can be easily accomplished.

Function Output Assignment

To change the assignment of the Function Outputs, select the Room Combine Set-Up screen under the Edit pull down menu. Figure 52 below details the Room Combine Set-Up screen:

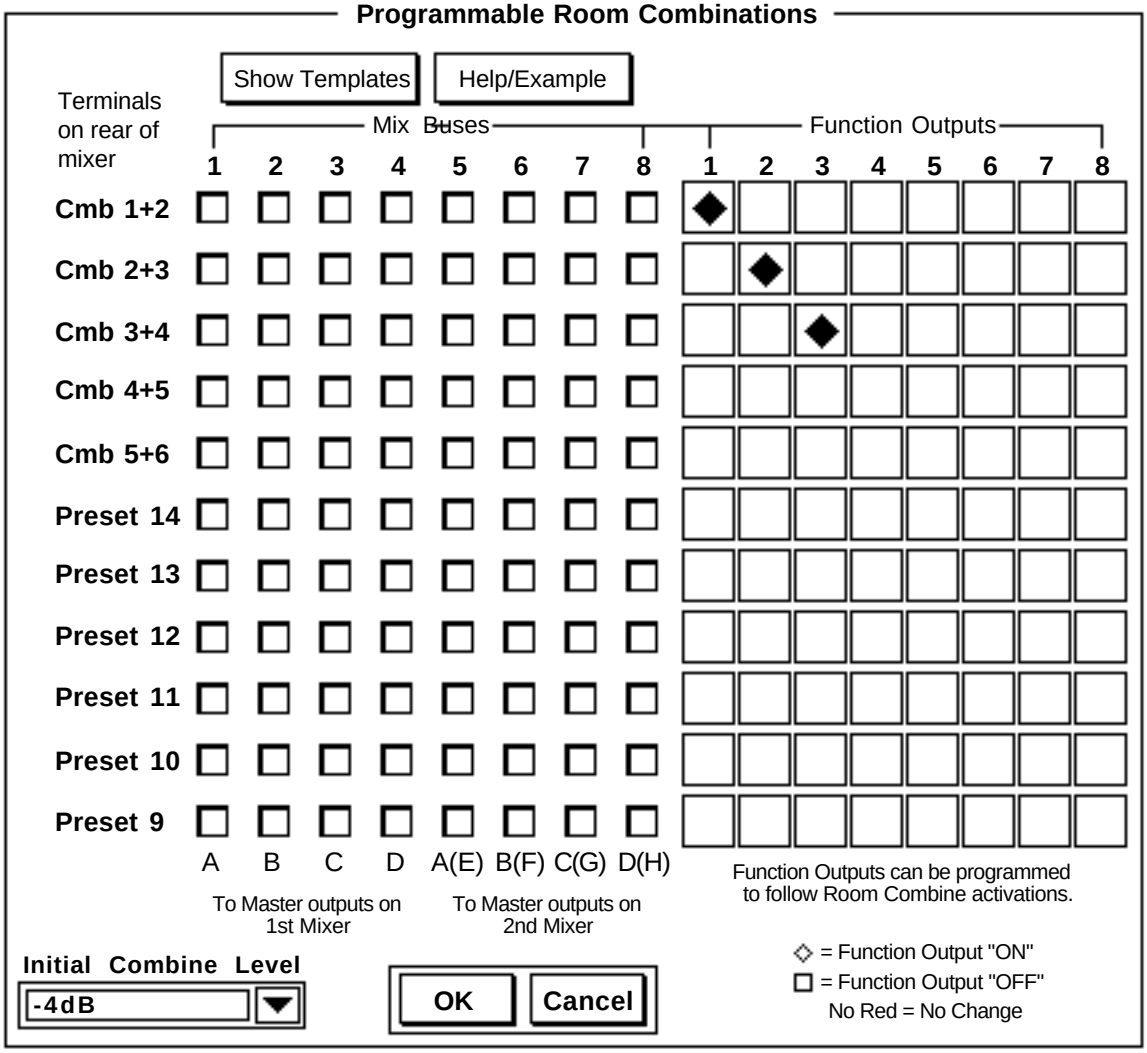


Figure 52

In Figure 52 on the preceding page, notice that Function Out 1 has been set to activate on Room Combine 1 + 2. Function Out 2 has been set to follow Room Combine 2 + 3 and Function Out 3 has been set to follow Room Combine 3 + 4. Function Outs 4, 5, 6, 7 and 8 have not been modified, so they would continue to adhere to the factory default setting of following gating.

A window has been added to the ANSW+ control software screen which illustrates the status of the Function Outputs. The status of all the Function Outputs can be determined at a glance. Figure 53 below shows this window in relationship to the other controls of the software screen:

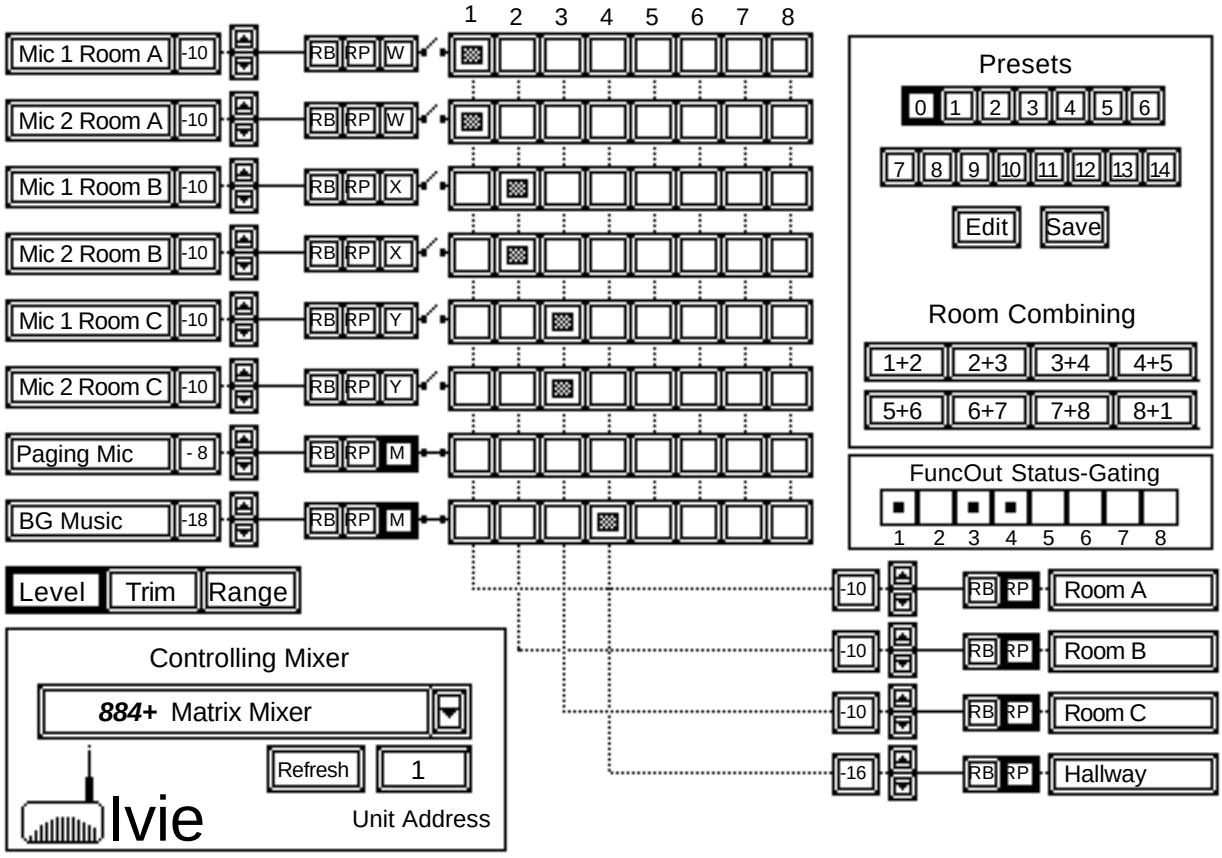


Figure 53

The Function Output Status window is located on the right-center, right below the Room Combining control buttons. As can be seen in Figure 53 above, Function Outputs 1, 3 and 4 are active.

Function Output Protection

Often the Function Outputs are used in conjunction with an external power supply to drive relays. If the output of a 24 volt power supply, for example, were directly connected to a Function Output by mistake, under normal circumstance the Function Output would be blown. To guard against this possibility, Ivie has added protection to all of the Function Outputs. A “circuit breaker” chip has been added which trips when a Function Output sees a voltage coming into it which could damage it.

When a Function Output circuit breaker trips, the power LED on the front of the mixer or input expander will begin to flash. It will flash for approximately three seconds and then the unit will attempt to reset the circuit breaker. If the fault still exists, the breaker will trip again causing the front panel power LED to flash for another three seconds. This cycle will continue until the fault is cleared and the circuit breaker successfully resets. Obviously, if a Function Output seems not to be working, the first item to check would be the front panel power LED.

Doing Custom Room Combining

The Room Combine Set-Up screen under the Edit pull down menu also provides for programming a mixer or input expander to do custom room combining. Additionally, the initial room combine level can be set using the click and drag control. The default pop-up window is shown in Figure 52. By pressing the Show Templates button, the pop-up window is modified as shown in Figure 54 below:

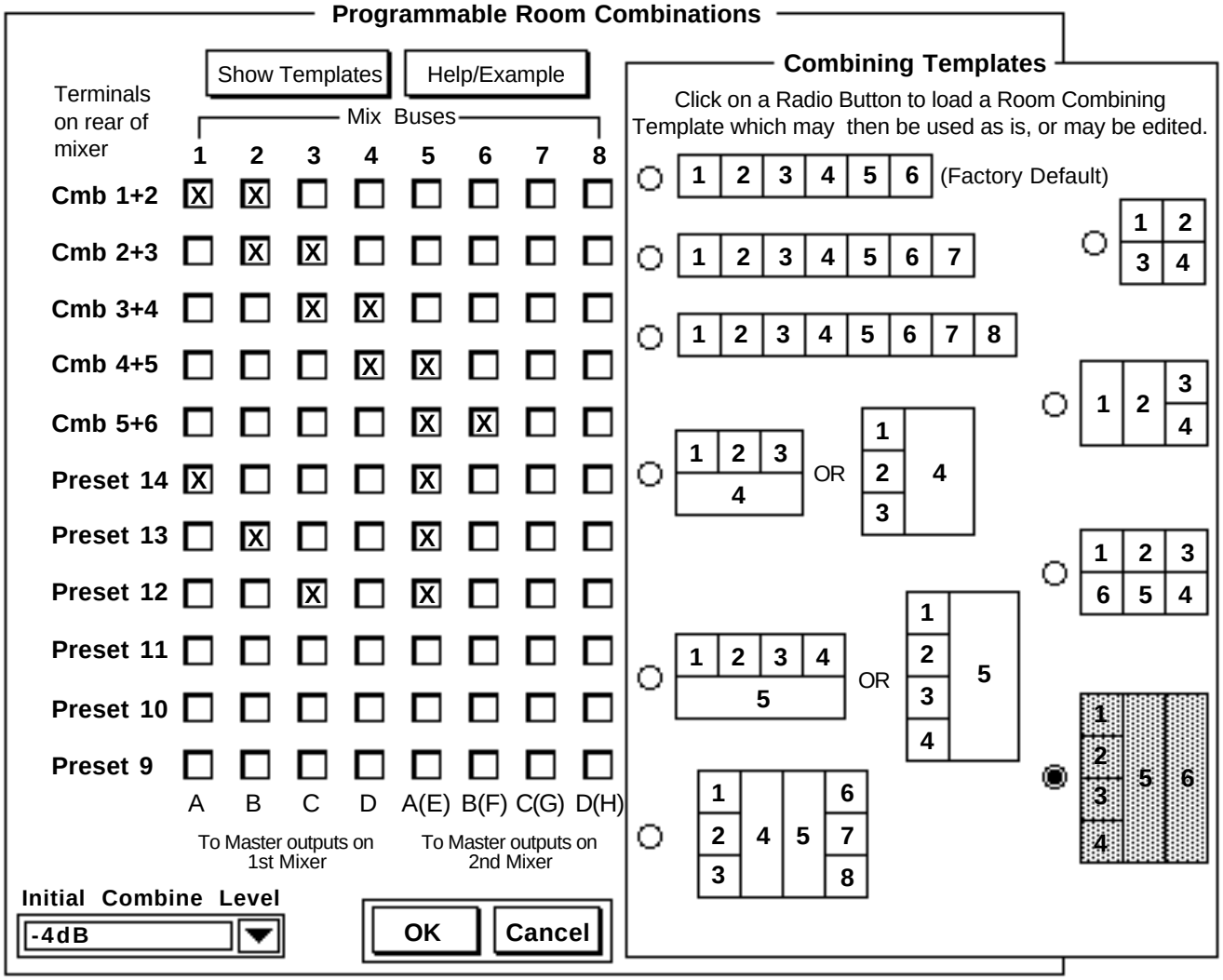


Figure 54

The templates shown in Figure 54 above provide easy programming if the room layout needed follows one of the templates. If the layout is different from any of the templates shown, selecting the template closest to the needed layout provides both a starting point and a programming example. In Figure 54 on the preceding page, the template at the bottom right has been selected. This template has six rooms and eight combining nodes (the eight combining nodes are: 1+2, 2+3, 3+4, 1+5, 2+5, 3+5, 4+5 and 5+6). Since the mixers and input expanders have only five sets of room combine contacts on the rear panel, the last three combining nodes "steal" Presets 14, 13 and 12 to use their rear panel contacts and their function as presets. For example, closing the contact labeled Preset 14 will combine rooms 1 and 5. Since the contacts have been remapped for use as room combines, they cannot be used for presets.

The ANSW control software screen gives visual indication when presets have been assigned for use as room combines. Figure 55 below illustrates this:

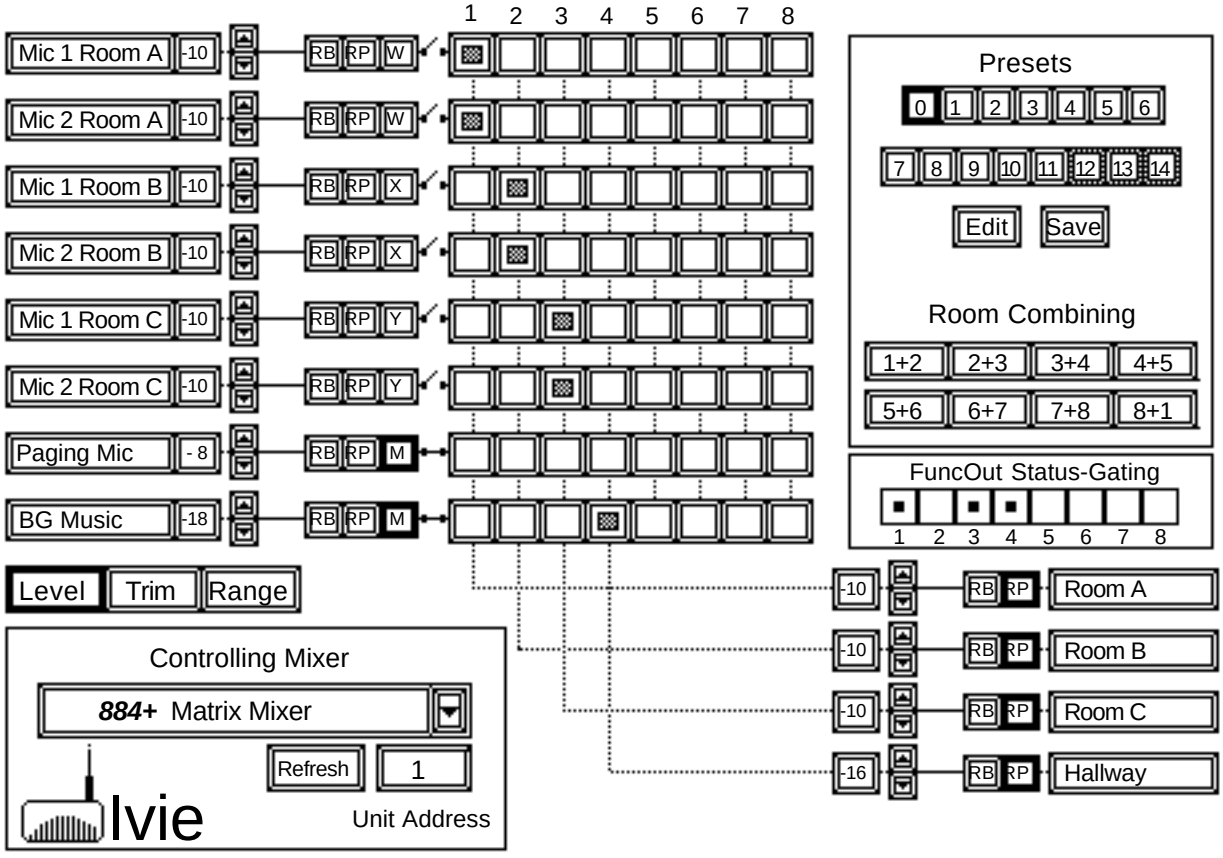


Figure 55

Notice that in Figure 55 above, Presets 14, 13 and 12 have a yellow band around the buttons indicating that they have been assigned to room combine functions. The ANSW software will not allow them to be edited or programmed as presets.

By selecting room combine templates and/or modifying them, or by bypassing the templates and programming using the radio buttons, and room combine combination up to eight rooms and eleven combining nodes may be successfully programmed.

Appendix V
Audio Net® T+ Products

Everything that has been said about the *780+, 784+, 880+ and 884+* applies directly to the *780T+, 784T+, 880T+* and *884T+*. A *780+* and an *780T+*, for example, are identical except for one feature: The *780T+* has three bands of tone control for each input channel and the *780+* does not. All of the *T+* units have tone controls added. The tone controls are a hardware implementation (rather than a software implementation) which requires that all *T+* units be two rack spaces high instead of one. The second rack space provides the mechanical space to house the 24 tone controls which are accessed at the rear panel. Except for the double rack space and the tone controls, corresponding units (*884+* and *884T+*, for example) are functionally identical in all respects.

The characteristics of the tone controls are as follows:

Low Pass: 320 Hz turnover with ± 10dB shelving.
Mid Band: 800 Hz, + 10dB boost or cut.
High Pass: 2kHz turnover with ± 10dB shelving.

Index

Index Continued:

Index Continued: